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SOUND ON SOUND™

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Strings

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IK Multimedia ARC

Can a plug-in really cure bad acoustics?

Focusrite Liquid 4 Pre

Exclusive review of "the most flexible preamp in history"

Foo Fighters

Rich Costey: mixing a hard rock anthem

www.soundonsound.com

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Are You Cheating?

I still remember trying to explain the concept of the Mellotron to a musician friend back in my college days, and she was of the opinion that it constituted gross cheating of the first order. After all, if you want strings, surely you should employ a string player, not have some mechanical 'piano thingie' that sounds like strings!

How times have changed. Today the debate isn't even about whether samples and loops are cheating, but how much creative input is needed before copy and paste becomes art. I'd be the last person to decry the use of loops, particularly percussion, as I often combine loops with my own programmed rhythms, but I do get the impression that some music technology students and would-be producers see 'making beats' as the be-all and end-all of creative music production.

Perhaps we shouldn't blame them, though. If you listen to the pop charts these days, you might hear where they get their inspiration. Take Soulja Boy's 'Crank Dat', which reached number two in the UK singles chart in December 2007; in terms of instrumentation, all you'll find is an 'orchestra hit' sample, some programmed percussion (including a prominent steel drum pattern, which gives the song its simple melodic content), and rap vocals. If you took the vocal off, you wouldn't be left with much!

I see the manipulation of loops and audio clips as being a bit like those Victorian montages made from photographs, postcards and prints of paintings. It is certainly a skill and an art, but it covers only a narrow part of the skill set required to be an all-round artist. You may be great at cutting out pictures and pasting them on a fire-screen, but if somebody wants a painting in the style of Monet or Dali, it isn't going to help you much.

Music production is much the same, and while a lot of dance and hip-hop music uses loops, there's a lot more to making a great track than figuring out which drum breaks to sample or by how much to time-stretch them. You also need to know about musical structure, melody and mixing. Loops and samples have evolved from being an almost completely separate area of expertise to being part of the general pop music recording process, and though they're no less important for that, someone who only knows about manipulating existing material is unlikely to be properly equipped for music production and engineering in the broader sense. Furthermore, even those who are currently successful working in a genre that relies heavily on loops and samples can't rely on that state of affairs continuing, because like every aspect of modern life, music production evolves and shows no respect for the people it leaves behind.

The core of broader music recording and production always has been — and is always likely to be — about putting microphones in front of real musicians. And while some elements can be replaced or enhanced by the use of loops and samples, the ability to use microphones and coax great performances out of singers and instrumentalists will always be a number one priority. So no, using samples, loops and clips isn't cheating, but if you're not prepared to look beyond your comfort zone, you might just discover that your skill set doesn't equip you for the kind of position in the music industry that you hoped for when you started out.

Paul White *Editor In Chief*

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

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

ISSN 0951-6816
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Small enough to suit the most compact production suite, or be easily carried from studio to studio, the new Duende Mini packs SSL console-grade processing into a DSP-loaded, mini desktop box. Including our classic EQ & Dynamics channel strip and trial versions of all the plug-ins in our expanding range, Duende Mini uses a fast, simple FireWire connection to bring 16 channels of SSL's hit-making magic to your PC or Mac, with the option to upgrade to 32.

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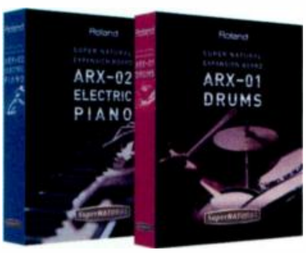
In January, the *Sound On Sound* team attended the NAMM show in Anaheim, California, where we saw lots of exciting new products. As ever, it's impossible to cover all the announcements in these pages, so watch out for more news in upcoming issues of *SOS*.

Fantoms go Supernatural

Roland update workstation synths

The latest iteration of Roland's Fantom series of workstation synths was unveiled at NAMM. There are three entrants in the Fantom-G line-up: the G8 has a weighted 88-note keyboard, while the G7 and G6 boast 76 and 61 'synth-action' keys respectively. All feature a new high-resolution colour screen, which can be fully exploited by connecting a USB mouse. A new 'power sequencer' can play back up to 24 audio tracks, as well as over 100 MIDI tracks, and there's also a powerful new effects engine which makes up to 22 effects routings available in 16-part Multi mode.



The new Fantoms are also part of the first generation of Roland instruments to support their new Supernatural ARX expansion boards. These each contain a complete, integrated synth, and the first two to be launched combine sample-based synthesis with modelling technology to provide responsive and realistic instrument sounds.

ARX01 Drums allows the user to alter parameters such as shell depth, tuning and muffling in real time, while processing the results through state-of-the-art effects. ARX02 Electric Pianos, meanwhile, presents controls such as tine angle, pickup distance and 'bell character', providing an infinitely mutable electric piano sound with no stepping between velocity layers or multisample key groups.

Roland UK +44 (0)1792 702701

www.roland.co.uk

Anamod AM660

Fairchild in a Lunchbox

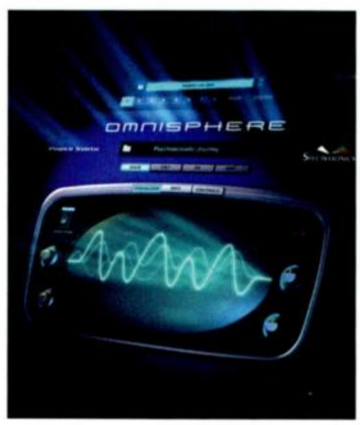
At last year's NAMM, we were introduced to the Anamod ATS1, a hardware device that uses components similar to RAM chips to store tape simulation models, allowing additional models to be retrofitted. At this year's show, Anamod were showing the AM660, a module designed for the API500 'Lunchbox' that models the legendary Fairchild 660 variable-mu compressor. It costs $1295 in the USA and is available directly from Anamod.

www.anamodaudio.com

Spectrasonics Omnisphere

Spectrasonics' founder and sound designer Eric Persing made an appearance at NAMM this year, having been absent from the past few shows. Rumour had it that he was updating the Atmosphere and Trilogy software instruments with new engines, in the same way that the Stylus RMX 'groove module' was updated from the original Stylus using the then-new SAGE engine — but this seems not to have been the case. Eric and his team were working on a new engine, but its first outing is to be with a new plug-in instrument called Omnisphere. The so-called Steam Engine underpinning this instrument is designed to support multiple synthesis methods and process samples in a very sophisticated way. While instruments based around samples are nothing new, Spectrasonics have a habit of coming up with something special and more than a hint weird.

Omnisphere was shown at the NAMM show using a limited sound set, and Eric tells us it won't be shipping until September 15th this year, as there's the huge task ahead of cataloguing and organising the samples that will go into its substantial library. After that, who knows what the age of Steam will bring?



Time & Space

+44 (0)1837 55200

www.timespace.com

www.spectrasonics.com

Avid's NAMM announcements

The audio division of the Avid group, which includes Digidesign, M-Audio and Sibelius, launched a number of products at this year's NAMM show.

New from M-Audio (www.m-audio.com) is the Profire 2626, a 24-bit/192kHz Firewire audio interface (pictured right) capable of recording 26 input channels and playing back 26 channels simultaneously. The front of the 1U rackmountable device has gain controls for each of the eight mic preamps (which are the same as those found in the M-Audio Octane), and volume knobs for the 2626's two headphone outputs. On the rear panel, combi XLR ports allow for XLR and TRS jack plugs to be connected to the inputs, while jack outputs serve the eight analogue outputs. Twin ADAT inputs and outputs allow for a further 16 digital connections to be handled by the device, and there's a serial port with a breakout cable that gives extra connections for S/PDIF and MIDI ins and outs.

Interestingly, the device has on-board DSP that allows the user to route signals internally and configure up to eight mixes, so different feeds can be sent to various members of a band, for example.

In other news, M-Audio have added to their Studiophile range of compact nearfield monitors. There are two new products, both 'Deluxe' versions of existing M-Audio monitors. The BX5a Deluxe and the BX8a Deluxe have better low-frequency extension than their predecessors, as well as improved stereo imaging, thanks to re-designed waveguides.

M-Audio have also announced a line of 'console' pianos and have extended the Keystudio range of USB/MIDI controllers with on-board sounds. The company say that the new products feature innovations from other Avid divisions, including the Advanced Instrument Research group (AIR), whose software instruments and effects have been used inside the new pianos. Finally, M-Audio have launched re-issues of their revered MIDISport 2x2 and 4x4 USB MIDI interfaces to celebrate their 20th anniversary. They'll be available shortly.

Elsewhere inside the Avid group, scoring stalwarts Sibelius (www.sibelius.com) have updated the Student version of their flagship software package. Sibelius Student was, until now, based on Sibelius v3. But with the update, users get some of the benefits of the latest version five. This means, in practice, that users can run their Audio Units and VST virtual instruments inside the software, and use functions such as the Ideas Hub to keep a record of disparate ideas on the fly. Sibelius also showed their software products designed for younger students. The Groovy range, as it's called, gives children graphical stimuli, which helps them to interact with the musical content in the program, and teaches them the fundamentals of scoring, notation and structure.

Finally, Digidesign (www.digidesign.com) have announced updates to their Hybrid and Strike virtual instruments. Version 1.5 of Hybrid includes a new voltage-controlled filter (VCF), which Digidesign say allows the user to "achieve the classic, retro analogue synthesizer sounds of the '70s and '80s". Hybrid 1.5 also has new filter saturation modes, giving the user more flexibility when shaping the tone of their synth patch. Digidesign's virtual drummer software plug-in Strike can now be expanded with a software add-on that brings 300MB of drum machine sounds and 100 new style settings. Both updates will be free to registered Hybrid and Strike users for a limited time.

For more information on all these new products, keep an eye on the web sites of the respective companies.

Novation's Nocturn

A 'turn for the better?

Novation are synonymous with innovative USB MIDI controllers, mainly due to the success of the Automap Universal protocol, which integrates seamlessly with DAW software and allows users to quickly and intuitively access software parameters using their hardware controller. The newest Novation controller equipped with Automap Universal is the Nocturn, a desktop device that looks suspiciously like a small DJ mixer (but don't let that put you off!).

It has eight touch-sensitive rotary encoders, a crossfader and eight assignable buttons, plus a further eight buttons for controlling a DAW's most common features. There's also a handy 'speed dial' rotary encoder that controls whatever the mouse cursor is focused on. The Nocturn is due to ship in February, for the extremely attractive price of around £70.

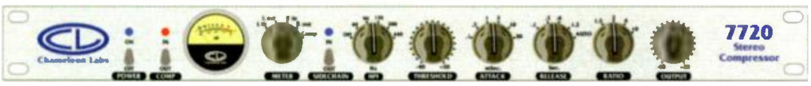
In other news, Novation have updated Automap Universal to version 2. New features include RTAS and TDM support, so compatible products can be used to control instruments and effects within Pro Tools, and there's a new 'heads-up transparent control GUI', which pops up to help you view and map your parameters. When it's released in February, it'll be fully backward-compatible, and will be available to all existing Novation SL and SL Compact owners.

Novation +44 (0)1494 462246 www.novationmusic.com

Chameleon 7720 compressor
Blends into its surroundings

Chameleon Labs are a relatively new company with a line of audio hardware that is, refreshingly, aimed at the budget-conscious user. Already on the market are a handful of products, including two mic preamps (one of which has a three-band EQ) and small- and large-diaphragm microphones. But the newest product from Chameleon is the 7720 stereo compressor, which we actually first saw at AES in New York, back in October 2007.

It uses VCA technology from THAT Corporation (formerly dbx) and features, say the manufacturers, the cleanest signal path possible, thanks to a low component count. Front-panel knobs give the user control over threshold, attack, release, ratio and



make-up gain, as well as the high-pass filter, which has five switched settings from 60Hz to 440Hz. The 7720 has an external side-chain input, which is engaged using a switch, and the compressor circuitry can be bypassed. There's a single meter that can display the input or output of each channel, as well as the overall gain reduction. The 7720 costs £470 including VAT and is available in the UK through Funky Junk, along with other products from Chameleon Labs.

Funky Junk +44 (0)207 609 5479 www.funky-junk.co.uk www.chameleonlabs.com

Peluso competition winners

The lucky winners of the Peluso 2247 SE competition from *SOS* October 2007 are James Hill-Walker from West London and Benjamin Young from Boston, USA. We had over 1000 entries to the competition, in which there were two Peluso mics to give away — one for USA/International edition readers, and one for our UK edition. In total, over £2700 worth of Peluso gear was given away!

We got contacted James (pictured), to get his reaction: "What a great start to 2008. Coincidentally, a couple of days before I discovered I had won this prize, a producer had asked what mics I had and didn't sound very enthused by my answer. He should be much more impressed now."

Benjamin Young, who plans to try his new mic on a violin in an orchestra pit, commented on the great sound of the mic, saying "it's fantastic".

Thanks again to Peluso Microphone Lab for donating the fabulous prizes.

www.pelusomicrophonelab.com

New for old
Newmann tube mic boasts the best of both worlds

After becoming frustrated at spending large sums of money and time purchasing and maintaining vintage microphones, musician and producer Steve Bull decided to design and build his own, using a combination of modern technology and vintage design to realise his goal. The result is the Newmann Retro, a microphone reminiscent of vintage 'lollipop' mics, which Steve claims has "tube warmth, clarity and character", but which costs £495: "less than the cost of a service for a vintage mic".

The Newmann Retro uses modern components and is assembled manually. Each mic is hand-tuned and supplied in a flightcase with a sturdy shockmount and a power supply. The mic has switches on the body to select the polar pattern (omni, figure-of-eight or cardioid), and also to engage a 10dB pad and a high-pass filter. An integral pop-shield adds a classy touch, but is also a very practical addition when the mic is used with vocalists. For further information, head over to Newmann's web site.

Newmann +44 (0)1323 730226
www.newmannretro.com

MXL move to 24-bit

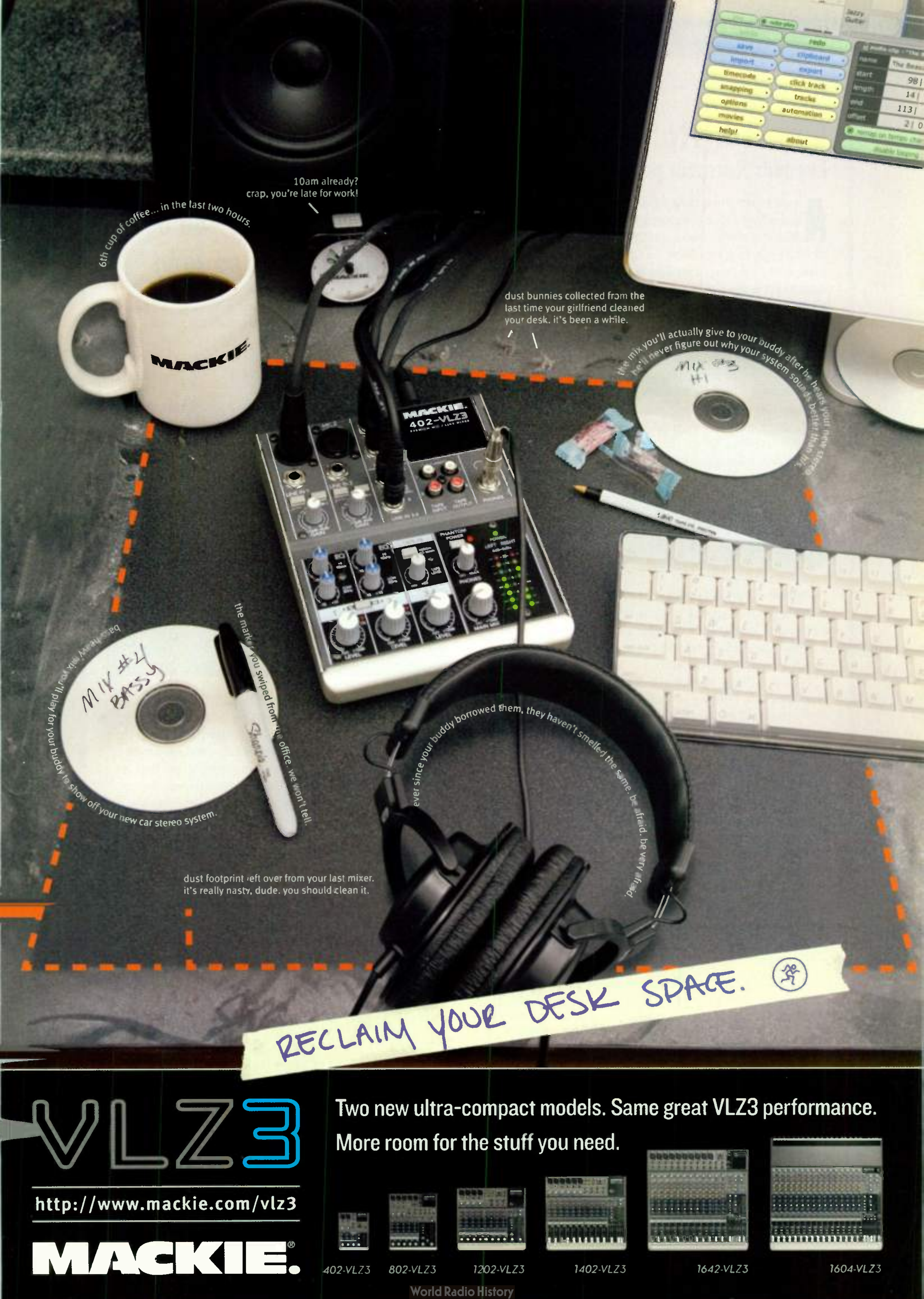
MXL (**www.mxlmics.com**) have extended their range of USB microphones with the USB 009.24, a 24-bit/96kHz version of the 008 created in response to customer demand. Like its older brother, it's a large-diaphragm condenser mic that's powered over USB, and it features a headphone output, allowing the user to monitor recorded audio while singing or playing into the mic. MXL also have a new flagship tube mic, the Genesis. Voiced to provide a richer, more obviously 'vintage' tone than their existing models, it is expected to retail for around $600 in the US, but no UK price is yet available.

Unsigned Heroes '08 gets under way

This year's AKG Unsigned Heroes competition, the annual battle of the bands contest organised by Harman Pro UK, has begun. This time, however, Harman have upped the ante, as the overall winners will be competing for the chance to play live at Glastonbury in June, in association with Strummerville, which aims to promote new music in memory of Clash co-founder Joe Strummer.

There are four heats in the competition, one each month from February to May, with three bands picked to play live in front of an industry panel each month, at the Comedy Café in London. The winner of each heat will progress to the 'vote-off' stage, where they'll face an additional four bands selected from on-line judging. Then the winner of the eight 'vote-off' finalists will be selected, along with second- and third-place runners up, who will play at the Jersey Festival and the Zoothousand Festival in Kent respectively. All three winning bands will receive an AKG Mic Pack and an endorsement deal, with the Glastonbury winners getting one of their tracks remixed by Unkle collaborator Aiden Lavelle, as well as four tickets to Kent's Zoothousand festival.

To enter, you must be 18 or over, and there must be at least three people in your band. You need to register on the Unsigned Heroes web site, and submit an MP3 of an original composition. To sign up and find out more, head to **www.unsignedheroes.co.uk**.



6th cup of coffee... in the last two hours.
10am already? crap, you're late for work!
MACKIE
dust bunnies collected from the last time your girlfriend cleaned your desk. it's been a while.
the mix you'll actually give to your buddy after he hears your new stereo
he'll never figure out why your system sounds better than his.
MIX #3 HI
MACKIE 402-VLZ3
bass-heavy mix you'll play for your buddy to show off your new car stereo system.
MIX #4 BASSY
the marker you swiped from the office. we won't tell.
ever since your buddy borrowed them, they haven't smelled the same. be afraid. be very afraid.
dust footprint left over from your last mixer.
it's really nasty, dude. you should clean it.
RECLAIM YOUR DESK SPACE.
VLZ3
http://www.mackie.com/vlz3
MACKIE.
Two new ultra-compact models. Same great VLZ3 performance.
More room for the stuff you need.
402-VLZ3
802-VLZ3
1202-VLZ3
1402-VLZ3
1642-VLZ3
1604-VLZ3

Direct Install

Expands Receptor potential

At the NAMM show, Muse Research announced a new technology called Direct Install, which allows users to load virtual instruments directly into their Receptor hardware plug-in player, using the CD or DVD drive of a Mac or PC. The Direct Install-equipped Receptor recognises the drive on the computer (connected over a network) and reads the data contained in conventional EXE or Setup files on a normal software installer disc. According to the manufacturers, this "vastly expands the range of software that you can run on the platform".

The company have also announced that their Receptor with Komplete 5 Inside, a hardware plug-in player with the entire contents of Native Instruments' Komplete 5 bundle installed and ready to play, is now available. Customers who purchase the product, which costs $2499, can use the plug-ins on the Receptor straight out of the box, as it's installed and configured by Muse Research before shipping.

The benefit of using a Receptor rather than a laptop computer for such applications, say the company, is that the Receptor has significantly lower latency, and increased stability, mainly due to its lack of superfluous non-music-specific components.

Muse are also are offering upgrades to customers with older Receptors. Different upgrades are available to owners of all versions of the Receptor. 'Rev C' owners will be able to upgrade their Receptor's CPU and hard drive to bring it in line with the latest Pro Receptor, while customers with the older A and B revisions of the Receptor will be able to upgrade to Rev C spec, which will soon be able to accommodate SATA drives, following a forthcoming software update. For full details of all the upgrade paths, check out Muse Research's web site, and click on the link to the Plugorama upgrade shop.

www.museresearch.com

IMSTA make NAMM turn blue

'Buy the software you use' campaign in the spotlight

A 'T-shirt day' set up by the International Music Software Trade Association (IMSTA) saw staff from software companies and supporting organisations wearing blue T-shirts (sponsored by Microsoft) emblazoned with the IMSTA logo, on the Saturday of the NAMM show. The stunt was

designed to publicise the 'Buy The Software You Use' campaign, which aims to cut the piracy of software through education.

IMSTA were set up in 2002 by Ray Williams, with the intention to support software manufacturers and encourage software users to acquire their software legitimately. They are a not-for-profit organisation, funded by subscriptions from member companies. One such company is Propellerhead, who posed for *SOS* before the start of the show on 'T-shirt day' (see above). For more information, and to support IMSTA, head to their web site.

www.imsta.org

CAD announce stereo USB mic

CAD (**www.cadmics.com**) were previewing their first stereo USB microphone at NAMM. The u2205T is an X/Y condenser that ships with a shockmount, table-top stand and Cubase LE recording software. It's expected to cost around $200 in the USA (there's no UK price yet), making it ideal for musicians on a budget. Given CAD's reputation, it'll probably be money well spent.

More gamelan!

If someone were to ask me which company were most likely to release a 24GB multisampled Balinese gamelan library, the words Sonic Couture would spring immediately to mind. In the past, they've released such titles as Ebow Guitars, Hangdrum and Kim, so the collection devoted to the range of Balinese instruments bolsters their slightly left-of-centre range accordingly. The library comprises 25 different instruments (a traditional gamelan orchestra consists of pot-like and free-standing gongs, metallophones and drums), all sampled at 24-bit/96kHz with up to 20 velocity layers. In total, there are over 4000 samples, and Kontakt patches with key-switches that provide multiple articulations. For full details of the sample library, which will retail for £299, head to **www.soniccouture-gamelan.com**.

Rode Mics in web site frenzy!

Despite there being no new product announcements from Rode at the NAMM show, the Australian mic manufacturers were pushing their latest venture, Rode University, an on-line education program. It aims to "explain the basics of microphone recording in a relaxed and exciting format", through on-line video tutorials and interactive tests. Upon completion of the course, students will get a graduation certificate, which Rode say will be a worthy addition to their "rock resume". Check out **www.rodeuniversity.com** for more info.

Staying on the Rode microphones and web site theme, the company have re-designed and re-launched their existing company web site. Check out **www.rodemic.com** to see what's new.


YAMAHA
n12
CUBASE READY
CONTROL ROOM
CONTROL ROOM LEVEL
TALK BACK
STEREO
DAW REMOTE CONTROL
WORK MODE
MONITOR REMOTE
CLICK REMOTE
TRACK CONTROL
TRANSPORT
REC

Portable Forte

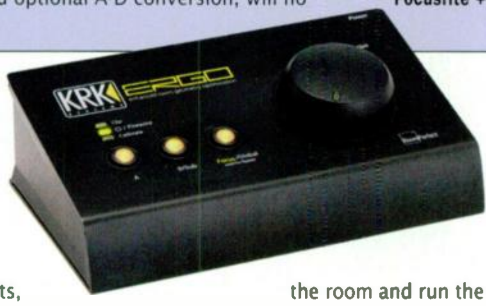
Focusrite's ISA One

UK hardware manufacturers Focusrite have expanded their ISA range with a new single-channel preamp, the ISA One. Based on technology from the ISA110 module, originally fitted to the legendary Forte mixing console, the ISA One is a solid-state input stage with instrument, mic and line inputs. Its neat form — a compact chassis with a handle on the top — makes the ISA One convenient for users who want a good-quality, portable preamp. But its additional features, including a stereo cue input that feeds the on-board headphone amp, and optional A-D conversion, will no doubt appeal to project-studio owners and larger commercial facilities alike.

For further details, visit Focusrite's web site, and check out the news pages on the *SOS* web site for a video demonstration.

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

KRK introduce room-correction system



As the inexorable rise of the project studio continues, more and more monitor manufacturers are acknowledging that their products are often used in less than ideal acoustic environments, and employing advanced computer technology to help out. The NAMM show saw KRK launch their Ergo digital room-correction system, which consists of a desktop DSP unit and a measurement mic. The DSP unit sits between your DAW and your monitor speakers, and can act as a basic monitor controller thanks to its large volume wheel and ability to switch between 'A' and 'B' speaker outputs. It also acts as a Firewire audio interface, allowing measurements to be taken from the calibration mic.

To set up Ergo, you locate the mic at a number of random positions within the room and run the measurement utility on your Mac or PC. This builds up a three-dimensional picture of the room's response, and allows it to calculate the frequencies at which problematic resonances reside. This information is then transferred to the Ergo unit's hardware DSP, which can apply up to 1024 narrow EQ bands to fine-tune the monitor response to the features of your room. For more information, should you need it, head to KRK's special Ergo web site, www.krkergo.com.

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

Embrace V-Link!

Roland encourage fellow manufacturers

At their NAMM 'press and technology conference', Roland sent an open invitation to competing manufacturers to embrace the V-Link protocol, which enables audio and video equipment to communicate using the basis of the standard MIDI specification. Roland likened the potential power of V-Link to that of MIDI, and encouraged other manufacturers to consider the future of the industry.

During the conference, they demonstrated various implementations of V-Link technology, with the new Fantom G8 (see page 6) and V-Synth GT generating and manipulating moving and still images, the SP555 triggering video samples, and an Edirol vision mixer being used to control both video and audio, by communicating with the new Roland V-Mixing system.

If Roland's competitors embrace V-Link, we could be in for some interesting new product announcements in the coming years, so keep your eyes on the pages of *SOS*...

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

Pro Tools control for Eventide processors

Eventide have announced a new RTAS plug-in called E-control, which provides real-time control over the company's H8000, H7600 and Eclipse hardware processors. Parameters inside the plug-in can be automated using Pro Tools, so that settings and alterations can be quickly recalled, without having to make manual adjustments to the hardware. It will become available at the end of April, costing $199 (around £100), although it's free if you purchased your product after September 1st, 2007. Visit the newly designed **www.eventide.com** to download the plug-in.

If you're an Eclipse owner, you can also download a free update for your hardware, which accelerates the operating system and doubles the number of parameters that can be controlled at one time.

Arriving too late to be included in our Time Factor review (page 136) is news that owners of the Time Factor and Twin Delay stomp boxes can download new software to enable extra features in their pedals. One of these, Spill Over, provides smooth switching transitions between presets.

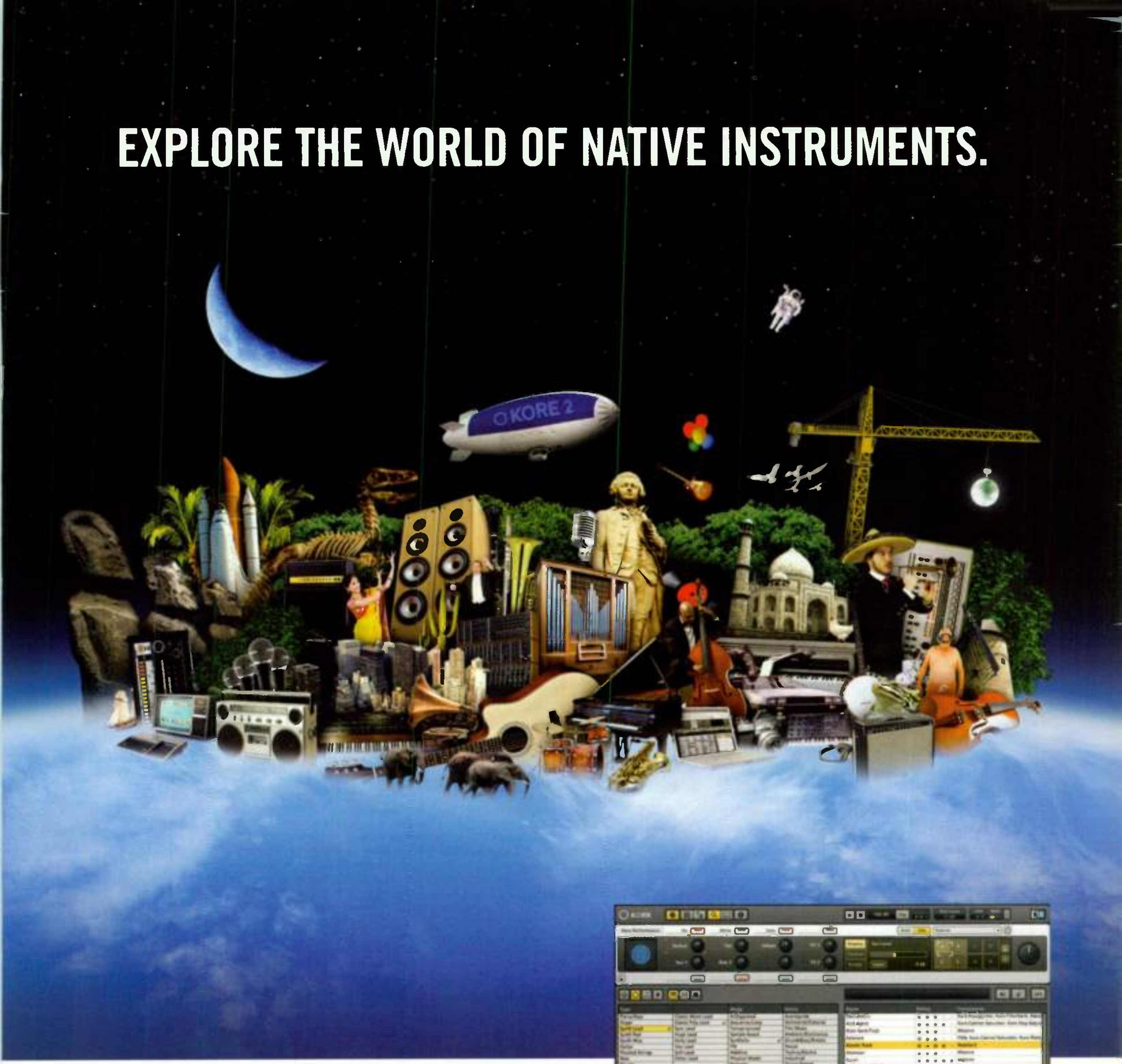
EP add-on from Modartt

French virtual instrument designers Modartt have announced a new update for Pianoteq (**www.pianoteq.com**), their physical-modelling software piano, which now includes an electric piano model. Modelled on the Yamaha CP80 (serial number 1982, in case you wondered), the new model in the Pianoteq line-up now gives the user a choice of acoustic and electro-acoustic pianos.

Two different variants of the CP80 come in version 2.2.1 of Pianoteq: an 'original', slightly beaten-up instrument, and a 'restored' version, which was modelled after the instrument was fitted with brand new strings.

Also new in the update are additional settings for a 'flat temperament', which Modartt say should help Pianoteq "fit" better alongside most synthesizers, particularly those with mathematically perfect octave separation.

Best of all, version 2.2.1 is free to all registered Pianoteq customers. Simply visit Modartt's web site and download the update.


EXPLORE THE WORLD OF NATIVE INSTRUMENTS.
KORE 2

TC Electronic's big NAMM news

The really big news from TC Electronic at NAMM was that the Danish company were in the advanced stage of talks to merge with guitar giants Gibson, with a completion date due sometime in February 2008. It's too early to say how this will affect the product line or TC's independence at this stage, so we won't speculate further!

On the product front, TC launched two new guitar pedals, the Nova Modulator and Nova Dynamics. Nova Modulator combines all existing TC modulation effects with two brand-new effects to offer chorus, a new tri-chorus, flanger, a new through-zero flanger, phaser, tremolo, and an upgraded vibrato. Its dual-engine design allows combination effects to be configured.

Nova Dynamics features studio and stomp compression modes (the latter of which is more aggressive), as well as a noise gate. A dry blend feature enables the user to mix in the dry signal along with the compressed signal for parallel compression effects.

Falling somewhere between the pedals and the high-end G-System, TC's new Nova System floor-based unit comes with programmable analogue distortion and overdrive, augmented by TC's usual array of high-quality digital effects. Nova System incorporates six effects blocks taken from the G-System: compression, EQ, noise gate, modulation, pitch, delay and reverb, all programmable and storable in 60 user presets alongside the 30 factory presets.

The Konnekt 6 joins TC Electronic's existing Firewire audio interfaces and includes a zero-latency monitoring control section, a tracking reverb and a three-mode high-resolution meter. The unit handles two simultaneous analogue inputs, one of which is an XLR input featuring TC's IMPACT mic preamp. There are two quarter-inch instrument-level inputs and two monitor outputs, as well as a stereo headphone out.

Powercore fans might be interested to know that Powercore X8 is also now shipping and features eight DSP engines, doubling up on the processing power of the original Powercore Firewire. Powercore X8 ships with 14 plug-ins as well as Powercore 3.0 system software and a $500 plug-in voucher that can be used as part payment for further effects.

At the NAMM show TC also took the opportunity to announce their Loudness Radar Meter Plug-ins for Pro Tools HD. These plug-ins display statistical information according to the ITU-R BS1770 standard to show instant loudness and loudness history. In the radar view, LM5 (below left) displays real-time loudness (LU or LFS) and true peak level. LM5D (below right) is an extended version of LM5 and includes two long-term universal descriptors: centre of gravity and consistency.

TC Electronic UK +44 (0)800 917 8926
www.tcelectronic.com

Roland acquire majority share in Cakewalk

The Roland Corporation (**www.roland.co.uk**) have, since 2003, held a minority share in Cakewalk (**www.cakewalk.com**), developers of Sonar and numerous virtual instruments. But just before the NAMM show, it was announced that Roland have acquired a 60 percent share in the company. Cakewalk are clear that they, and their current product line, as well as the existing support infrastructure, will remain intact. The company also say that "a series of impressive software/hardware products" are to be jointly developed by the two companies, and in a Roland press conference, the words "Cakewalk by Roland" were uttered more than once. Cakewalk have told their customers that they are "not becoming a 'division' of Roland", and that they "remain committed to developing stand-alone software, as well as hardware/software products". So watch out for new, joint-developed products from the two companies.

Rack 'em up!

Swedish sample developers Power FX have unveiled World Loop Spice Rack, an interesting collection of ethnic percussive loops and samples. Instruments from Africa, China, Australia, India and the Middle East have been recorded at 16-bit/44.1kHz, and hand-played loops have been turned into Acidised WAVs. In total, there is almost 900MB of content from over 50 instruments. Visit **www.powerfx.com** to purchase the sample library, and for further details. It costs £36.

ATB24 price comparison

In the Toft Audio ATB24 review in last month's issue of *Sound On Sound*, we referred to a competing product as the "Oram 8T-24", when the unit in question, whilst designed by John Oram, is manufactured and sold by Trident Audio Ltd as the Trident Series 8T-24. We also stated that the Trident Series 8T-24 was "slightly more expensive" than the Toft Audio ATB24. In fact, the Trident Series 8T-24 retails for £3524 including VAT, compared to the ATB24's retail price of £3878 including VAT. We apologise for any confusion that this may have caused.
www.oram.co.uk
www.toftaudio.com

Surgin' USA

Alesis storm the market

Alesis presented an impressive range of new products at the NAMM show, including two new electronic drum kits. Both the DM5 Pro and the USB Pro are equipped with real drum heads (albeit foam-dampened), and what they're calling Surge cymbals. These are essentially real metal cymbals with a plastic damping layer and touch pad on the underside which can be used for muting, or 'catching' the cymbals.

The DM5 Pro kit has much improved hardware over earlier versions but still uses the old faithful DM5 module, which has 540 sounds with 21 programmable drum kits, as the sound source. The kit comprises a dual-zone snare, bass drum pad, three tom pads with rim triggers, a 12-inch electronic hi-hat, a 13-inch crash cymbal, and a 16-inch dual-zone ride cymbal, all mounted on a rack-style stand. The DM5 can also function as a stand-alone trigger-to-MIDI converter, and features 12 trigger inputs.

For the studio user, Alesis' USB Pro kit offers similar hardware but with an Alesis Trigger IO instead of the DM5, and a bundled 'light' version of FXpansion's BFD virtual drummer software. The hardware offers 10 TRS inputs for single or dual-zone triggers, 20 programmable presets and a hi-hat pedal input for on/off or continuous control of the hi-hat sound. The hardware connects to a computer (both PCs and Macs are supported) using USB, and it also has a MIDI output.

Those who prefer to program their drum parts might like to take a closer look at the new SR18, which improves on the still-current SR16 with a much larger sound set (32MB), an on-board bass synth, and built-in effects. It covers both traditional and modern drum sounds and can be run from batteries or a mains adaptor. The SR18 boasts 24-voice polyphony, an instrument input and a library of 175 presets, with space for 100 user patterns. Footswitch control is available for starting and stopping playback and recording, and for controlling the count-in and fill functions.

Alesis were one of the first companies to produce gear designed to team up with Apple's iPod, and their new Multiport turns your iPod into a stand-alone desktop stereo recorder that has USB connectivity for hooking up to a computer. It works with all docking iPod models when used through iTunes, and is able to play back one piece of audio while recording another. The unit offers both mic and line inputs, as well as level metering and decent-sized controls.

Also new from Alesis is the iMultimix 16 USB, which might suit you if you're wanting to use your iPod to record gigs or provide backing tracks. It is based around a tabletop mixer with an integral iPod dock and features 100 digital effects, a built-in peak limiter, and transport controls. Four of the 16 inputs are mic/line preamps with 48V phantom power and there's a three-band EQ on each channel.

However, if you'd rather record to a computer than an iPod, Alesis also have the Multimix 8 and Multimix 16 USB mixers, which can stream multiple audio channels into your Mac or PC. The Multimix 8 USB 2.0 has 10 direct outputs for recording and two return inputs for monitoring. For processing there are 100 on-board effects, including reverbs, delays, chorus, flanging, pitch and multi-effects, though if you need something a little larger, the Multimix 16 will give you 18 direct outputs for recording and two return inputs for monitoring. Both models offer 24-bit/96kHz operation and are compatible with Mac OS X and Windows XP.

The USB format must be contagious in the Alesis paddock, as even the diminutive M1 Active 320 USB monitors are able to plug directly into a computer. They operate as a USB audio interface and feature a headphone output, an eighth-inch stereo mini-jack input and magnetic shielding. Each one has a three-inch woofer and a one-inch silk-dome tweeter, covering frequencies from 58Hz to 25kHz. Prices for the new gear were still unannounced at the time of writing.

Numark +44 (0)1252 341400
www.alesis.com

Sonnox SuprEsser New addition to the Oxford Plug-ins family

Plug-in manufacturer Sonnox (formerly part of the Sony Oxford group) have added to their range of RTAS, Audio Units and VST plug-ins. SuprEsser is primarily a de-esser, but can be used to eradicate other unwanted noises in an audio track, such as plosives or low-frequency rumbles.

At its heart is a responsive 1024-tap FFT display, which provides a detailed overview of the audio spectrum, but can be 'zoomed' to show a band of frequencies in more detail. Superimposed over the display is a red line that shows where the plug-in thinks the dominant frequency is between two selected frequencies, which are determined by the band boundary lines — also superimposed on the FFT display. Inside the band boundaries is an intuitive, upside-down, triangle-shaped gain-reduction meter, which gives real-time visual feedback on the status of the plug-in. In usual Sonnox style, there's detailed metering for input and output levels.

SuprEsser is due to ship in March, at a cost of £188 including VAT, and discounts are available when purchasing multiple Oxford-branded plug-ins.

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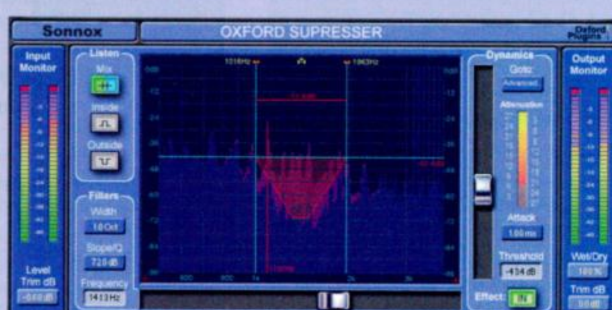
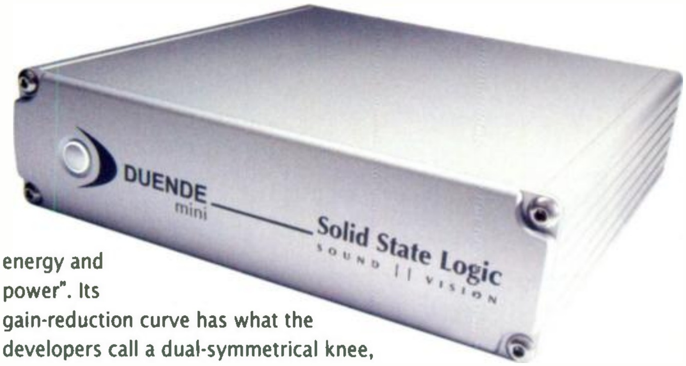
www.sonnoxplugins.com



Duende Mini!

Portable DSP processor packs a punch



In just under two years, SSL's Duende DSP platform has become one of the most respected on the market. But so far, there's only been one way to embrace it: by purchasing the rackmountable Duende processor. Now, however, there's a new, more affordable Duende that can be upgraded to pack the power of its bigger, older brother. The Duende Mini is a one-third rack-width desktop device that connects to a computer via Firewire. It can be bus-powered or plugged into the mains, and as standard it can power 16 channels of plug-ins. The Duende Mini ships with the SSL EQ and Dynamics plug-in, but can, like the Duende Standard (as it's now called), be upgraded with additional

SSL plug-ins. Further details can be found at SSL's web site.

The UK-based company have also released more plug-ins. The first, X-Comp, is a stereo compressor, and one that SSL claim is so versatile that it can be used for "everything from 'invisible' subtle dynamic control for mastering, to dramatic 'brick wall' effects that inject raw energy and power". Its gain-reduction curve has what the developers call a dual-symmetrical knee, which enables the user to shape the characteristic of the compressor to suit their sound, using an intuitive GUI. A built-in side-chain EQ can be configured intuitively using a logarithmic frequency plot, and a wealth of metering options give the user feedback on the activity of the plug-in. There's stereo input and output bargraph metering showing both peak and RMS levels simultaneously, an amplitude histogram that displays real-time pre- and post-compression signals, and a gain-reduction history.

Another new plug-in is X-EQ, also for the Duende platform. It's a parametric EQ with 10 bands, each of which has a variety of response curves and can operate from 20Hz to 20kHz. X-EQ has an FFT frequency analyser, an A/B comparison feature and comprehensive metering sections, as with X-Comp. You can also control the parameters using MIDI. Both new plug-ins will be available in the first quarter of 2008.

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

New virtual instruments join the Big Fish Audio shoal

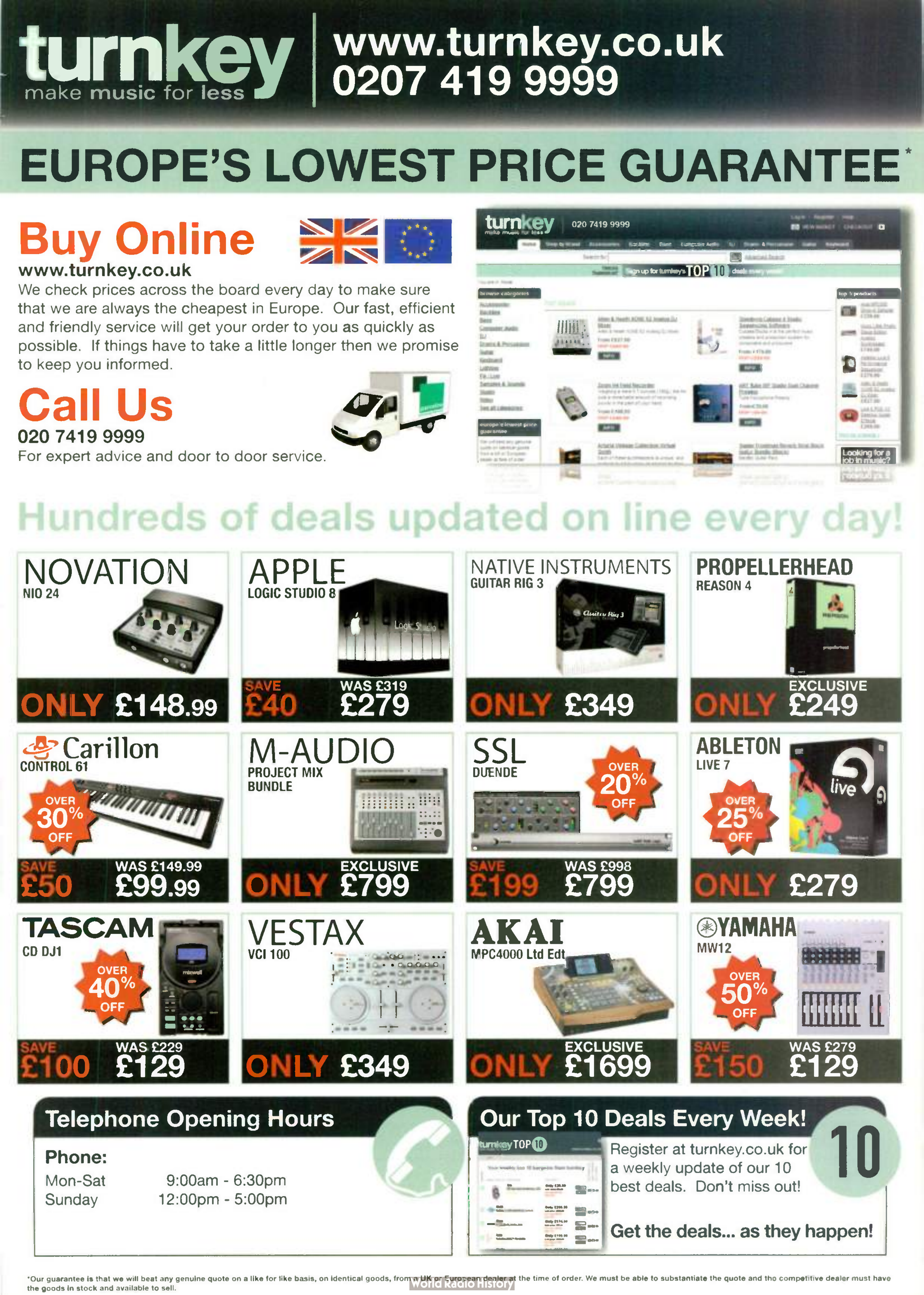
Big Fish Audio had plenty to announce at NAMM, with the headlines going to three new Kontakt-based virtual instruments from Vir2. Elite Orchestral Percussion, which lists at $399 in the US, is a 19GB library stuffed to the gills with, well, orchestral percussion. Over 250 instruments were sampled, from the obvious timpani and snare drums to handbells, wind chimes and woodblocks, and advanced scripting techniques have been used to make the results as playable and realistic as possible. Articulations such as rolls, flams and chokes can be programmed very easily in the instruments' Performance view, and a range of EZRoom convolution impulses is available to treat the output.

BASiS, meanwhile, is devoted to the lower reaches of the audio spectrum, collecting together 7GB worth of electric, upright and synth basses. All the electric basses were sampled both via a DI and an amp, and the two signals can be mixed to taste; likewise, acoustic basses were recorded using both a pickup and a mic. From slaps and slides to harmonics and fret noises, if the bass can make a sound, it's included here. There's a comprehensive range of bass-oriented effects, and a custom legato tool allows realistic 'finger-smooth' playing. BASiS will cost $199 in the US, which should give UK and European readers an idea of local prices.

Next off the block will be the Mojo instrument, which aims to provide a complete package for those seeking to recreate classic pop, rock and funk horn sections. It covers all the saxes, along with trumpets, trombones, flugelhorns and clarinets, and, once again, the range of articulations on offer looks to be exhaustive. Swells can be sync'ed to host tempo, and a single control allows you to alter the number of musicians in your virtual ensemble. Legato and vibrato are handled in a flexible and intuitive way, and if the audio demos we heard were anything to go by, this library will be one to look out for. Pricing for Mojo was unconfirmed at the time of writing.

There's no let-up in the company's programme of loop-based releases, either, with new products ranging from Latin Jazz, World Percussion and Found Percussion to urban titles such as Straight Outta NYC, Plush and Hip Hop Mood Swings. Fresh!

www.bigfishaudio.com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Q&A

Your technical questions and queries answered

Q What's the best way to get audio into my portable recorder?

I am looking for a more suitable solution for the stereo recording setup which I currently use to record classical music and jazz. I have a pair of Sennheiser MKH800 microphones, and I want to record to Compact Flash card. I currently use the mic preamps in my Yamaha i88x, but I have to use an unbalanced line signal from the insert point using a cheap jack-to-RCA adaptor, to plug it into my Tascam HDP2 stereo recorder.

If I could utilise the S/PDIF from the i88x then I would have no problem here, but the S/PDIF input and output on the i88x is mLAN exclusive, so you can't get the direct mic preamp signal from there. You can only route it to the unbalanced insert point or the monitor output.

I am after the best possible quality in terms of A-D conversion and mic preamps for around £1000. I would also like to convert from analogue to digital as early in the signal chain as possible, so ideally I'm looking for a high-end, two-channel mic preamp with an S/PDIF output. Alternatively if there are any other rackmount Compact Flash products in the pipeline with AES input, then a mic pre with an AES output would be even better.

I am also keen, despite scepticism from others, to record at 88.2kHz rather than the now almost standard 96kHz as I believe that, as the final destination is audio CD, then the algorithmic processing involved in bouncing down to 44.1kHz is much simpler and therefore likely to be more accurate. I have proposed this to various industry contacts and had very mixed responses. What's your take?
Ian Simpson

Technical Editor Hugh Robjohns replies: The first and simplest option, which I presume you have already considered and discounted, is to use the mic preamps in the Tascam machine itself. These aren't bad for a machine of this type, and they can provide phantom power for your Sennheiser mics. That would obviously be the lowest-cost solution, although not the absolute best quality route.

After that, it really depends on how much of your budget you want to spend. There are numerous two-channel mic preamps around with built-in A-D conversion. At the high end (but stretching your £1000 budget) you could look at the Neve 1073DPD, for example. But being a little more realistic, I would suggest the Audio & Design DMA2 (reviewed in *SOS* January 2008, and on-line at www.soundonsound.com/sos/jan08/articles/audioanddesigndma2.htm). This is an extremely neutral and accurate preamp, with resettable digital controls, excellent A-D converters, and a really well thought-out feature set. Best of all, it is astonishingly well priced and will meet your needs perfectly.

The output meets, technically, the AES3 standard, but you won't have any problems coupling it to the Tascam's S/PDIF input provided you keep the cable reasonably short and wire the RCA connector between pins 2+3 (use twin-core screened cable, but leave the screen connected to pin one at the XLR end, and isolated at the RCA end). If you want to be technically pedantic, you should use a 'balun' to convert between the AES3 output and the S/PDIF input — and appropriate in-line adaptors are available from suppliers like Canford Audio and Studiospares — but I have found that they are rarely required in practice.

Regarding your desire to record at 88.2kHz rather than 96kHz, the DMA2 can operate at either sample rate, so you'll have no problem there, although in fact the 88.2kHz mode wasn't included in the original design and was added later following user feedback (which explains the slightly odd sample-rate indicator arrangement!)

The idea that integer sample-rate conversion is more accurate than non-integer sample-rate conversion — which is what you are assuming by suggesting that 88.2kHz is easier to convert to 44.1kHz than 96kHz — is no longer true. The best of the modern sample-rate converter (SRC) systems perform astonishingly well regardless of the input and output sample rates, because they calculate the precise sample amplitude required at the precise time it is needed. Older (and we are talking 15 years ago) oversampling-based systems did favour integer-related sample-conversion rates, but the technology has moved on a very long way since then. Indeed, the current crop of single-chip SRCs easily out-perform the best D-A converters in terms of noise and distortion.

The Audio & Design DMA2 (reviewed in *SOS* January 2008) is a firm favourite among the currently available two-channel mic preamps. What's more, it's got an AES output, so it can be connected to Ian Simpson's Tascam HDP2 stereo recorder.

Q Why don't my synth basses sound good on a hi-fi?

I'm mixing a project which has a number of synth parts that sound great in my studio, but I've just been listening to some rough mixes on my home hi-fi system and things are not good. Firstly, the sounds go very low and are quite bass-heavy, and they distort my normal hi-fi speakers despite coming over great on my studio monitors. Also, some of the instruments just do not sound like they do in the studio. I'd go so far to say that lumps of

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The Fostex D160 (left), reviewed in *SOS* December 1997, was a powerful machine in its time. These days, Fostex have the D2424LV (right) which, as you can see from these pictures, shares more than a striking resemblance to its older sibling.

obviously need to have analogue inputs for my mixing desk as I do not want to replace anything else.
Paul Hazel

Features Editor Sam Inglis replies:
I would imagine the most likely cause of your problem is a dead hard drive. It sounds as though the D160 itself is functioning, at least well enough to tell you that there's no hard drive! It is fairly well-known that hard drives tend to fail if stored for long periods of time without being 'spun up' occasionally. The first thing I'd suggest would thus be to track down a suitable replacement and install it. You may need to consult Fostex as to what sort of hard drive should be used, and perhaps to get hold of a replacement — given that the machine is 10 years old, it may not be compatible with all modern IDE drives.

Fostex are distributed in the UK by SCV London (www.scvlondon.co.uk), who should hopefully be able to help you with your old machine. But if it's unsalvageable, they'll also be able to suggest a modern Fostex machine to replace your D160. They may well suggest the D2424LV, a very well-equipped 24-track, rackmountable digital multitracker, or the MR16HD, a more budget-friendly desktop device. Of course, alternatives from other manufacturers (including Mackie, Tascam, Marantz and Zoom) are available, and it would be wise to shop around — and check out some *SOS* reviews — before taking the plunge and buying a new bit of gear.

Q How do you know what to cut?

Thanks for featuring my band Imprint in the Mix Rescue article of *SOS* January 2008; we were really happy with the results that Mike Senior achieved. But whenever I read such articles, there are always mentions of very specific EQ cuts or boosts at very specific frequencies. In Mike's write-up on our mix, he says "drum harmonics were dominating at the low end — cuts of 3-4dB at 47Hz, 59Hz, 96Hz and 174Hz were all that were needed". How are these frequencies arrived at? Is it just a case of knowing the problem frequencies, is it with some sort of spectrum analyser, or it is trial and error? Also, how are the cuts done — is it just using a multi-band EQ?

I think this is the weakest part of my mixing side, and is probably the part I want to work on next, but I am not really sure where to start.
Blink (Imprint)

***SOS* contributor Mike Senior replies:**
Drum samples, such as the ones in use in your track, don't usually change pitch, so you can use very narrow EQ cuts to home in on individual harmonics of the sound, and this is great for dealing with unwanted resonances in particular: a ringing sound at a particular frequency, for example. The problem with over-prominent individual harmonics is that they stop you fading up all the other harmonics of the sound far enough in the mix. By the time the sound as a whole is audible enough, those little resonant frequencies poke out too far. Very narrow EQ cuts make very little difference to the rest of the sound (the narrower the better really), so they're one of my favourite tactics when I can get away with them. They won't work on melodic instruments, though, because the harmonics move around as the pitch changes.

In terms of which EQ to use, it doesn't matter at all, as long as you have a Q/resonance/bandwidth control and you can accurately enter frequencies — particularly at the low end, where 1Hz can mean a noticeable difference in pitch.

From memory, in your particular case I seem to remember that the kick part had two pitches to it, and I felt that both had low harmonics to them which were a little too prominent. This made them boom and hang on too long for my liking, especially as I knew that the effects and everything else would need space to move in. One of the pitches was more problematic than the other, hence the rough harmonic relations between some of the cuts. Still, the cuts weren't particularly severe. I normally find myself cutting more severely when using this technique. Here these notches were effectively just 'pulling down the fader a bit' for those individual kick harmonics without affecting the level of the kick as a whole.

The way I found those particular frequencies is by using a fairly well-known trick. Take a peaking filter and ratchet up the resonance (or Q value) to maximum. If you then apply a big gain boost with that filter, it acts as a kind of audio magnifying glass, picking out very narrow frequency bands and even individual harmonics of a pitched sound. If you sweep it around the frequency range for a while you'll soon get an idea of where the problem frequencies lie, at which point you can reverse the gain control setting to cut instead of boost.

You do need to be careful with this approach, though, because it's easy to be tricked into hearing problems that aren't actually there. What I do as a reality check is, once I've finished setting up the cut in question, I bypass it for a few seconds, and

listen for the thing that I don't like. Only when I have it pinned down in my head do I re-engage the filter. That way I know for sure whether my EQ setting is doing the job I initially wanted it to. Funnily enough, I have a vague recollection that I tricked myself a couple of times with that very kick sound (maybe by initially setting the harmonic relations by eye rather than by ear), and I did come back to it and tweak it a couple of times as things progressed. The exact cuts at the two lowest frequencies in particular had to be quite finely judged — if you follow the fader analogy I made above, it makes sense that you'd need to adjust them as finely as you would any other fader.

My bottom line with EQ is that if I can't find a main fader level that I'm happy with for a given instrument, it's often because I actually need more than one fader for different frequency regions of that instrument. Using EQ makes a lot more sense if you think of it as just giving you these extra faders. In some senses, a graphic equaliser makes this easiest to visualise, but a parametric EQ will probably sound better, and it will offer more accurate control, especially for things such as killing pitched resonances. So I usually use a parametric. SOS

Sound Advice

#9 Recording brass in the home studio

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.

Paul: The most common brass instruments you'll find recorded in home studios are trumpets, trombones and saxophones (the sax is technically a woodwind instrument, as it has a reed, but is so commonly found in 'brass' or 'horn' sections). Recording brass is something I do relatively infrequently but it has never really presented any serious problems. As long as you make some effort to get the mic where it sounds best, and make the player aware that if they move around, the timbre of the sound will change dramatically due to the extreme directionality of brass instruments, the rest is pretty easy!

Hugh: In the case of trumpets and trombones, the majority of sound comes from and projects in the direction of the instrument's bell. The 'polar pattern' of the instrument becomes increasingly directional with rising frequency. If you stand behind a trumpet you'll hear very little direct sound and no HF components at all. If you move to the side you'll pick up the lowest frequency components, but it is only directly in front that you'll hear the higher frequency components and harmonics (above about 4kHz).

Paul: So for the sharpest, crispest sound you'll need to place a mic directly in front of the bell – either on a fixed stand or clipped directly to the instrument itself. If you're concerned about room reflections, then some suspended duvets behind and to either side of the player will help damp things down. As is often the case, if the room doesn't flatter the instrument, then it is better to damp out as much of the room as you can and then replace it with a more sympathetic reverb when you come to mix.

Hugh: You need to bear in mind that brass instruments are designed to be loud. Trumpets have been measured at four metres to produce over 96dB SPL, and around 130dB SPL just 0.5m from the bell. A trombone is roughly 5dB louder still. So microphone choice means finding a microphone that can accommodate huge peak levels without excessive distortion.

Paul: But every cloud has a silver lining, so they say, and the high SPLs of brass instruments mean that even noisy computers in the same room are so far below the levels of a typical brass instrument, to the extent that, unless you have the mic set up right next to the computer, noise isn't going to be a problem. Again, some acoustic treatment may be necessary, though, because while you can drown out background noise, you can't drown out sound reflections — they get louder as the instrument does!

Hugh: When it comes to mic choice, a typical studio approach for pop music would be to use classic large-diaphragm condensers such as the Neumann U87 or AKG C414, operating with pads switched in and placed within half a metre of the bell. In practice, most large-diaphragm mics will work fine provided they can cope with the peak SPLs. Failing that, dynamic mics are always a reliable choice that will cope with the SPLs without trouble. Ribbons are also popular, both for pop and classical brass recordings. The Coles 4038 and AEA R44 are favourites, but you need to place them no closer than a metre and use a pop screen to prevent wind blasts popping the diaphragm.

Paul: Unless you have headphones with very good isolation, you may not be able to hear the effect of mic movement when working 'live', so recording a test section while moving the mic and describing the mic positions into the mic as you do it may work better. Further to what Hugh suggested about choosing a mic that can handle high SPLs, I suggest that you leave adequate headroom at your mic preamp and DAW input, as the quirky waveforms produced by some wind instruments can produce misleading meter readings on some less-sophisticated metering systems.

Hugh: The saxophone needs to be treated a little differently, because it generates sound in the same way as that conventional woodwind instruments do: along the full length of its body. This means that a different miking technique is appropriate: the mic should be placed to 'hear' the whole body of the instrument. However, in pop music, it has long been fashionable to mic tenor and baritone saxes very close to their upturned bells (as in the picture), and this captures a very specific kind of sound. It's not the natural sound of the instrument as heard in the room at a distance, but a bright, raspy sound that everyone recognises and now expects.

Paul: Some final advice: when recording, pay attention to the sound quality. It changes as the instrument (and player) warm up, and as the instrument inevitably fills with spit! It pays to give the player plenty of opportunity to clear the instrument and rest the lips, particularly if the playing involves a lot of stabs and high notes. Brass players can tire quickly — these aren't easy instruments to play — so be realistic in how much recording and overdubbing you can do.

Focusrite Liquid 4 Pre

Four-channel Convolution Preamp

The manufacturers proudly claim that this is "the most flexible four-channel preamp in history." And it's no idle marketing boast...

Hugh Robjohns

Focusrite introduced the original Liquid Channel — a channel strip comprising a convolution-based preamp and dynamics section, and a digital equaliser — back in 2004. It was a revolutionary product, combining the powers of Sintefex's dynamic convolution technology with Focusrite's ability to design fabulously transparent and versatile analogue preamp circuitry. You can read my thoughts on that product in *SOS* December 2003 (preview) and July 2004 (review): both are available from the *Sound On Sound* web site at www.soundonsound.com.

Following on from that groundbreaking product, Focusrite launched the Liquid Mix 32-channel equaliser and dynamics processor — again based on the dynamic convolution idea. The latest in the line, the Liquid 4 Pre, was introduced at AES New York at the end of 2007, and it revisits the Liquid Channel's core technology to deliver not one, but four channels of convolution-based preamplification in a single box. It is a kind of cross-fertilisation between the classic ISA428 four-channel preamp and the Liquid Channel — in fact, it is even styled to resemble the 428 in some ways. Focusrite claim that this new preamp is "the most flexible four-channel preamp in history," and although I don't go much for the hyperbole they could have a point!

Overview

First, lets clear up what the Liquid 4 Pre isn't. It is *not* four Liquid Channels in the same box — there are no dynamics or equalisation facilities here. What is on offer is four mic preamps, plain and simple, with the bonus that the preamps can be individually configured to behave like pretty much any classic or vintage preamp you care to think of, thanks to the Liquid Channel's core technology.

The original Liquid Channel's preamp stage was the first of its kind and extraordinarily complex. It involved a lot of sophisticated switching of inductors, capacitors, resistors and even a transformer, to radically alter the impedance and other electrical characteristics of the input stage. The aim was to change the input stage so that, from the microphone's point of view, the input circuitry appeared to be exactly the same as the actual preamp being emulated. The reason is that the performance of most microphones is affected in significant ways by the characteristics of the input it is feeding, and this interaction is a vital part of any preamp's sound character. Furthermore, this interaction cannot be emulated easily using convolution — it has to be at electronic

component level in the analogue front end!

One problem when designing a multi-channel version is that this preamp circuit's complexity meant it took up a lot of space — almost half the available floor area of the Liquid Channel's rack-mounting case. So getting four of these analogue sections into a 2U case would have been very difficult, not to mention extremely expensive. Fortunately, technology moves on, and the Focusrite boffins have spent the last three years locked away in darkened rooms trying to find a way of reducing the size and complexity of the preamp circuitry without changing the fundamental way in which it works, its flexibility or, most importantly, its performance.

The launch of the Liquid 4 Pre is the evidence that they have achieved their goal, and a squint inside the box reveals a far more compact (but still very busy and complex) pair of circuit boards, one mounted above the other. The most obvious circuit change is that the large transformer used in the Liquid Channel has been omitted from the Liquid 4 Pre's input stages completely. However, its influence on the circuit characteristics is still provided by smaller, cheaper components, including bespoke inductors.

Apart from the revised input circuitry, the Liquid 4 Pre is essentially the same as the original Liquid Channel, with the same dynamic convolution engine running the dynamic convolution, which lies at the heart of the Liquid Channel and the Liquid 4 Pre, works by using impulse response measurements taken from real preamps at a series of different amplitudes (all the way from the clipping point down to the noise floor), which together create an accurate acoustic image or 'sample' of the characteristics of the preamp. These characteristics are then imposed on the input signal via a process of convolution. This is a very intensive number-crunching exercise, requiring a very powerful DSP — and with all four channels running at 192kHz, the Liquid 4 Pre DSP is processing 256 million samples per second!

Since the Liquid 4 Pre uses the same convolution engine as the Liquid Channel, it has been launched with the identical default factory set of 40 vintage and classic microphone preamplifier emulations. It can also be expanded to use any of the Liquid Channel's additional preamp emulations (25 new emulations were introduced in 2005, and 12 more in 2006).

Given the multi-channel nature of the Liquid 4 Pre, the other major difference from the Liquid Channel is in its interfacing facilities. The rear panel accepts mic and line inputs for each of the four channels, all on XLRs, as you would expect. The preamp outputs are provided as both analogue line level on four more XLRs, and as two pairs of AES3 digital on two more XLRs. There is also a single AES3 (stereo) input, on yet another XLR. This input can be accessed by any or all of the preamps, if required.

There are also three ADAT lightpipe ports, labelled as Main and Aux outputs plus a 'Slave' input. As usual, elevated sample rates reduce the number of channels available in the ADAT interface, but this will only be a practical issue for rates above 96kHz, when you lose two channels from the interface. The Slave ADAT input is provided to allow the outputs of a second Liquid 4 Pre to be routed through the first, so that the combination of channels can form a full eight-channel ADAT stream (assuming the unit is operating at 44.1kHz or 48kHz sample rates).

Even more interesting is a blanking panel over an interface slot for future alternative interface cards. Already listed, but not available for the review, is an (eight-channel) Ethersound card, which will appeal to the live sound fraternity in particular. If the Slave ADAT input is being used, the eight-channel combination can also be routed out via the optional Ethersound card.

A little more humdrum in comparison is the pair of BNC connectors for word clock in and out, and an RJ45 socket for Ethernet (100baseT) connection to a computer, for software updates and remote control applications. This interface was chosen instead of USB since it can be used over greater distances and allows simultaneous control of more units — up to 32 devices (providing 128 channels) can be operated using the supplied Liquid 4 Control software.

For the sake of completeness, I should also mention the IEC mains inlet with a four-way mains voltage selector (although I was slightly concerned that the labels on the selector didn't match the options listed ▶

Photos: Mike Cameron

Alternatives

The Liquid 4 Pre is unique — there is nothing else on the market with comparable tonal versatility (other than its single-channel sibling, the Liquid Channel). However, there are many traditional high-quality mic preamps around the same budget level that would warrant consideration. A personal favourite is the Benchmark PRE420, which provides four very high-quality mic preamps with an integral stereo mixer and monitor section. The API 3124+ four-channel preamp costs roughly the same and has a well-deserved high-end reputation, as do the Universal Audio 4110 and the Focusrite Red 1 — all three of which are included in the Liquid 4 Pre emulation library. Priced slightly higher than the Focusrite box, the GML 8304 is a fantastic preamp (it is my own high-end reference), as is the Maselec MMA4 and the Neve 4081 (four classic 1081 preamps in a box — another inclusion in the emulation library). So the Liquid 4 Pre sits amongst some venerable competition, but certainly isn't shamed by any of it. Furthermore, for the price of any one of these units, the Liquid 4 Pre provides almost all of them at the flick of a switch. Food for thought...

SOUND ON SOUND

Focusrite Liquid 4 Pre £2114

pros

- Four independent channels.
- Memory recall of channel parameters, singly or *en masse*.
- Versatile and comprehensive analogue and digital I/O.
- Very clean and transparent native preamp character.
- Unrivalled preamp emulations.
- Excellent A-D and D-A conversion.
- Master-slave mode and remote control options.
- Multiple gain-ganging modes.

cons

- Boot-up lightshow!
- LCD screens can be annoyingly bright in a dim room.
- Menu operation not as intuitive as the front-panel controls.
- Quality sound still costs.

summary

The Liquid 4 Pre takes the core technology of the original Liquid Channel and redevelops it to form a more versatile, and in many ways more useful four-channel preamp.

FOCUSRITE LIQUID 4 PRE

Each channel has a decent-sized screen with clear metering and other channel information.

on the back of the unit itself — but then the review model was an early production unit).

The Liquid 4 Pre occupies 2U of rack space, extending 270mm behind its highly polished rack ears. The unit is quite heavy, weighing in at 5.3kg, and although the left-hand side gets noticeably warm around the power supply there are unlikely to be any cooling issues.

Controls

The front panel mimics, to a degree, the Focusrite ISA428 preamp, thanks mainly to the use of four LCD panels that act as meters (and configuration screens), each with a rotary encoder below.

The panel is divided into four identical preamp sections, plus a configuration section at the left-hand side. Neat, illuminated push-buttons are used throughout the panel, along with the four rotary encoders and distinctive grey columns with a trio of LEDs to indicate various modes.

The leftmost section includes a power button which illuminates blue, along with the 'ff' logo above — very stylish! Two LED columns show which of the six available sample rates (44.1 to 192kHz) is in use. A button below (green) selects the external word clock option, and another green LED indicates whether the external clock is valid (steady) or not (flashing). A third button (that flashes green) accesses the Setup menu, which is displayed on the first channel's LCD screen.

Each preamp section has a column of five illuminated buttons to the left of the display, and four more at the bottom, to the right of the encoder knob. The top three buttons in the column switch on phantom power (red), polarity inversion (green), and the high-pass filter (green). The latter has a 12dB/octave slope and turns over at 75Hz (-6dB point).

The next button isn't illuminated, but cycles around the available inputs of mic, line or a digital input, this last option being pre-selected via the preamp's setup menu. The options are AES (left or right), ADAT (channels 1-8) or option card inputs (1-8). At elevated sample rates the ADAT and option card inputs will only provide four (88.2 and 96 kHz) or two (176.4 and 192 kHz) channels. The selected input is indicated by another trio column of small green LEDs. The bottom button (yellow) activates the Session Saver function, which is carried over from the Liquid Channel. If activated, this essentially resets the preamp's gain setting to a lower value automatically if the input signal reaches 0dBFS. An associated LED illuminates if the Session Saver has been triggered.

To the right of the encoder knob is the array of four buttons mentioned earlier. The top-right one (green) forces the input stage to provide the highest available impedance (usually 10kΩ), regardless of the correct impedance for the selected preamp emulation. High-impedance mic inputs are quite fashionable at the moment, and generally result in a brighter, and often slightly stronger, signal. They can also have the effect of reducing resonant peaks in the response of some dynamic mics. We are talking about a subtle creative tonal shaping option here — but a useful one nonetheless.

The bottom left button allocates the encoder knob to adjust the Harmonics function, which introduces progressive distortion. On a technical level, this provides the means to replicate the kind of variability that afflicts ageing vintage preamps, and on a creative level it allows a degree of thickening or dirt to be introduced to beef up the sound character. The amount of harmonic distortion is scaled simply from 0 to 15, and the character varies with the current preamp emulation. However, at the top end of the range the harmonics function adds as much as 10 percent second harmonic, 20 percent third harmonic, and 10 percent of fifth-order distortion (for near peak-level signals). The first half of the scale (below about 8) generally only adds second-order harmonics, while the top half applies much more in the way of odd harmonics — which, intuitively, is what you would expect. The button flashes green when pressed, to remind the user that the encoder is configured to adjust the level of harmonics, and the LED ring around the knob shows the current setting (along with a numerical value in the display).

The top-left button configures the encoder knob to scroll through the available preamp emulations, and pressing the knob selects and activates the one currently displayed in the channel's LCD screen. Again, the button flashes green while in this mode.

"If you want squeaky-clean and accurate, it can do that. If you want characterful it can do that. If you want dirt and attitude, it can do that too. And if you want to rework something at line level, it can even do that."

Finally, the bottom-right button accesses the channel's setup menu, which is revealed on the LCD, and the button flashes green once again while this mode is active. I'll come back to this menu in a moment.

The encoder knob, when not being used to adjust harmonics or select preamp emulations, adjusts the preamp's gain in 1dB steps, in the usual way. The available gain varies with the selected source, ranging from +14 to +80dB for the mic input, and -10 to +32dB for the line and digital inputs. Once again, the LED ring around the knob indicates the approximate gain setting, while the LCD reveals the precise numeric value.

The four display screens are very bright blue-on-white affairs, but they are easy to

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read at a distance, thanks to their uncluttered layouts. In normal operation the screen is divided into five information areas, with the top line providing the preamp number, a user-defined name, and the current preset memory number (if in use). Below this is a meter display, which is configured globally (from the System Setup menu) either as a peak-reading horizontal bargraph or a VU-style meter — the latter with user-defined calibration for the 0VU mark between 0 and -24dBFS. Supplementing the meter display is a bright red LED above the phantom power button, and this illuminates when digital clipping occurs.

The next display line shows the current preamp emulation name and the harmonics level, and below that to the right is a double-height display of the preamp's gain setting. To the left, the two bottom display lines show the selected input source and its status. These really come into their own when the digital input is selected, since the display reveals which of the digital options has been preselected and whether the clocking is correct. It also indicates how much attenuation has been introduced by the Session Saver function (if activated).

While this may all sound rather more complicated than your average mic preamp, the basic configuration and operation is actually entirely logical and intuitive. It is not so much complex as extremely flexible and versatile — but easy to use all the same.

Menus

The only slightly confusing and less-than-intuitive parts of the design are the Channel Setup and main System Setup menus — but, to be fair, the logic of these becomes clear enough after ten minutes of messing around with the unit.

The basic functionality is to scroll through the available options listed on the screen, with the encoder knob. Pushing the knob accesses the current function, allowing its value to be changed by turning the knob again. A second press of the knob exits the parameter adjustment mode and allows list scrolling again.

The size of the LCD screen means that a maximum of four options can be shown at a time, and most screens only show three parameters, so the majority of screens include a 'More' option, which enables you to access the next page. Some also have a small open box included in the scroll list for each page, in the bottom right-hand corner. Clicking in this box either moves back up to the previous menu page, or exits the menu altogether. Pressing the Setup button on the front panel of the Liquid 4 Pre always exits the menu, regardless of the current page.

The Channel System menu comprises just two pages, and starts by allowing the channel preamp to be named (up to 12 characters in upper or lower case, numbers and symbols, such as question marks and exclamation marks), and channel-setting memory presets loaded or saved (99 are available, and can be named). Current preamp settings are stored internally every 10 seconds anyway, so that the unit will power up with the last-used configuration.

The second menu page allows the digital input to be pre-selected and a linking bus established. There are four link options (as well as the option to switch off linking), the first two ganging gain-settings directly, and the second two ganging them while maintaining any current level offsets. The first mode is essential for normal stereo mic arrays, for example, while the second is handy for MS pairs and some kinds of surround array.

The main System Setup menu is a little more involved, with three pages. The first provides options to set the sample rate and clock source (internal, word clock, ADAT, AES, Ethernet or word clock with a 75Ω termination), and to load or save preset unit memories. There are 99 of these as well, which can store the entire configuration of the machine — all four preamp channel parameters, sample rate and clocking, and so on. As with the channel memories, they can be named, but (strangely) the Load and Save menu functions operate slightly differently from the Channel menu, using an additional screen for naming and selecting them. Usefully, there is also a System menu option to reset the entire machine to a default start-up condition.

The second menu page provides functions to disable all front-panel controls (apart from the System Setup and mains power buttons), to configure the unit as a master or slave (in terms of the ADAT channel amalgamation and setting the necessary clocking), and to name the unit for remote-control recognition. The third page allows the LCD meter-type to be determined (digital peak-reading bargraph or VU, as mentioned earlier), the 0VU reference point to be set, and the Ethernet access parameters to be configured (Auto DHCP, or manual with options for IP address, mask and gateway).

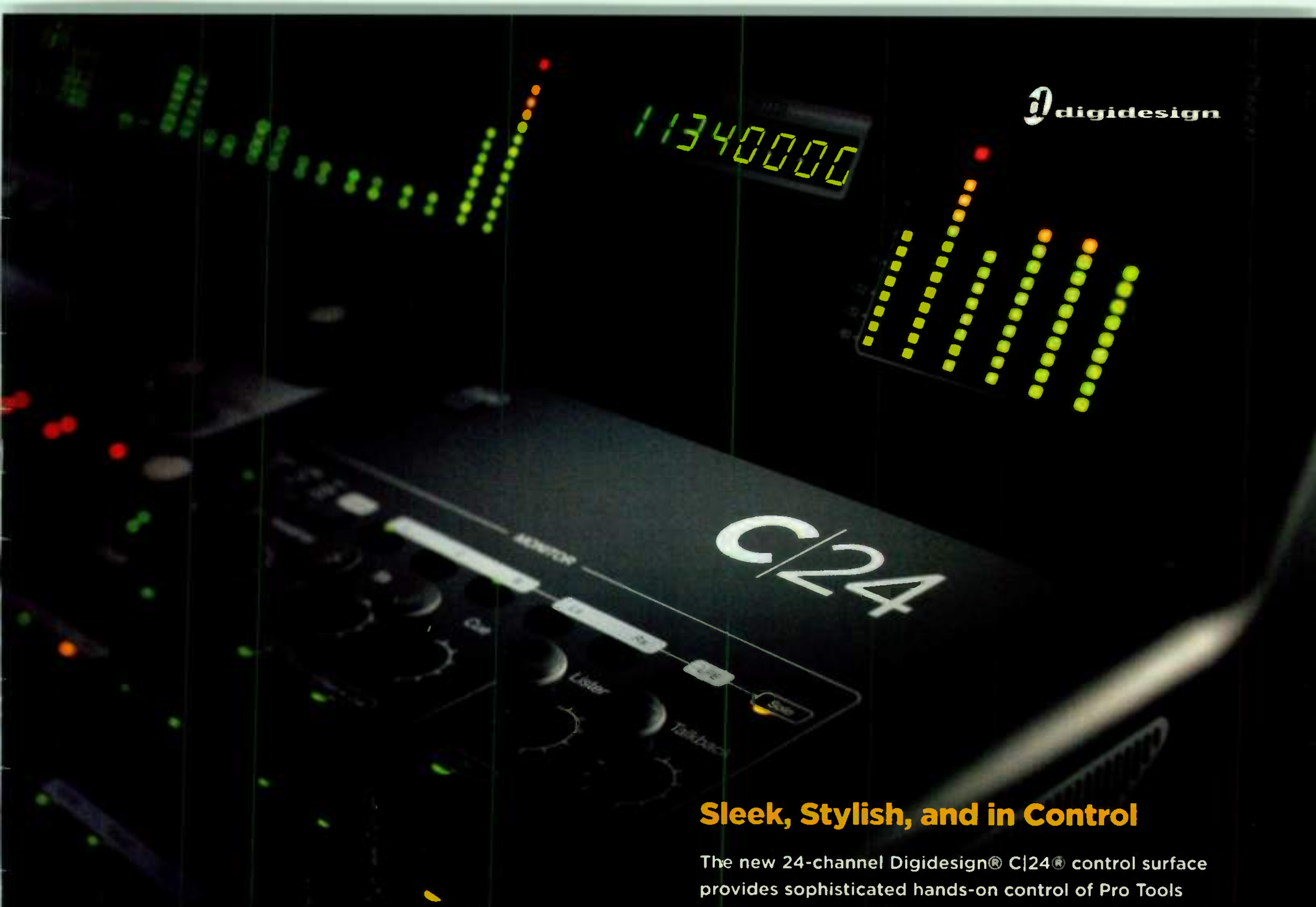
Liquid 4 Control

The supplied software provides very comprehensive control of all the preamp's functions and preset memories via an Ethernet link. As mentioned earlier, up to 32 Liquid 4 Pre units can be controlled simultaneously, and the software has been designed to work within your DAW environment. A Pro Tools TDM/RTAS plug-in version also allows integrated remote control via Pro Tools hardware and software, including Icon and Venue systems.

Up to 32 Liquid 4 Pre units can be controlled simultaneously using the remote-control software.

Impressions

The Liquid 4 Pre very clearly builds upon the original Liquid Channel — and that unit impressed me enormously, for a number of reasons. The Liquid Channel's analogue preamp stage is quite exceptionally transparent in its 'natural' state, the digital converters are extremely clean and accurate, and the preamp emulations provide phenomenal tonal and creative variations at the touch of a few buttons. In some ways, it is almost irrelevant whether the emulations are truly accurate or not — the fact is that they provide a wealth of creative options to colour and shape the sound in ways that simply aren't available in any other device. But when I was able to make direct A/B comparisons with some of the very units that Focusrite had sampled for the original emulations, I found the Liquid Channel to be astonishingly accurate — and all but indistinguishable when working at the higher sample rates.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Frontier Design Alphatrack

If you want hands-on control over your DAW, without the expense or inconvenient size of a multi-channel controller, the Frontier Design Alphatrack could be just what you're looking for...

Paul White

Having used a desk full of Logic Control surfaces and expanders for the past three or four years, I have come to realise that some things are still done faster by mouse, and most of the time I use only the transport controls and the faders. When it comes to accessing and adjusting plug-ins or tweaking EQ, I still gravitate towards the mouse. If you have the same kind of relationship with control surfaces, you may well be interested in Frontier Design Group's Alphatrack, which measures a compact 8.5 x 6 x 3 inches (or 22 x 15 x 7.5cm) yet provides all the key DAW control functions in a very friendly and stylish format. Most common DAW software is supported, including Pro Tools, with new applications being added whenever possible.

One-track Mind

The Alphatrack connects to your PC or Mac, and also draws its power, via USB, making it ideal for mobile use with laptops, as well as for desktop systems.

The unit is equipped with a single motorised, long-throw, 100mm fader, three rotary encoder knobs with integral push-switches, 22 buttons (with 21 status LEDs) and a horizontal, ribbon-controller-style pad for shuttling the ▶

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The Feeling: Recording *Join With Us*

Mike Senior

The Feeling stormed the charts with a debut album recorded in a shed. For their follow-up, they relocated to a country mansion, but elected to stick with the DIY approach.

There can't be many people sneaking elements of Supertramp, Jellyfish and ELO into the charts, and certainly not with as much pop savvy as the Feeling. What makes this feat all the more surprising is that the band recorded much of their debut album *Twelve Stops And Home* in their own homes, using project-studio gear that will be very familiar to *SOS* readers.

Despite their subsequent success, the group stuck to this self-op ethos for the recording of their new album *Join With Us*, albeit within the salubrious surroundings of Bradley House, home of the Duke of Somerset, and with the help of a few more studio toys. I joined Dan Gillespie Sells (vocals/guitar), Kevin Jeremiah (guitars/vocals), Ciaran Jeremiah (keyboards/vocals), Richard Jones (bass/vocals) and Paul Stewart (drums) several weeks into the sessions, to see how things were progressing with the new songs and to find out more about how they had produced *Twelve Stops And Home*.

Us & Us Only

Paul: "For this record, it was a case of expanding on the bits we enjoyed about making the last record. We're doing the same thing again, really — just us in the countryside — but back then we were effectively in a large shed and now we're in this place! We also never really had a direction for the last record — we just followed our noses. We didn't think we'd be making a commercial record, just the music we wanted to make, so we thought we might as well do the same this time."

"We had the 'go and work with this guy or that guy' and 'go to LA and make and album'," adds Richard. "All that kind of stuff came up, but we felt that the best thing to do was what we did last time. We have enough ideas between the five of us at this stage — it's only our second record!"

During the two-year period of touring *Twelve Stops And Home*, Dan had been busy writing material, which meant that most of his new songs were already written before the recording sessions began. Richard explains the different ways Dan's ideas were developed by the band as a whole: "We've got two techniques for how we tend to record. For some of the stuff, Dan would have been beavering away working on a demo, forming a good starting point, and then we'd take this and start laying things on top, adding parts, replacing parts.

"In other cases it would be more of just a straight song idea, and then we'd just sit down together and start putting things down. We've got this V-Drums technique we use, where we get the arrangement roughly, and then record a take together using the V-Drums so that you don't have the drums on the other tracks. Paul records real drums along to those parts, so it's almost like you're playing together, and then we'll start laying

The Feeling outside Bradley House, their 'home studio' for the summer. From left: drummer Paul Stewart, keyboard player Ciaran Jeremiah, singer Dan Gillespie Sells, bass player Richard Jones and guitarist Kevin Jeremiah.

Photo: Richard Ecclestone

The Feeling's unusual tracking approach involves laying down basic takes as a band, but with Paul Stewart playing V-Drums to avoid spill. These are later replaced by real drums.

on all the other parts from there. That way we can focus on getting the drums right, because we don't have to all sit there and play at the same time. Besides, we definitely prefer the layering approach — it suits our music more — rather than doing three takes all together and choosing one of them."

Paul: "We used the V-Drums on the previous record too. One of the main reasons was that we didn't have as many mics then, and it wasn't easy to pull in lots of mics when we were borrowing them from friends. The V-Drums were also good from a noise perspective when doing demos — we'd use V-Drums for kick, snare and toms, and then have some nice cymbals so we only had to use three mics."

Richard laughs. "It never sounded that good though! We started off initially doing our demos with our Delta 1010 and some crap mics we'd borrowed, and ended up with a pretty badly recorded kit. To try to make it more convenient we started using the V-Drums, but they actually sounded not as good as the crap recording of the live kit, funnily enough!"

"As good as V-Drums are," adds Paul, "and they are brilliant, it's very difficult to get the kind of sound we're after with them. Certainly cymbals-wise, there's absolutely no substitute for the real thing." In the end, a lot of the first album's drum parts ended up being replaced during overdubbing sessions at Olympic studios. "The drum recordings we did in the shed were awful," emphasises Richard, "but if the performance was really special, then we'd find a way to keep it — Spike Stent [*mix engineer for several songs on the album*] did all kinds of stuff at the mix, like miking up a speaker in his bathroom and playing the kit through that."

Everyone's An Engineer

Although engineering their own recordings has involved developing a few unorthodox strategies (like their V-Drums technique), the band were keen to stick with this approach even as their career took off and budgets expanded. Richard: "We're still engineering most of the recordings for ourselves, although we do have Owen, our technical handyman, who's around to stop us going mad setting everything up and plugging everything in. But all the mic placements and so on we deal with ourselves, and for comping or editing it's actually a lot quicker to do it ourselves half the time. You don't have to explain things and point at the screen, and someone else doesn't have to try to interpret what you're saying."

"We've known Owen since childhood," says Paul. "He's an old school friend, so it doesn't change the dynamic of the session. Having someone who's just being paid to be there feels a bit too much like being in a studio to us. The whole reason for recording the way we do is that we get left alone."

Richard: "We've always just picked stuff up and learnt things as we went along, ever since recording on four-tracks when we were 15 years old. We also learnt a lot about angles of mics and their placement from a great engineer called Cenzo Townsend, who recorded a lot of the overdubs on the last album." Paul nods his agreement: "First time around we didn't really know what we were doing, but we've learnt a lot, and now we think we know what we're doing!"

Perhaps surprisingly, every member of ▶

Vocal Recording: Then & Now

"The vocal mic we're now using is a Neumann valve mic, the M149," Richard reveals. "We used a Neumann TLM103 for the vocals on the last record — Kev, Dan and I have all got one, and we used them previously for drum overheads as well. The TLM103 became part of our vocal sound, and we thought we'd end up using it again, but the M149 is a very rich-sounding mic.

"That goes into the Avalon preamp, which is great. There's a history with that. I did a tour with a front-of-house guy who swore by Avalon stuff for vocals, back when Avalon weren't that well known. I heard one of those Neumann handheld vocal mics through one and thought it was brilliant, so I eventually got one for the studio. You get the warmth of the valve, that tiny bit of drive, and you can give it a bit of an EQ on the input — nothing too extreme."

The band are advocates of Avalon preamps for vocals. These feed Digidesign recording hardware, on a Mac running Logic.

RECORDING *JOIN WITH US*

The larger live room at Bradley House.

the band is happy to take on the role of engineer as required. "We call it 'Fatsing' — being the Fat Controller — and you normally have to wear the hat," laughs Dan, indicating the dapper bowler perched on his head. "We quite often end up Fatsing ourselves as well."

When it comes to making production decisions, all members of the band play an important part, as Richard explains: "It's a very natural process. We've got a good symbiotic relationship in the band, and various roles that everyone has. Everyone's got a good set of ears, but slightly different tastes, so when something sounds good, and we all agree on it, then you know it's right. But if something's not quite right, then one of us will normally point it out. It's a good filter, a five-man filter."

Despite this even-handed approach, there's still space for each person to play to their strengths. "For example," says Richard, "Dan, as chief songwriter, is the primary Musical Director, while Kev is the best engineer and is getting really good at mixing — he mixed most of our demos and some of the 'B' sides on the first album. I tend to be in the producer's seat, the man with the clipboard! Anything from booking other musicians, like string quartets, to steering the recordings in a certain direction, giving them colour and shape, so to speak. All these roles cross over, of course, but I think the fact that we have different strengths individually is what makes the process work so well when we are all together."

Logical Songwriters

For the new record, The Feeling invested in the latest rocket-powered Intel Mac and racks of MOTU and Digidesign interfaces, but it's a far cry from where they started out. Richard reminisces: "I remember the moment when I got Logic on my first computer and I was thinking 'Wow! This is amazing! I can't believe I can do all this on a PC with a little 30-quid Soundblaster!' We used to run 'built in the shed' 800MHz PCs — you'd put on three plug-ins and they would crash. The worst thing, though, was when you'd get some random crackle or click on the audio takes for no reason. We never got away from that — it was a curse. You'd do a great take and there'd be clicks on it. Plus most of the first record was done through an M-Audio Delta 1010, which meant that eight inputs was the maximum. We'd have to do one thing and then the next, which made the process much more long-winded."

Despite the Digidesign hardware, the band are still committed to Apple's Logic as their sequencer of choice. Richard: "We like the concept of working in Pro Tools, but we've spent so long in Logic that we don't want to be slowed down by changing software. What we find with Logic is that it's very easy to be creative with the effects and software instruments. It's really quick. The Tape Delay plug-in in Logic is also incredible. I've played with the plug-ins in Pro Tools, and they're not as easy to get the sound you want. I think that would hold us back creatively."

Stately Home Studio

The recording setup at Bradley House revolved around three main rooms, two set up for recording, and the other one between them acting as a control room. The larger live room was used for the drums, guitars, bass, and most of the keyboards. "That room has great acoustics," comments Paul, "because of the combination of all the wood and tapestries. It's a really lovely live room."

This live room was only separated from the control room by a small entrance hall, which had some practical advantages, as Richard made clear: "We set up all the amps in the live room so that we could all sit in a circle and jam, but with guitar overdubs we'd just run a lead into the control room. We wanted the best of both worlds, really."

The smaller carpeted room to the other side of the control room was used for a variety of overdubbed instruments. Kevin: "There's a honky-tonk piano that Dan brought down, an accordion, a harmonium, and we also do vocals in there. It's a deader room, a bit more like a living room — we were used to singing in a living room from before..." The room was also used to capture acoustic guitar and mandolin parts.

Taking pride of place in the control room was a 48-channel Mackie Onyx console, a straightforward choice for the band. "I've got the 16-channel Mackie Onyx mixer at home," mentions Richard, "which we used a lot on the first record for the vocals and things, so the new desk has been a natural step up. We know how to work with it — it's just been an extension of what we were already using. We don't really need that many channels, only it saves time because everything's always plugged in. If we want to do guitars, we can just get going, rather than spending 20 minutes moving mics and cables."

"The mixer's great," agrees Kevin. "We got it because basically we needed preamps, and for the money you get so many really clean, really nice preamps. We can't afford to buy a vintage EMI desk, and we know that if Spike mixes for us he's going to run things through all his vintage stuff anyway, so we might as well just record it clean. We also needed a desk with enough sends for the monitoring."

Mics & Drum Setup

The band's preferences in terms of microphones are constantly evolving. "We've been begging, stealing and borrowing mics for a long time," remarks Paul. "Plus we've been taking note of what was used when we've been in places like Olympic, and our front-of-house guy has also been helping us out with our selection."

"But the main thing that happened for this record," continues Richard, "was that we developed a relationship with Sennheiser through touring, and they also look after the Neumann brand. So we put together a master wish-list, and although we did wonder what the hell we were doing, it was actually worth it." Kevin chuckles: "The only thing we didn't get was the dummy head! We considered it..."

When it came to miking up the drums, Neumann U87s were used for overheads and for the ride cymbal, while Sennheiser MD421 close mics were deployed on the snare and each of the toms. A Sennheiser e602 was placed just in the hole in the kick's front head, and was supplemented by a Yamaha Sub Kick. Richard: "We don't really monitor the Sub Kick while we're working, but when you give it to the mix engineer they've got something at the low end to work with, so we can try to avoid having to use samples."

Equally crucial to the sound, though, has been Paul's careful selection of key elements of the kit: "I've had this kit for a while now, but I recently managed to pick up a separate Ludwig 400 snare on eBay for £100. It looks like an old heap of crap, but that obviously doesn't affect the sound, because it's the best drum I've ever had! There are two more snares I've got over there, and probably another five outside, and for the first period of recording here I lined them all up and tried then all out on different tunes, but I've always ended up coming back to this. It's brilliant — I'd swear by it! Certain cymbals also just don't record as well as others, and I find yourself coming back to the same set of hats and the same ride."

Casting a second glance over the kit, I noticed another Neumann U87 in an unusual position at the side of the snare ▶

Paul Stewart's acoustic drum kit is extensively close-miked, with a Yamaha Sub Kick adding extra substance to the kick sound.

A variety of guitars, amps and mic placements contribute to the Feeling's sound, often layered for extra depth.

drum, and asked Paul how it was being used: "We've only started doing that since we've been in here. It sounds great with that shell, because there's no nasty ring off it like there would be off a brass shell or anything like that. Whenever you get really close to a drum head you get all sorts of strange frequencies you don't hear from a distance, so it sounds really odd. The side mic gets a much more overall view, and captures the

Stewart's favourite Ludwig snare, with U87 positioned to capture the 'beef'.

top and bottom sounds together, but without the real upfront sound."

"It's been much more useful having that side mic than even a bottom mic a lot of the time," adds Richard. "What you get out of it is 'beef' — it's the thing we've always been looking for. You get the 'crack' from the top mic, fizziness from the bottom mic, and then this side mic seems to find that 'beef' area. It's the side of beef!"

In addition to the close microphones, three other mics were used to capture stereo ambience: a Neumann USM69i stereo mic, and a pair of boundary microphones. These ambient mics were also used for pianos, guitars, and other instruments, but were always recorded to separate tracks to keep the mixing options open.

Recording Guitars & Bass

Although Richard talked of the band using a wide variety of different amp and microphone combinations for electric guitar recording, there were no real general rules he could pass on, because the mic setup was adjusted to suit each different sound. "It's good to have all the options, but we laugh about the fact that sometimes all we're using is an SM57 on the 4x12. The mic we've got on the Vox AC30 just now is the Sennheiser e606, which gives a good sort of gritty rasp to the amp."

Despite changing mic setups, a trick to which the band frequently return is layering takes for a meatier sound. "Quite often we'll do two takes of everything so that we can spread them," says Richard "even if we're using three to four mics on each guitar sound. So we'll do two takes of each guitar sound, to give it a thickness, and even down-tune and do another take."

For recording his bass, Richard has a tried and trusted approach based on his experiences making the last record: "I've got an all-valve Ampeg SVT2 re-issue, which is really warm, and it's miked with a Sennheiser e602 on the 8x12 cab. I always DI and have another channel with a Tech 21 Sansamp, which is a trick I learnt from Spike Stent, who mixed some of the last album. Our first single was essentially our demo with a couple of extra parts, and he got an awesome bass sound. I said 'What have you done to the bass? It sounds incredible!' He just pointed at that box. He apparently uses it for every bass that comes through his studio, so I bought two straight away! I use it live as well now. It's brilliant! The DI gives you the pure sound of the bass, the amp gives you the bottom end, the air moving, and the Sansamp just gives you some grit — but it's good grit, not really distorted."

Three Upright Pianos

Keyboards are of course vital to The Feeling's sound, so it's no surprise that the band have been using a lot of them for the latest sessions, not least three different upright pianos. Richard: "The Kawai is the one that we mainly use. It's brighter, a better recording piano. In tracks where it's just piano and vocal, the Bechstein gives better tone — it's more vintage-sounding — but once the band's in you need a brighter sound. There's also the honky-tonk in the other room — we spent a couple of hours putting drawing pins into the hammers to get that sound!"

Dan goes into a bit more detail: "Bechstein pianos are great, but this one was knackered, so I had to have the whole lot restored: restrung, new pins, new mechanism. I wanted to have a piano with ivory keys, woody and old-fashioned. It's a bit out of tune at the moment, which I quite like as well — it's a textural thing. This piano's got loads of character to it, it's warm. You'd never get the same sound out of a modern piano, especially in the mid-range. It's kind of honky and it reminds me of old records. The way they make modern pianos is just so different.

"But a modern piano is lovely for recording, because it's bright and dynamic and it cuts through the band, and that's what the Kawai's for. This Kawai's slightly warmer than most of them and it was slightly out of tune when we used it to record all the piano parts on *Twelve Stops And Home* — it was in my living room at the time. I always pull all the panels off the piano while recording. It just lets the sound get out a bit more."

When it came to miking the uprights, Dan had a pair of U87s set up about 18 inches from the strings, over the keyboard on either side of his head and pointing at the hammers. "Miking the back of the piano and the underneath of it and all that stuff is a complete waste of time for us. I want to hear the click of the hammers, so I have the mics near my ears, over the keys. Otherwise you just have to EQ the hell out of a piano when you're recording because it's not bright enough. You get sniffles and funny little bits of noise from when you're playing, but I like that. We have all sorts of background noise over the recordings, but you only really notice if you solo them. As long as it's not car horns, you're alright. There was quite a lot of traffic noise over the stuff we did for the first album back at Richard's place. It helps having the room mic as well — if you get it the right distance away you get the brightness of the piano and the room. We normally have it about six foot away."

Being familiar with the layout of Mackie Onyx desks, the band chose to use one for the album sessions rather than investigate vintage or ultra high-end alternatives.

Keyboard Playground

Other favourite keyboards include the Roland Juno 106 (Dan: "The best keyboard in the world — all over the last record."), Wurlitzer, Rhodes, Hohner Clavinet and Moog Voyager. "We use the Moog a lot, and we used it on the first album as well," recalls Richard. "It's actually my wife's, and it's one of those things that if you have it lying around it just finds itself on tracks because it's got such cool sounds. It kind of suits the way we've always worked. You surround yourselves with instruments, and then you pick up what's around and it starts to colour the production of the track you're working on. The more interesting instruments you have, the more they take you in different directions stylistically."

And no other keyboard in The Feeling's collection was provoking as much interest during my visit as the recently acquired Cordovox CDX-0632, dubbed the 'Astro Sound' by the guys on account of its front-panel legending. Richard explains how they came to own it: "When we were on tour in Cleveland there was a shop next to this quite famous venue, and all the bands go there so prices are a little bit higher than they would be, but they've got about five floors of stuff. I bought a Gibson Hummingbird acoustic there, which we've used on loads of stuff. They had a keyboard room with Wurlis and everything, and the Astro Sound was sitting in the corner. It was $1000, and the guy in the shop said that every band that had gone in there for two years had said 'That's amazing!' and then walked out without buying it. So we got it, and we've already used it on about half the songs on the album!

"It's got a very unusual distinct sound to it. The only other person I think has one is Beck's keyboard player, but he only has the shell, and he's put a controller inside it to use live. I've been looking on eBay for another one as a spare, but I can't find one — if it breaks we're screwed! The Repeat function is really cool — it's a sort of du-du-du-du sound which you can speed up and slow down. The Phase Shifter on it gives a real rotary-speaker feel, because you can really hear it speeding up and slowing down as you switch it on and off. I hope we can find another one of those, because we love it so much. It's the way you can dynamically change the sound as you're playing that makes it great."

Pride of place among the band's many keyboards goes to the vintage Cordovox.

Finishing The Job

With most of the rhythm tracks completed at Bradley House, the band plan to head back to London to press on with overdubbing. Although Olympic was used for the last record, they're planning to go it alone this time. Richard: "We're looking to get a room in London — we've got all this gear, so we could just move it straight in — but it's quite hard to find actually. We want something treated, ideally with a live room and control room, but with nothing in it. They exist, but they're normally on leases that producers have.

"We'll do a lot of stuff at home as well: keyboards and a lot of vocals. Dan particularly enjoys working on his own on his lead vocals, because he can have his own space, and when his voice feels right he can just go. For BVs we just all get together for an afternoon to do them."

Plans are also already forming for the next stage once tracking is complete. "There's a couple of mix engineers in the frame at the moment," says Richard. "When the tracks are at the point where we feel they're ready to mix we're going to get them to a couple of guys and hopefully go in with Spike and do something with him. A lot of the time it's the mix which really shows you whether you've got everything you need or not." SOS

IK Multimedia ARC

Room Correction System

Could this be the holy grail of studio acoustics — a room correction system that actually works?

Hugh Robjohns

Room acoustics, especially the acoustics of the monitoring environment, are the bane of every audio engineer's life. Every room suffers from low-frequency (LF) modes, the low-frequency peaks and dips that are caused by sound waves bouncing around inside the room producing interference patterns. Traditionally, since the LF room modes are caused by physical reflections, a physical solution is normally employed — specifically, bass trapping and broadband absorbers. These soak up the sound wave's energy as it approaches the walls, so there is nothing to reflect, and thus nothing to cause interference peaks and dips in the room. This is an extremely effective (and relatively cheap) solution, but it can take up a lot of space.

Consequently, quicker, easier, electronic fixes for room modes have been sought for as long as the problem has been recognised. The earliest version was simple equalisation — measuring the room's response at the 'listening position' and applying an inverse response to the monitor speaker's amplifier chain using an equaliser. The end result should be a near-perfect flat response... but the idea is fundamentally flawed and generally does way more damage than good.

For a start, the room's response varies enormously throughout its area, because different reflected frequencies interact with each other in different places. So moving the measuring microphone a small distance one way or the other typically produces very different corrective response plots. The perfect equalisation for the measured position may well end up making the situation far worse at other positions in the room.

Another problem is that response irregularities caused by cancellation nulls (where reflected sounds arrive in opposite polarities and cancel each other out) can produce very deep response notches — often 30dB or more. Equalisation can't easily address this, partly because the amplifiers and speakers are unlikely to be able to generate sufficient energy to fill those dips, but also because if a dip is caused by reflected waves cancelling each other out, more energy won't resolve the problem, as the cancellations will still occur.

More significantly, room modes aren't just about frequency response. We're dealing with resonances here, which means stored energy and time-domain responses, often referred to as modal ringing, which normal equalisation can't address.

SOUND ON SOUND

IK Multimedia ARC £469

pros

- Easy to operate and use.
- Good support for multiple platforms.
- Supplied with a measuring microphone.
- Effective, but not an excuse to avoid all room treatment!

cons

- Accurate only with its own microphone.
- Shifts the emphasis from proper room treatment.
- Expensive.

summary

A sophisticated software room-correction tool that works better than most.

For all these reasons, simple monitor EQ is rarely effective for correcting a room's inherent acoustic issues, but it *can* be useful in fine-tuning a room which has already been properly treated to resolve the major room mode and reflection issues. The more sophisticated the approach, the better the results are likely to be. Digital equalisation can provide incredibly narrow response peaks and notches, along with precise phase correction and, to some extent, simple echo-cancellation, so the ideas of electronic room-correction are becoming popular again (many monitor systems now include some kind of digital room-mode correction facility).

Overview

IK Multimedia's ARC 'advanced room correction system' works as a software plug-in for DAW-based studio environments. The room is measured using the supplied mic and analysis software, which calculates the necessary corrective equalisation. The DAW plug-in (which comes in VST, RTAS and AU formats) uses the calculated response to correct the output from the monitoring system.

The room-correction processing is derived from a technology produced by Audyssey Laboratories (www.audyssey.com) that has been implemented mainly in sophisticated hi-fi products from Denon, Marantz, NAD and others. The system is called the 'Audyssey MultEQ,' and in these hi-fi applications is able to correct surround sound installations as well as stereo setups, so presumably a surround sound version of ARC may become available in the future.

It is claimed that the MultEQ system analyses patterns in the frequency- and time-domain responses, measured at multiple points around the listening area, and then generates correction responses for each channel, so as to optimise overall accuracy across as wide an area as possible. Pattern-recognition and 'fuzzy logic' algorithms are involved — Audyssey don't give much away in their explanations, but they do explain that the MultEQ process uses time-domain based FIR (Finite Impulse Response) filters, with the emphasis on resolving low-frequency irregularities.

ARC Package

The software application comes on a CD-ROM, and there's also a manual and a tough plastic case containing a small-diaphragm electret measuring microphone (with foam windshield and stand adaptor) which requires phantom power.

The user obviously needs an interface of some kind to get the microphone's signal into the computer, and the corrected monitoring signals back out from the DAW. Of course, the frequency and phase response of the hardware interface (its preamps and converters), is included within the acoustic measurement loop, so the more neutral and accurate the interface's sound quality, the better. Any strong tonal character or coloration will tend to get ironed out!

Interestingly, the measuring microphone's grille looks very Bruel & Kjaer-ish, although I suspect this is a less expensive Far Eastern alternative. The mic has a stated tolerance of ±1.5dB between 20Hz and 16kHz, and the software compensates for its falling HF response above that (which means you're likely to get false results if you use a different mic — even one with a flatter, wider response.

The software will operate on PCs (XP or Vista with a minimum 1GHz Pentium or 1.33GHz Athlon XP processor and 512MB of RAM), Power PC and Intel Macs (minimum 866MHz G4 Power PC) running OS 10.4 or later with 512MB of RAM, or 1.5GHz Intel Macs with 512MB of RAM, running OS 10.4.4 or later.

Installation

I found installation remarkably fast and trouble-free, and as the ARC licence allows installation on up to three computers (provided only one instance is used at a time), I installed the software on both desktop and laptop PCs. There is a 10-day grace period to register (after which the software reverts to demo mode), but on-line registration was quick and easy. Having registered, I was invited to download the latest update (V1.0.1), which corrects a problem with "'slight phase inaccuracies in low frequencies in certain speakers/room configurations."

My preferred DAW is SADiE, along with Wavelab 6 and Adobe Audition, and I was able to operate the ARC system in all these without difficulty. I have a Roland M1000 digital mixer which incorporates a USB

Your audio card needs to be set to a 48kHz sample rate when measuring the room response. A prompt to change this appears if it can't be done automatically, as was the case with the interfaces used in the review.

interface, so I fired that up for testing with the ARC software, too. Whatever interface you use, it must be ASIO compliant for use in Windows XP and Vista, or Core Audio compliant for use on Mac OS X.

Room Measurement

I tested the ARC system in two very different monitoring environments. One was my domestic listening setup, currently using a pair of PMC IB1 three-way monitors. The room has corner bass-trapping, along with four Realtraps broadband absorber panels to tame the acoustics, and I'd suggest that the overall quality is very good. The second wasn't really a monitoring environment at all: it was my office, with a pair of tired old two-way speakers (I'll not name them to avoid embarrassment...) sitting either side of the desk (one in a corner formed with a bookshelf), and used purely for background music while I'm typing. This setup is quite obviously bass heavy and lopsided, with lots of early reflections thrown in for good measure.

Neither of my interfaces have mic inputs, so I used a Sound Devices MP2

Alternatives

Some monitors now include DSP room correction, but I can't think of anything else on the market that is directly comparable with the ARC system.

location preamp to handle the test mic's output and provide phantom power. The mic was mounted on a stand and arranged to point straight up towards the ceiling at approximately ear height, as per the instructions: like most test microphones, it is balanced to work in the diffuse field, and so it is important that it doesn't face the monitor speakers directly.

With everything plugged up and ready to go, I fired up the measurement software. The first step is to select the required interface hardware and configure the appropriate microphone input and monitoring outputs from drop-down lists. A minor stumbling block here is that the measurement has to be performed at the 48kHz sampling rate. Your interface should switch to operate at 48kHz automatically, but some (including mine) can't be remotely controlled, in which case a warning dialogue box prompts you to change the sample rate manually.

Once this is all configured properly, the correct replay and record levels must be established. A button allows the test signal to be generated continually, while a bar-graph meter shows the level coming back from the microphone. The idea is to adjust the replay level so that the 'chirp' (a fast frequency sweep) is reproduced at a representative listening level, and then to set the mic gain so that the signal peaks within the 'OK' region of the meter.

Next comes the room measurement process, which takes about ten minutes. For each test position, 10 separate chirps are replayed from the left monitor, followed by 10 more from the right. A minimum of 12 different locations are needed, although up to 32 measurements can be made for even better results. The manual provides several examples of how to select the appropriate measurement positions, but in essence, the sequence starts at a central reference listening position, followed by a variety of positions around the entire listening area in symmetrical left-right pairs, finishing with a second centre-line position.

In the office, I started where I normally sit, and then sampled pairs either side, slightly forward and back, slightly closer


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and wider, until I had built up a good cross-section of my listening locale. In the listening room I covered a rather wider listening sweet-spot, since several people often listen together in different chairs and sofas.

Once all 12 (or more) positions have been measured, the software automatically calculates the appropriate equalisation and phase corrections, which will subsequently be implemented by the DAW plug-in. Filter parameters are calculated for all standard sampling rates from 32kHz to 96kHz, along with four 'target curves.' These provide the default flat response, an HF roll off, a mid-range dip, and a mid-range dip with HF roll-off. The HF roll-off is intended to compensate for stronger reflected HF in smaller rooms, and the mid-range dip is to correct potential crossover anomalies.

The calculated filter parameters can then be saved as a user preset and named, which is handy if you need to correct for different speakers in different environments used at different times. The software includes graphic representations of a variety of popular monitor speakers, which can be selected and saved within the preset as an *aide memoire*. Amusingly, the PMC IB1 was amongst the options — along with familiar looking Genelecs, ADAMs, JBLs, Tannoys and many other popular monitors. Sadly, there was nothing that came close to resembling my tired old office speakers. Oh the shame...

Once the parameters have been saved, the measurement program can be closed and you can launch the DAW application. The appropriate form of ARC plug-in is then loaded into the monitoring or main output bus. If the main output is being used, then the plug-in will need to be bypassed when you output your final mix, otherwise the mix will incorporate all the speaker EQ tweakery, which wouldn't be a good idea!

Opening the plug-in window allows it to be configured, which is simply a case of selecting the appropriate measurement parameters file from a drop-down box, and then selecting the required target response curve from another drop-down. A pair of frequency-response charts is displayed, with three traces on each. An orange line represents the original measured monitoring-system response, a green line shows the intended target response, and a white line shows how closely the system claims to have come to that target.

Interestingly, the manual states that the software works out the LF response of the monitors for itself and doesn't attempt to drive the speakers beyond their natural bass extension limits. This became very clear when I compared the plots. The little office speakers were shown to roll off from about 70Hz, whereas the big IB1s went right down to about 30Hz.

Several measurements are required by the ARC system in order to build a good picture of the room response. By taking a measurement at the listening position and then further ones each side, you create a better cross-section of the range of normal listening positions that are likely to be used.

A bar-graph meter indicates the signal level through the plug-in, and a large 'Correction On' button to the right allows the processing to be switched in and out for comparison. Usefully, there is also a level trim control to help balance the bypass and processed signal levels.

Impressions

I approached the ARC system with a large dose of scepticism. I've experienced the failings of simple analogue and digital monitor-equalisation countless times before, and while this is a much more sophisticated approach than many, you still cannae change the laws of physics, Captain! So, starting with the listening room setup, what did I hear?

Well, I could hear an improvement, although it wasn't anything like as dramatic as the plots suggested. My perception was mainly of a (worthwhile) reduction of a slightly boomy and ringy frequency range between 50Hz and 70Hz. Low bass instruments and kick drums sounded a little tighter and more controlled and, as a direct result, the mid-range sounded a little more open and transparent. I also noticed a slightly wider stereo image across the higher frequency range, which I think was probably down to some correction for HF reflections from a stone fireplace. I'm sure that similar or better LF correction could be achieved by installing more traps and absorbers, but I've already reached my 'domestically acceptable' limit there! So I'd say the ARC correction was making a subtle but worthwhile improvement to the overall balance.

Moving to the office — which is a completely untreated acoustic environment with poor speaker placement and more reflective surfaces than a mirror ball — the results were rather more dramatic. As I mentioned earlier, this setup has a tendency to be bass heavy and noticeably lopsided because of the bookshelf at the side of the desk forming a corner, and the imaging isn't too hot because of all the reflections.

After running the measurement software, the plots revealed the peaks I was hearing, with a much larger peak on the right-hand channel, just as I'd expected. There was also clear evidence of desktop reflections causing response dips in the mid-range, and although the displays are heavily smoothed and not very precise, they certainly hinted at the problems I knew to be present.

The 'After' curve looked impressively flat by comparison, and listening to the results I have to say I was astonished at what I heard. The boominess had virtually gone, as had the lopsidedness. The mid-range sang with improved clarity and the stereo imaging was noticeably better too. The ARC software hadn't turned these cheap and tired stereo speakers into state-of-the-art mastering monitors, but the results were far beyond what I would have thought possible for those speakers in those positions! It was as if the rear and side walls and table top had been removed (or covered in foam!). To check that this wasn't a fluke, I measured the room again, using different pairs of test positions over a wider listening area. The response charts came out virtually indistinguishable from the first test, and the improvements in monitoring quality were the same.

Acoustic Alchemy?

All in all, the ARC software does seem able to bestow some improvements upon less than perfect monitoring systems and rooms — although I think it fair to say that the better the room to start with, the better the end results will be. ARC is no substitute for proper acoustic treatment, but it can maximise performance and reduce minor response irregularities very well, and with negligible quality degradation. But then, you can buy an awful lot of acoustic treatment for the price of the ARC software! SOS

information

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Mix Rescue

This month we explore a mix in which distortion, reverb and modulation effects provide powerful alternatives to EQ.

Mike Senior

Steven Separovich records under the artist name column:inches, and sent in his track 'Aftershow' because he felt that it was too one-dimensional and wanted to see if we could achieve more space and depth within the mix. Many aspects of his original mix were already pretty good, with a sensible arrangement and most of the tracks balanced well, but I agreed with his concerns and felt that the sounds seemed to be coming across as a bit anaemic and lacking in character. On a more practical note, the lead vocal line (provided by Tina Norden) was vocoded in such a way that the lyrics were getting lost, and this also needed attention.

Attacking The Kick

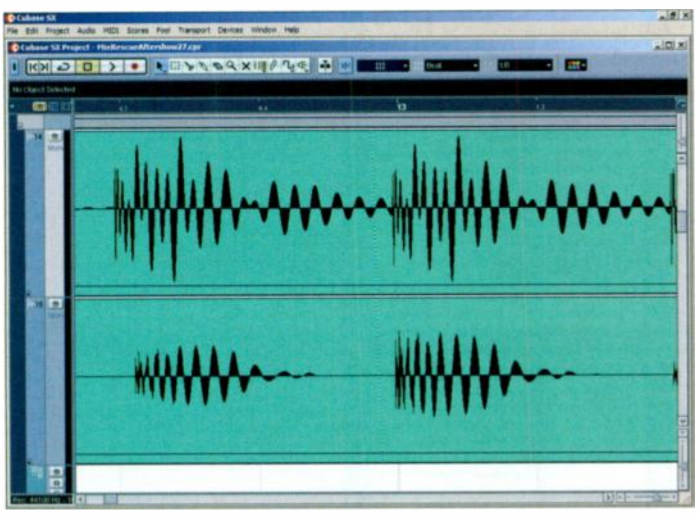
The most important element of the track seemed to me to be the drums, and the kick in particular. The drum tracks comprised two kick-drum samples (which played the same part most of the time, giving a layered sound), snare and clap samples (again playing the same parts), and a hi-hat sample. Listening to the two kick-drums playing on their own, it was apparent that they weren't always playing at the same time, but were drifting in and out of phase. I had this problem with Richard Campbell's mix back in *SOS* September 2007, and the result is that the tonality of the composite sound shifts in a complex way from beat to beat. There was the additional problem here that the gap between the two samples was occasionally wide enough to give a kind of flamming sound, which made the mixes sound less punchy and also affected the rhythmic feel of the track. Being reluctant to embark on the rather tedious task of lining up all the kick-drum hits, I initially tried to make the best of things with processing, but the sound simply wouldn't shift into focus and in the end I was forced to admit that phase-matching the hits manually would be the only remedy.

The low-end punch of one of the two kick-drum samples was improved with a low-pass-filtered Digital Fishphones Dominion send effect.

The mix of the two sounds didn't seem to have quite enough low-end attack, so I turned to a technique of Jack Joseph Puig's that I'd read about in *SOS*'s November 2007 feature on Fergie's 'Big Girls Don't Cry'. He split a track to two channels and used the Waves TransX Wide plug-in to isolate specific frequencies of the attack portion on one of them, mixing it with the unprocessed track. I used to do a similar parallel-processing trick in the analogue domain with SPL's Transient Designer, but hadn't yet got round to implementing the idea in my sequencer, so I was keen to see if it might help.

Of course, not everyone has the Waves plug-ins, and I turned instead to Digital Fishphones' freeware Dominion plug-in to provide the Transient Designer-style processing — so if you like the way it sounds in the audio files, you can easily try out the technique. One thing to bear in mind, though, is the issue of latency compensation. Some sequencers, such as Pro Tools LE, don't automatically compensate for plug-in latency. Most others now do, but even then you can face problems with some plug-ins; your sequencer may not be able to compensate for the latency of Digital Fishphones' freeware plug-ins, for example, as they don't seem to declare their latency to the host application, and this can result in phasing problems. Cubase SX2 has automatic delay compensation but wasn't able to solve this issue — although, fortunately, I was able to find a workaround.

One of the two kicks (I called it Kick 1) seemed to have a more solid attack than the other, so I set up the parallel Dominion processing on that one, boosting the attack level and cutting the release level to make a punchier sound. Because I was after more



Here you can see the original waveforms of the two different kick-drum samples. It's clear that they are drifting in and out of phase with each other. The resulting phase cancellation made it impossible to arrive at a consistent sound, so Mike had to edit them back into phase before processing.

low-frequency punch, I set the attack length to its 100ms maximum (shorter lengths don't let as much of the low end through) and then low-pass-filtered the track, nominally at 7kHz, but with such a gentle slope that it was already rolling off at 500Hz. Mixing the processed and unprocessed tracks together, I cut another 3dB on the processed track using a narrow peaking filter at 125Hz, just to match the tonality of the attack sound a little more to what I wanted.

I turned to the other kick sound (Kick 2) to add weight overall, boosting over a one-octave range with a peak of 6dB at 30Hz, but combined with a high-pass filter at 20Hz to avoid any useless subsonic peaks. The combination of the two kick drums now worked better for me, but I still felt that Kick 2 didn't sound quite powerful enough: it sounded a bit too 'digital', with too much click and not enough body. Some soft clipping from GVST's GClip plug-in improved this, though, after which I moved on to the rest of the drum parts.

EQ and a very short SIR reverb were used to adjust the hi-hat sample's tonality, while two send effects widened the stereo picture, the first a chain of MDA's Detune and Cubase's internal Double Delay, and the second a chain of Cubase's Mod Delay and GVST's GStereo.

Snare, Claps & Hi-hat

The snare benefited from the same parallel processing technique, although I also hard-clipped the output of the Dominion track in this case (as I used to with the SPL Transient Designer — and as it appeared Jack Joseph Puig was also doing on the Fergie track) to contain the scope and adjust the tonality of the added attack spike. No EQ was then required for the snare, so I faded up the claps. These lacked edge, and were also very narrow-sounding, so I applied a very small amount of heavily-driven Storm Studio Exciter to give them a bit more attitude and then applied my usual stereo-widening send effect — 12ms and 15ms delays in the left and right channels respectively, combined with opposite five-cent pitch-shifts.

The hi-hat track needed a little more work, because the raw track sounded very hollow and hissy, which meant that it didn't really help fill out the track very well. This was partly an EQ problem, but partly that the whole character of the sound needed more complexity, so I decided to bring in a technique I'd recently read about in Alex Case's *Sound FX* book: applying a very short reverb effect without pre-delay, not to add ambience but to change the tone colour of the sound.

I opened the SIR convolution plug-in and, as I was after something with raw personality and lots of presence, headed straight for my collection of spring reverb impulse responses. A little searching turned up one which filled things out really well, particularly in the mid-range, even though I'd reduced the impulse file's length from its original 3s to 0.22s in order to avoid any hint of a reverb-like effect. Some careful EQ was still needed to de-emphasise the hissiness from 2.7kHz upwards, but a few decibels of cut with two fairly narrow peaks and a high shelf made a fair job of it. Some of the previous stereo widener and an additional quarter-note stereo delay completed the picture, the latter widened a little using GVST's GStereo multi-band MS matrix plug-in.

Layered Basses

The main bass guitar part comprised two layers, both overdriven versions of the same performance, and this part underpinned the mix during the majority of the song. The less overdriven of the two sounds didn't really sound meaty enough to me, and neither did it have enough bite, despite having been passed through a Tech21 Sansamp distortion. My solution to the first problem was to set up a send to Jeroen Breebaart's Ferox tape-emulation plug-in, turning the high cutoff control down to 100Hz to isolate only the low end and then pushing the Saturation up to 100 percent. The extra hardness in the sound came courtesy of another send, this time to MDA's little Combo distortion plug-in, operating in its Radio mode: the small-speaker simulation added in just the kind of high-mid emphasis I wanted.

The other layer was completely mashed with a hard fuzz effect, and as such was very harsh-sounding. While I figured that Steven wanted a fuzz-style sound, it also needed to be kept under control, to avoid it trampling over the other parts to come, so I bracketed the track using high-pass and low-pass filters to leave a region centred on 2kHz, adding it in judiciously with the other bass sound, which was now providing the guts of the combined sound.

The main bass guitar sound is replaced by a different mixed bass sound, made up of three separate synth layers, for a short section in the middle of the song, but this needed very little work. I compressed two of the layers to try to keep the mix of the sound consistent in the face of filter modulation on these parts, and also high-pass filtered two of them to keep the low-end region solid and free of phase cancellations. Feeling that the combined sound lacked a little warmth compared with the main bass part, I also set up a send to the bass-guitar's Ferox plug-in from one of the parts and low-pass-filtered another at 2kHz. All very simple stuff, really, but you often don't need much processing when you have control over the balance of the different layers of a sound like this: it is when several layers are already mixed to a single track that you tend to encounter more problems.

I chose to go slightly easier on the high-pass filtering with the bass sound that opens the track (and recurs for various drops in the arrangement later, too) than Steven had chosen to do, putting the cut-off of Cubase SX2's Tonic plug-in at 460Hz, albeit with fairly high resonance and drive settings. I figured, however, that it made sense to narrow what was originally a stereo track into mono to enhance the arrangement contrast.

Fattening Up The Synths

With the drums and bass now working together, I began adding in the rest of the arrangement parts. There were four synth parts to mix in, and these all seemed a little lifeless to me, so more processing was required to resuscitate them. The rhythmic synth that you can hear most clearly at 0:53-1:07 probably took the most work, because I wanted to give it much more sustain — it was pretty much inaudible in the original mix because of masking by the kick drum. Limiting it helped a little, but a good dose of distortion from Craig Anderton's Quadrafuzz and from GVST's GClip were still needed to give more thickness to the sound. Being able to adjust the drive in different frequency bands, as you can in Quadrafuzz, is really useful for concentrating added harmonics into those parts of a sound where you need them most. That said, I did need to slice out a good chunk at 220Hz with EQ to get the tonality right in the mix. Another trick I used to make this part more audible was panning

Rescued This Month...

Steven Separovich and Tina Norden make up column:inches. Their track was put together using Apple Logic Pro 7.2.3 and Native Instruments' Vokator, layering EVOC20 and Vokator together to achieve a layered vocoder sound. Although Steven used live vocals and instruments as well as synths, he deliberately tried to blur the boundaries between them on 'Aftershow' by making the guitars and vocals sound more like synths, and synths sound more like guitars. You can contact the band at houseofsandn@btinternet.com.

Steven Separovich.

A combination of two send effects was used to process the main bass-guitar sound: an instance of Jeroen Breebaart's Ferox tape-emulation plug-in, saturating only low frequencies below 100Hz, and an MDA Combo amp-simulation plug-in, set to its Radio mode to add upper-mids.

the track to the right and then using a quarter-note delay panned to the left, doubling the note rate and thereby getting at least every second note out of the way of the kick-drum hits.

My first step with the lead synth that arrives at 0:37 was immediately to high-pass filter at 215Hz, getting rid of a lot of redundant low end that was simply clouding the mix. An instance of Silverspike's Ruby Tube put some hairs on the sound, and another short reverb impulse response (only 0.05s) added complexity, giving a much more satisfying result. To enhance the stereo movement inherent in the sound, I sent to two further effects: a Kjaerhus Audio Classic Phaser plug-in and an instance of Cubase SX's rotary-speaker simulator, the latter set up via its virtual Leslie miking positions to maximise the width of the stereo image.

The rhythmic synth at 1:25 sounded a bit lazy to me, so I made it spikier with another instance of Dominion, feeding it on to GVST's GClip to keep the peaks under control. Another MDA distortion send, again on the Radio setting, improved the presence and bulked out the sound's spectrum in a pleasing manner, after which the existing phaser and a further Kjaerhus Classic Flanger pulled the sound more towards the edges of the stereo spread and complemented the metallic timbre.

Synth hits like the pitch-dive that first appears at 1:47 can be difficult to mix — even when they're balanced correctly they will often sound rather small and unimpressive. The way around this is to give them a sense of real space, which I did in this case using Silverspike's room-ambience simulator Room Machine, mixing in just enough to do the job without giving an obvious 'effect'. The Kjaerhus phaser and flanger also came in handy here, to widen things a little.

Hear The Differences For Yourself!

Find out exactly what difference Mike made to Steven's mix by listening to the before and after audio files, available for download as WAV or MP3 from the *SOS* web site, at www.soundonsound.com/sos/mar08/articles/mixrescueaudio.htm. As well as the full mix, there are files that show the changes made to the individual sounds within the mix.

Guitars & Vocals

The guitar tracks provided two processed layers based on the same performance, but no matter how I combined them they sounded rather uninspiring. So I threw caution to the wind, bussed them to a single channel and selected something mad-looking from my plug-in collection: Betabugs Flo Fi. Who knows what it really does (it sounds like a cross between a ring modulator and The Wicked Witch Of The West), but within a few minutes of twisting controls I had something more engaging that also sounded more synthetic, in line with Steven's artistic vision for the track. Another curtailed reverb impulse from SIR (the inside of a coffin this time, apparently), in tandem with some tape saturation effect from Ferox, smoothed a few of the rougher edges left by Flo Fi. The overall sound now had real character, but it was also way too bassy for its place in the mix, which meant that I had to high-pass-filter at 400Hz and cut a further 6dB at 500Hz with a peaking filter before it fit around everything else going on, particularly when the texture reached its fullest at 2:08.

Bar a few little reverse cymbals and a similar white-noise-based sound, which I just widened with some sprinkles of flanger, phaser and reverb, that left just the vocals to attend to. I'd normally attend to the vocals earlier than this in the mix process, but I left them until the end on this occasion. I didn't feel that the part was musically more important than any of the other synth or guitar parts, so didn't need any more space left for it, whereas I knew that making the lyrics intelligible was going to be the real challenge and it would only be

Silverspike's Ruby Tube and Christian Knufinke's SIR added complexity to the main lead synth part's frequency spectrum, while Kjaerhus Audio's Classic Phaser and Cubase SX's in-built Rotary plug-in provided extra stereo interest.

Vocalist, and one half of column:inches, Tina Norden during a recording session.

Remix Reactions

Steven: "One of the key things I'd like to have achieved with my mix is more 'three-dimensionality'. Partly, I feel that my current studio setup didn't help the mix. I'm limited to Logic's plug-ins at the moment, and would have ideally liked to mix my track on a range of high-end hardware and software instruments. I also have very small monitors, which don't really provide the best picture of the track. I do, however, realise that equipment is not necessarily the key to a good mix, and that an understanding of the mixing process is important.

"On the whole, I was blown away by Mike's mix — his interpretation of the track was spot on, and he definitely headed in the right direction. Initially, I was most impressed by the 'bigness' of the sound Mike created. The track has more width and depth, and the individual sounds appear richer, fuller and more clearly defined and separated in the mix. I was also impressed that Mike managed to make the vocals more intelligible (through a careful mix of wet and dry sounds) and also address the timing issues associated with the vocals. I can't wait to read how he achieved his results!"

possible to judge this with the whole arrangement to compete against.

As I'd suspected the first time I heard Steven's original mix, there simply wasn't enough information in the vocoder tracks to allow the words to cut through — the high frequencies in the vocoder's carrier signal couldn't sufficiently support the vocal formants and enunciation. Anticipating this, I'd asked Steven to include the original unprocessed vocal that he'd used to drive the vocoder, and resolved to mix this in with the vocoded signal, to get the lyrics coming through more clearly. In this context, all the vocoder tracks needed was some level control (courtesy of MDA's little Limiter plug-in) and a send to another tiny reverb impulse response to give them more of a stereo feel — and I also took some high end out of the reverb return to darken the tonality a touch.

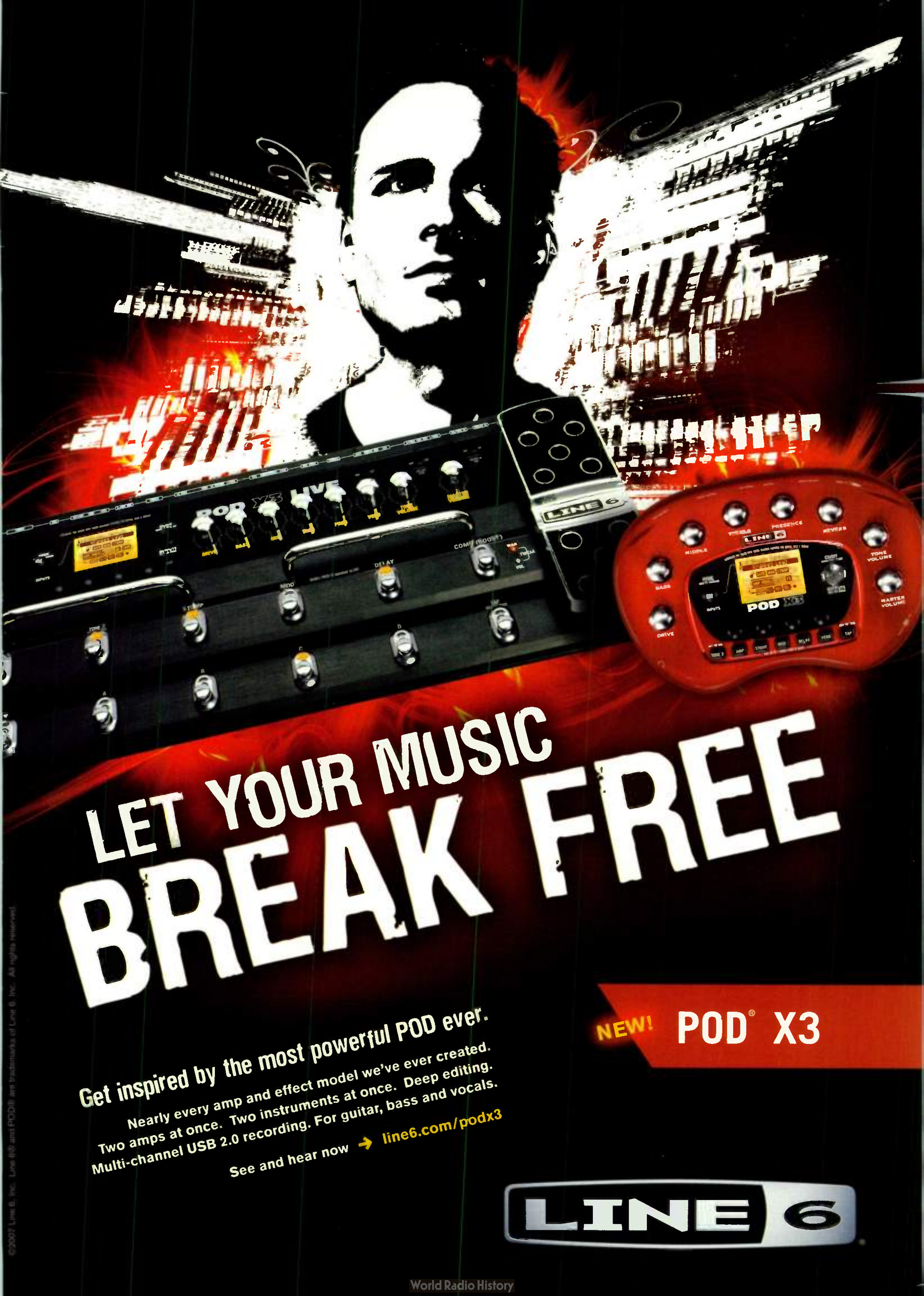
The dry vocal parts were high-pass filtered to isolate the region from about 1kHz upwards — where the formants really start to get going — and then heavily limited to bring up all the details in the performance. Because the vocals sounded a bit 'stuck on' to start with, I also treated them to some of Cubase's Metalizer process, as well as a shovelful of stereo widener, and this lent them a diffuse, synthetic flavour, which blended better with the sound of the vocoded tracks. Another thing that was making the dry vocal difficult to blend was that, although the part was basically spoken, it still had enough pitch in it to clash with the track's key at a number of points. Setting about the vocals with some ridiculously fast and abusive pitch-correction, courtesy of GVST's GSnap, I managed to nail it to a consonant pitch most of the time, dragging the body of the vocal sound further back into the mix compared to the elements responsible for intelligibility. Result: I could fade the intelligibility up higher without the other aspects of the vocal performance taking up too much space in the track.

Once the vocals were in place, I bounced down the mix as a 24-bit WAV and re-imported it into a new Cubase Project, so that I could bring up the detail a little more using a bit of multi-band compression from Cubase's internal Multiband Compressor plug-in. I set up the plug-in as a parallel process, a configuration which lets you add a lot more background detail without reducing the punch of the drums. A few decibels of extra loudness were gleaned from Buzzroom's Buzz Maxi, without too many side-effects, and I also let the tops of the drums clip a little, as (judging by his original mix) Steven wasn't averse to this tactic. Referencing the mix against a few well-known tracks made me pine for more low end, so I also added in a couple of decibels of shelving boost at 50Hz for this, before sending off a draft to Steven to get his views. Although he was pleased with the new sound, he had some concerns about the timing of the vocal parts, now that the dry vocals were more prominent, so I went back into the original mix project and spent a while editing syllables around to tighten that up for him before creating a final version.

Tina Norden's original vocal recordings were mixed in with the vocoder parts to improve intelligibility, but not before they were high-pass filtered and limited to emphasise the vowel formant and consonant details. Cubase's Metalizer matched the timbre of these vocals better with the vocoded tracks, while some extreme pitch-correction from GVST's GSnap made some of the pitched elements more in tune with the track, thereby pulling them more into the background.

When EQ Isn't The Answer

One of the things that Steven professed himself most keen to learn more about was EQ, but it was pretty clear to me from the outset that EQ wasn't really the key to improving on Steven's original mix, because EQ mainly just adjusts the frequency balance of a sound — if the character you want from a sound isn't there in the first place, no amount of cutting or boosting frequencies can reveal it! In an arrangement that's quite sparse like this, each sound needs to have a lot of body to fill out the overall production, and I hope I've been able to demonstrate how distortion, modulation, and super-short reverb effects can be more effective alternatives; they meant that I used comparatively little EQ for this mix, in fact. I was also pleased to rediscover the usefulness of Transient Designer-style parallel processing, and will certainly be returning to that in future... SOS



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Novation Nio

USB Audio & MIDI Interface

Tom Flint

Novation's Nio offers a slightly unusual perspective and a generous collection of effects and processors, but does it have what it takes to stand out from the audio interface crowd?

There are, it seems, countless small Firewire and USB audio interfaces on the market at the moment, so manufacturers are being forced to work very hard to come up with designs and features that will set their product apart from the rest. At first glance, the Nio appears to be a fairly standard two-in, four-out audio and MIDI interface, connecting to both Apple Macs and PCs via USB, but there is more to it than that.

Most significantly, the Nio is bundled with a carefully selected bank of 20 software effects, which are linked to the hardware using Novation's Direct FX technology. All the effects function from within a stand-alone program called the Nio FX Rack installed on the host computer, which helps the effects operate at fairly low-latency settings. The hardware itself offers some FX Rack-related monitoring controls so that, to a certain extent, the effects feel as if they are part of the hardware rather than remotely operated plug-ins. In keeping with the idea, Novation have modelled the Nio to look like a guitar effects pedal, right down to the pair of rubber gripper pads glued to the underneath — just as you find on the majority of stomp boxes. Even its size is comparable to a typical dual-switch pedal!

As a whole, the effects — a number of which are derived from Novation's own Supernova II synthesizer — form a comprehensive set of processing tools. Focusrite's current ownership of the Novation brand has resulted in the FX Rack menu including several of the parent company's

Alternatives

The Nio faces some strong competition from a range of products, many of which are already quite well established and have unique selling points of their own. The long list of current USB alternatives in a similar price range includes the Lexicon Lambda and Omega, Line 6 TonePort UX1 and UX2, Alesis IO2, Tapco Link USB, M-Audio Mobile Pre and Fast Track Pro, Emu 0404 and 0202, Tascam US144 and US122L, and Edirol UA25. Most, if not all, of the above demonstrate good signal-to-noise figures and key features of their own. The 0404, for example, offers soft limiting, S/PDIF I/O and up to 192kHz operation. Of all the above, the Line 6 products are probably most comparable, in that they offer similar guitar-related amp and effect modelling. Then, of course, there are numerous Firewire equivalents that are also quite comparable!

own well-respected signal processors and effects, and these are joined by a collection of physically modelled amplifier emulations and distortion pedal recreations programmed by Italian software specialists Overloud.

The DAW software Novation have chosen to bundle with the Nio is Ableton Live Lite 6, perhaps as a nod to the product's suitability for live performance. Live ships on the Xcite+ Pack DVD-ROM, along with more than a gigabyte of royalty-free Loopmaster samples and a number of attractive goodies including Waldorf plug-ins, a virtual version of Novation's own Bass Station synth and Arturia's Analogue Factory.

Operating systems supported include Mac OSX 10.3.9 or higher, Windows XP with Service Pack 2 and Vista 32/64. Only 256MB of RAM is required for both Mac and PC, so even fairly aged models should cope.

Ergonomics & Build Quality

The black parts of the Nio, formed from tough plastic moulds, and thick sheet-metal silver panels have been bonded neatly to create a very rigid casing. The various knobs and switches also feel very solid, adding to the impression that the device has been well engineered.

Despite the remarkably small size of the box, absolutely none of the controls are cramped together, as is usually the case when only the front and rear faces are used in rack-compatible layouts — few people outside of a studio are likely to want to rack up their small interface anyway. The eight knobs, for example, are spaced so widely that even the most sausage-fingered operator can turn each one without nudging its neighbour.

Hardware Overview

Although the Nio only handles a maximum of two input signals simultaneously, it is able to deal with guitar, line and microphone levels. On the rear, a pair of RCA phono connectors are provided for stereo sources such as CD players or sound modules, and there is also an XLR input with the option of 48V phantom power, obviously for use with microphones. In addition to this, the front face is home to a quarter-inch jack socket labelled Input 2, intended for guitar DI signals. This latter input has a three-position switch to adjust the impedance from something suitable for guitar to something more appropriate for the line and microphone inputs. Similarly, the XLR input has its own line/mic switch, but doesn't offer the high-impedance guitar setting.

Control over the input gain is provided by two rotary knobs that hardly require explanation. However, the rest of the controls are a little more complex, as they deal out ▶

SOUND ON SOUND

Novation Nio £149

pros

- Compact size.
- Great effects and processors.
- Spacious front panel layout and solid hardware design.
- Flexible signal bussing and facilities for dual headphone monitoring.

cons

- The second pair of outputs is commandeered by the FX Rack return monitor path when the effects are in use.

summary

A compact two-in, four-out audio interface with good monitoring options, offering a very high-quality selection of effects and processing, predominantly aimed at the guitarist.

a range of monitoring options designed to work together with the FX Rack software. It should be made clear, at this point, that the Nio's internal routing is not done via a software mixer/control panel, as is the case with some similar interfaces, but by the setting of switches on the unit itself.

Audio output is provided on four RCA phono sockets and a pair of quarter-inch headphone sockets. The four RCA phono outputs are best thought of as the stereo outs from two independent busses, as that is how they are configured. The monitoring is set up in such a way that the two busses can be auditioned — via the two headphone outputs — individually or together in a submix, the percentage you hear of each being determined by the position of the Mix knob found on the front panel. This is handy for DJs who want to monitor one song or loop while introducing another on the second bus. A further control makes it possible to fade from the aforementioned mix balance, whatever it may be, to a state where only the input signal is being heard. This dial is useful for overdubbing situations in which you want to hear the playback from the DAW recorded tracks and the input of the instrument part being performed at that moment.

Further complicating the routing options, however, are two switches which enable different signals to be routed to the RCA outs. For example, 1 and 2 can output either the identically numbered outs from the sequencer or the monitor mix established by the status of the Monitor section's hardware controls. Outputs 3 and 4 are more complicated still, providing exactly the same options as are available to Outputs 1 and 2, but adding a third choice in which the routed signal can be a separate bus from the DAW.

Such variables may sound a little convoluted on paper, but, simply put, they represent an equivalent setup to the aux bus arrangement found on the majority of small mixers, where one mix can be sent to the engineer's control room monitors and another to the mixer's master output.

The Nio's rear panel, sporting most of the unit's I/O. Input 2 and the two headphone sockets can be found on the front and three switches on the right-hand side set input impedance level and turn Input 1's phantom power on or off.

The two headphone outputs are placed on the front edge of the Nio, each with its own top-panel level control. Both outputs always monitor the same signal bus as one another, but can be set to listen to outs 1/2, 3/4 or the monitor mix.

Metering is provided by two banks of seven LEDs, the upper ones glowing red to warn of clipping. A two-position switch determines whether the meters are representing Inputs 1 and 2 or Outputs 1 and 2, but strangely there is no option to meter Outputs 3 and 4.

The only other important hardware features left to mention are the MIDI In and Out sockets and the all-important USB port, all of which are located on the rear panel.

Setting Up

Installing drivers and getting interfaces up and running is often far from straightforward but, on my computer at least, Novation's procedure proved to be very fast and trouble free. By the end, the host computer is equipped with the FX Rack program, accessed from its own desktop icon, and the simplest of control panel software, featuring alternative settings for bit depth and sample rates (16-bit to 24-bit and 44.1 to 48 kHz respectively), and a single 1-20ms buffer slider, simultaneously affecting both the input and output latency.

Music technology novices will be reassured to know that the manual documentation is thorough and well presented, although, annoyingly, it comes as a PDF only. There is, however, a printed Getting Started guide which concisely covers the basics of setup.

FX Rack

When the Nio is hooked up to the computer a magenta version of Novation's logo illuminates on the top to indicate that all is well. Pull out the USB lead and the light goes out — simple! Similarly, opening the FX Rack program immediately causes the monitoring section's FX LED to glow red, reminding users that effects are potentially in circuit.

The FX Rack itself has a very simple, single-page interface with its own meters and level controls at both the input and output

In The Nio's FX Rack

Below is a list of the modules in the FX Rack.

Overloud distortion and amp models (the products they are modelled on are listed in brackets):
- Green Overdrive (Ibanez Tube Screamer)
- Fat Pie (Electro Harmonix Bug Muff)
- Distorter (Boss DS1 Distortion Pedal)
- 70s Fuzz (Dunlop Fuzz Face)
- V-AC (Vox AC30)
- Tweed Twin (Fender Twin)
- Brit Rock (Marshall JCM900)
- US Valve Modern (MESA Boogie)
- Tweed Bass (Fender Bassman)

Plug-ins derived from Novation Supernova II:
- Filter
- Chorus
- Phaser
- Delay
- Tremolo

Focusrite plug-ins:
- EQ
- Compression
- Gating
- Reverb

NIO's bespoke plug-ins:
- Hot Tuna guitar tuner
- Smart Hum Killer

stages. By default, the rack is empty, but it can be filled up quickly by selecting a preset patch from a drop-down list. Alternatively, you can load up whatever processors and effects are required via the 'add FX modules button'. It doesn't seem possible to drag modules up and down in the rack, but they are easily muted or deleted, and obviously their parameters are all editable. It's also perfectly possible to have multiple instances of the same processor positioned anywhere in the series. Finally, it's possible to save the entire patch setup as a preset.

The FX Rack effect interfaces are all designed to look like their hardware forefathers, so the user is in no doubt about what they are supposed to be getting. Usually patch presets are over the top and only useable when severely edited, but that's not the case here. Novation have assembled a collection of presets that are immediately useable, although they do have a general bias towards rock.

Presets aside, the physically modelled effects are of high quality and should please almost every guitarist. Even the best virtual amps tend to sound less dynamic than the real thing — and the ones here do display that characteristic — but that's not always bad when recording. As for the Supernova and Focusrite reverbs and other modulation effects, they all sound very nice. Auditioning amps, processors and effects in various combinations is great fun when it's this easy and the components are this good!

This review isn't the place to discuss the merits of the Xcite+ Pack software in detail, other than to say that it is all great stuff to have, and makes a pretty decent dance and electronic music production system when combined with everything else on offer.

In terms of its routing, the Nio works very nicely indeed, and the rather complicated-sounding switching and bussing options make perfect sense in use, and provide a lot of monitoring flexibility. Perhaps the main drawback is that the second stereo bus is hijacked by the monitoring system when the FX Rack is in use, meaning that it cannot be used as an output, but as the effects are only really intended to be used during recording, rather than during mixing, this isn't a tragedy.

Unlike some other budget interfaces I've tested, the headphone amps demonstrate a pleasing lack of whine and are fairly noise-free. The mic preamp hasn't quite the punchy, accurate sheen of more expensive products of this type, but it handles vocals reasonably well all the same, and demonstrates a little more class than cheaper interfaces.

Conclusion

The Nio may qualify as a low-cost interface, but it offers a lot for the money, and the build quality is reassuringly good. The effects and processors will tempt many guitarists away from more straightforward product packages, I'm sure, as they're high quality and are organised in a program that's easy to use.

There's not much here to complain about. Perhaps the biggest compromise is the loss of the second bus when using the FX Rack, but this is still not really a huge problem. Some potential customers may wish for 96kHz operation, as having the option is sometimes useful, but at this level I don't think it's absolutely necessary.

Potentially of interest to guitarists as well as DJs, who can exploit the bus mixing output monitoring system, the Nio is undoubtedly a success. SOS

information

- **£** £149.95 including VAT.
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- **E** sales@novationmusic.com
- **W** www.novationmusic.com

Lennon's 'Because', and, best of all, a Clavioline, the charismatic, buzzy little monophonic organ used on 'Telstar'. For some reason the Clavioline samples aren't looped, so its 'long' notes are rather short.

The library's vintage Ludwig Downbeat drum kit exactly matches the one Ringo Starr played with the Fabsters. EWQL sampled it in a variety of styles and created nine kits which correspond to specific Beatles tracks: one of them is 'A Day In The Life', featuring a John Bonham-esque booming bass drum and stentorian, timpani-like ringing toms. Adopting the opposite approach, the manual proudly states that the 'Come To Drums' kit was recorded with 'tea towel on all drums', the advanced recording technique responsible for the dull, lifeless drum sound that marred many '70s rock records. Given this collection's fanatical attention to detail, we can assume that this it's the same type of tea towel that was originally laid on Mr Starkey's drum heads, and that the stain on it came from the same brand of tea the Beatles used to drink.

The icing on the cake (or the mango chutney on the poppadum) is the selection of Indian instruments the boys helped to popularise back in the day — sitar, some fine tabla drums hits and the surmandal zither used to great psychedelic effect on 'Strawberry Fields'. Unfortunately, the hypnotic bass drone of the tambura (used on several Beatles songs) is absent, but you can get a similar effect by slowing the sitar's attack and repeatedly playing a bass note with the sustain pedal held down.

For many people (myself included), the Beatles are musical gods, and even after 40 years their spontaneous, sparky, imaginative and groundbreaking recordings still sound like a breath of fresh air. Although these samples won't enable you to write like Lennon and McCartney, they should provide compositional inspiration to anyone with an ear for iconic '60s sounds.

California Dreaming: Recording Fab Four

If you're planning a sample library dedicated to recreating the heady sounds of the '60s, where better to record it than the studio in which Brian Wilson created the legendary 'Pet Sounds'? Founded by Bill Putnam in 1961, United Western Recorders at 6000 Hollywood Boulevard played host to America's biggest stars over the years, boasting a client list that included Frank Sinatra, Ray Charles, Elvis Presley, Phil Spector, the Mamas & the Papas, Tom Petty, Whitney Houston, REM, Madonna, the Rolling Stones and Elton John. In its heyday the studio was seen as America's Abbey Road. Being a huge Beatles fan, EastWest's Doug Rogers jumped at the chance to buy the studio complex when it came up for sale in 2006. Its acquisition spurred the decision to record the Fab Four sample library, a project Rogers had dreamed about for years. But accurately recreating the sounds was by no means easy, as Doug Rogers explains: "I did not want to start this project without having the right tools available... I knew I had to start by getting the original recording equipment, instruments and amps together."

The quest became obsessive. Rogers ended up scouring the world's second-hand markets for the ultra-rare EMI REDD and EMI TG12345 mixing consoles used on some Beatles recordings. Against all the odds, he also managed to locate a pair of valve Studer J37 four-track tape recorders, the type of machine used to record *Sergeant Pepper*. By the start of the sampling sessions, the producer had assembled "well over a million dollars worth" of rare period instruments, amplifiers, microphones, recording desks, EMI REDD 47 pre-amps, outboard equipment, Fairchild limiters, EMI RS124 modified Altec compressors and tape recorders.

The complex has now been re-named EastWest Studios. Its proud owner says of his new sound library, "These are just great sounds. The vintage tube equipment, developed with the finest audio components, provides a character that cannot be reproduced with today's digital equipment." What does he expect from Fab Four's users? "I don't imagine people are going to use this virtual instrument to make Beatles music; that wasn't my objective. I'm hoping they're going to use it to make new music." You can be sure the Beatles would have agreed with that sentiment!

Doug Rogers and Ken Scott at the vintage EMI TG12345 desk in EastWest Studios. Behind them you can see an EMI REDD 37 desk and a Studer J37 four-track tape recorder. And what's Doug reading? Could it be Kevin Ryan and Brian Kehew's *Recording The Beatles* book?

Ministry Of Rock

Fast forward several decades. The Beatles' reign is over, but by a strange quirk in musical evolution, dinosaurs once again rule the earth. Yes folks, we've entered the era of heavy rock, which first reared its shaggy head in the '70s and has refused to die ever since. It's interesting to compare the guitars, basses and drums on QL Ministry Of Rock with their equivalents on Fab Four — the sound has a more distilled and manic aggression, and the recording techniques are aimed at focusing that sonic fury.

MOR opens with a Ibanez seven-string instrument known as 'the ultimate death-metal guitar'. I made the mistake of playing it through headphones without first checking the headphone-amp volume level, and the blood pouring from my ears confirmed that the description is accurate. Be that as it may, the sound is so exciting that I found myself jamming enthusiastically on the instrument for several minutes before my body's defences kicked in and warned me to turn the volume down! Played through a Krank amp (a name which suggests terrifying loudness levels), the Ibanez sounds unbelievably heavy, especially down on its low 'B' string. A separate power chords program contains chugs and smashes of tremendous potency.

At the quieter end of the decibel range (assuming you haven't already gone deaf) you'll find a beautifully-recorded Gibson J160 acoustic guitar delivering a chromatic set of strummed chords. You can keyswitch between various chord types, including the sus 4ths and add 2nds often used in contemporary rock. There's a choice of long and staccato strums played with up- and down-strokes, and strummed deliveries played in a slightly more 'broken' style can be accessed by pushing up the mod wheel. When combined, these musical options will produce very realistic rhythm guitar tracks. Back in Loudsville, the classic 1970s

distorted sound of a Les Paul Deluxe sounds mighty strong when layered up into a Brian May-style 'guitar choir', and also works reasonably well for Holdsworthian faster-than-light solo lead lines. Play's 'auto-detect legato' mode helps smooth the transition between notes for that sort of quicksilver delivery. However, although it's a big improvement on the clunky sound of a series of disconnected initial notes played in quick succession, the resulting legato effect is not altogether convincing.

The guitar highlights go on and on — PRS and Fender Stratocaster guitars contribute some mad thrash rhythm samples, the Telecaster plays terrific 1950s palm-muted pizzicatos à la 'Shaking All Over' and there are oodles of iconoclastic heavy metal noises — plectrum scrapes, whammy-bar howls, ZZ Top squealing harmonics and so on. Without wishing to sound like an ad copywriter, even the most pusillanimous, knock-kneed, six-stone weakling can sound like a rock god with these guitar samples!

Metal Drum & Bass

Moving down the scale, MOR's basses provide suitably muscular support to the guitar front-line. Sampled bass guitars often sound too clean and polite, but that's not the case here — the emphasis is definitely on attitude, and if that means that some notes aren't played totally cleanly, that only reflects how rock bassists play in real life. Played fingerstyle and with a pick, a Kubicki bass combines a big, booming tone with a hint of edgy distortion. A Musicman bass borrowed from the Quantum Leap Hardcore Bass library also packs plenty of welly and wouldn't sound out of place in a Quentin Tarantino film soundtrack. You might prefer the Specter model for rock ballads (although it's advisable to keep it well away from firearms), while the trusty Fender Jazz and Precision basses will work with virtually any style of electric music.

MOR contains four drum kits: the Ayotte kit is the cleanest, while the Gretsch 'Black' kit (named after Metallica's album) has a more '70s sound, mainly due to its rather low-pitched snare. I enjoyed the Octaplus kit's open, ringing sound and liked the natural room ambience surrounding the Ludwig set, but felt all the drums could use some extra processing to help them compete with the death-dealing decibels of the guitars. You can mix and match the kit elements, which is very useful. Volume levels are controlled by a little bar to the right of each element's name — but unfortunately

Fab Four & Ministry Of Rock Instrumentation

Fab Four (13GB)

Guitars:
- 1956 Epiphone Casino
- 1957 Les Paul Goldtop
- 1956 Fender Stratocaster
- 1951 Fender Telecaster
- 1959 Gretsch Country Gentleman
- 1960 Gibson SG
- 1965 Rickenbacker 36012 12-string
- 1966 Gibson J200 acoustic
- 1966 Martin D28 acoustic

Basses:
- 1963 Hofner 500
- 1964 Rickenbacker 4001S

Drum Kit:
- 1960 Ludwig Downbeat

Keyboards
- Steinway B piano
- Hammond B3 organ
- Lowrey Heritage Deluxe organ
- Baldwin electric harpsichord
- Harmonium
- Clavioline
- Mellotron flutes

Miscellaneous:
- Sitar
- Surmandal (Indian zither)
- Tabla drums
- Cowbell, claps, tambourine
- Screaming girls

Ministry Of Rock (20GB)

Guitars:
- Ibanez 7-string
- Les Paul Standard
- Les Paul Deluxe
- Fender Telecaster
- Fender Stratocaster
- Paul Reed Smith
- Gibson J160 acoustic

Basses:
- Fender Jazz 5-string
- Fender Precision
- Kubicki
- Musicman
- Specter

Drum Kits:
- Ayotte
- Gretsch
- Octaplus
- Ludwig Vintage

What do you need to get that Beatles guitar sound? Vox amps, Neumann mics and, of course, an authentic, '60s-style linoleum floor.

System Requirements

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

Both platforms require an iLok key (not included).

this doesn't have a numerical readout, which makes level-matching a little unpredictable.

Don't be fooled by the acronym: MOR is definitely NOT middle of the road. It's a screeching, howling, booming, clanging, clattering, bashing noise-fest that will fool your neighbours into thinking you have a tame death-metal band lurking in your studio. It's hard to imagine any rock library getting much more powerful than this, and bearing in mind the enduring popularity of the genre, I have a feeling it will be one of Quantum Leap's biggest sellers.

Graphic Novel

I don't usually give much thought to the visual appearance of virtual instruments, but I have to say that the Play user interface graphics are extremely attractive. The player changes its colours like a chameleon according to which library you're using, so if you open Fab Four and then load a Ministry Of Rock instrument, the UI obligingly performs a costume change from a dark grey Vox AC30 look to a more contemporary yellow-ochre and brown. The Play engine has a very usable selection of built-in effects: both libraries offer built-in mono delay, ADT ('automatic double-tracking', the fast tape-delay effect John Lennon loved hearing on his vocals), a good-quality (though CPU-hungry) convolution reverb and a five-stage amplitude envelope. MOR also has a low-pass filter, very handy for extreme timbral manipulation and synth-like effects. One slight design glitch is that the ADT and delay effects currently ignore the filter setting, which sounds rather weird! I hope EWQL will remedy this.

As the name suggests, the new Play libraries are designed for maximum playability and minimum technical fuss. The downside for inveterate tweakers like myself is that it's frustrating not to be able to (say) extend the bottom note of an instrument down by a semitone or two. The inability to alter the player's pitch-bend range is a more serious problem: a full bend up gives a pitch rise of just under three semitones, while a full bend down detunes notes by something less than a full tone. This inaccurate calibration means that if you layer a Play instrument with another manufacturer's instrument and perform a pitch-bend in either direction, the two instruments will go out of tune with each other. EWQL should address this.

But enough of such quibbles; overall, Play is a triumph, and I can unreservedly recommend both of the new libraries. Users will have a lot of fun with these instruments and the colourful iconic sounds on Fab Four will be an asset for composers who work on film soundtracks and TV ads. I look forward to more sonic mayhem from Quantum Leap, and live in hope that Doug Rogers' next project will bring us the sounds of Brian Wilson and the Beach Boys! SOS

information

£ Fab Four 284 Euros; Ministry Of Rock 357 Euros. Prices include VAT.
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Karl Heinz Stockhausen

Pioneer Of Electronic Music

Tim Whitelaw

Few individuals have influenced the development of electronic music as much as Karl Heinz Stockhausen, who died last December. We look back at his life and celebrate his many achievements.

As you glance over Peter Blake's iconic cover artwork for *Sergeant Pepper's Lonely Hearts Club Band*, take a closer look at the face fifth from the left on the top row. It is the face of Karl Heinz Stockhausen, the German composer of over 350 pieces of classical music, in his mid-30s in this photo, who died on December 5th at the age of 79.

For over 50 years, Stockhausen was a giant of the classical music world: iconoclastic, innovative, and often controversial. He was renowned for his refusal to accept conventional forms and boundaries, and his music was often conceived in outlandishly large terms — his seven-day-long opera *Licht*, 26 years in the making, will finally receive its first performance in 2008. But his influence and admirers extend far beyond the frontiers of classical music, and artists as diverse as Björk, the Beatles, Kraftwerk, Pink Floyd, Brian Eno, Frank Zappa, David Bowie and Miles Davis have all noted or paid tribute to Stockhausen's influence on their work.

Post-African Repetitions

At first, this might seem strange; after all, Stockhausen was a fastidious critic of popular music, complaining of its reliance on repetition and consequent predictability. In a memorable exchange published in *The Wire* in 1995, he recommended that Aphex Twin (aka Richard James) listen to more of his music "because he would then immediately stop with all these post-African repetitions". Richard James retorted that Stockhausen should listen to more Aphex Twin; "then he'd stop making abstract random patterns you can't dance to".

So how did Stockhausen, whose own music could ostensibly not be further from most popular music in its sensibility and scope, end up being admired by popular musicians around the globe — not to mention having a spot on the *Sergeant Pepper* album cover alongside such pop culture icons as Marilyn Monroe, Marlon Brando and Bob Dylan?

The answer lies in his prodigious output of electronic music. Throughout the course of his 50-year composing career, Stockhausen produced over 140 works employing electronics in some capacity, and many of those works (particularly those from the '50s and '60s) played a pivotal role in anticipating and helping to form what might be described as the grammar of electronic music. He was among the first to employ techniques such as sampling, directional sound, the blending of live and electronic performance, the complex analysis of acoustic sounds and the mimicking of their characteristics in electronic music, and much else besides. In the days when electronic music was largely uncharted territory, Stockhausen was one of its boldest and most influential pioneers.

The Art Of The State

To understand his impact, a little context might be helpful. Although popular music has largely led the way in producing innovative electronic music techniques and developments over the last 30 years or so, it wasn't always this way. Before the advent of commercial synthesizers, the expense of the equipment required for creating electronic music (not to mention the space needed to accommodate it) was so prohibitive that it largely confined experimentation to universities and state broadcasters — institutions which tended to patronise classical composers. At that stage, the union of electronic music, then based on tape recording techniques, and popular music, then based very much around live performance, seemed to be a far-fetched idea.

Therefore, in the postwar period, Europe's most technically advanced facilities were often run by state broadcasters: the WDR Electronic Studio in Germany, built in 1951, hosted many of Europe's electronic composers in the '50s and '60s. Later, the BBC's Radiophonic Workshop, established in London in 1958, would lead the way in British electronic music. In the US, meanwhile, the world's first programmable electronic music synthesizer — the room-sized RCA Mark II — was built at Columbia University in New York in 1957, funded by a massive grant from the Rockefeller Foundation. Suffice to say, in its infancy, electronic music creation was a serious and expensive business, beyond the reach, and probably the interest, of most pop musicians.

It was onto this landscape that Stockhausen emerged as a young classical composer in the '50s. Born in the village of Moedrath, near Cologne, in 1928, he became a student of musicology, philosophy and German literature at the University of Cologne in his late teens. Following the completion of his degree, he studied in Paris for a short time, before taking up a position at the then newly established WDR music studio in Cologne. He dabbled, as did most electronic composers at that time, in *musique concrète* — the art of creating musical pieces from recorded sounds and manipulating them via tape techniques.

But the limitations of this approach soon became apparent to Stockhausen, and he began to seek to expand the creative horizons of electronic music.

His first important electronic pieces — *Studie I* and *II* of 1952 and 1953 respectively — are sonic explorations using pure sine waves, sometimes reverberated, cut off or reversed. Although the pieces are fascinating artifacts of early electronic music, Stockhausen later admitted that they were constrained by both his command of the technology and the limitations of the technology itself. But they are undoubtedly milestones of a sort, in that they are among the first pieces composed by a musician using electronic sounds created from raw waveforms. The task of creating music in this way was complex enough that up to that time few musicians had attempted it, *musique concrète* being the preferred medium.

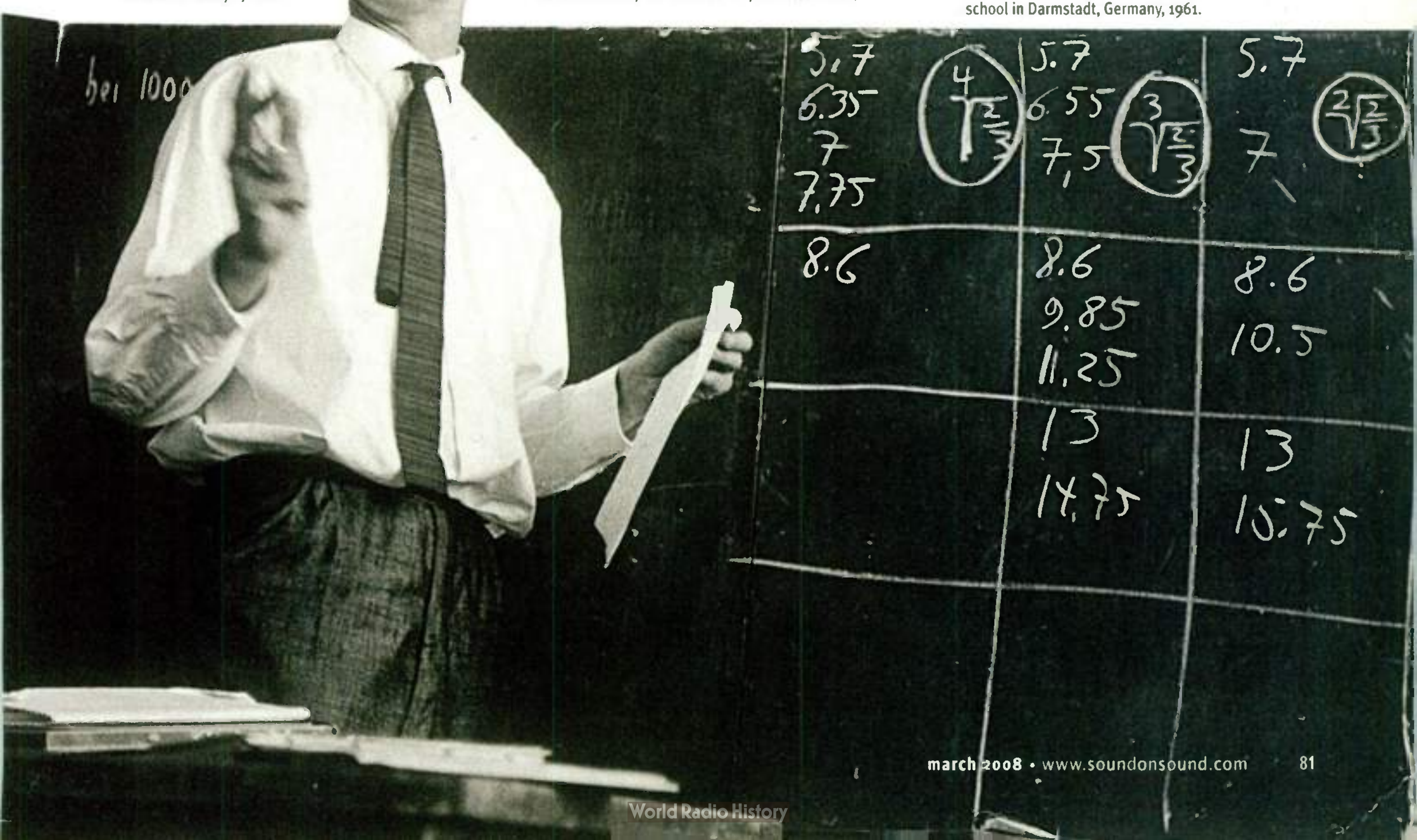
Making Contact

Stockhausen's first electronic masterpiece arrived in 1956 with *Gesang der Junglinge* (Song of the Youths) — apparently, Paul McCartney's favourite piece of his. Created at the WDR studios, it is a 13-minute work of beguiling complexity. It is built around 11 basic electronic elements (mainly sine waves, filtered and modulated in different ways, and electronic clicks) interacting with recordings of the voice of a boy singing (hence the title), producing some highly intricate and fresh-sounding musical effects. Significantly, it was the first piece that combined synthesized sounds with *musique concrète*, setting the purity and sterility of one against the familiarity of the other, a dramatic contrast then quite new in electronic music.

Stockhausen at work on the electronic elements of *Hymnen* in the WDR Electronic Studio, Cologne, where much of his most important work was completed.

The piece also represented one of the first musical experiments with spatial effects: creating the piece for five-channel tape, with each channel played back through a different loudspeaker, allowed Stockhausen to begin exploring the directionality of sound in performance, adding another dimension to electronic music performance, which he would develop further in subsequent works.

In 1960, Stockhausen completed *Kontakte* (Contacts), which would soon be ▶



Karl Heinz Stockhausen lectures on *Kontakte*, one of his most significant works, at the well-known summer school in Darmstadt, Germany, 1961.

regarded as a key work in the evolution of electronic music. It was among the first to combine electronics and live performance, employing a four-channel tape recording along with live percussion instruments and piano.

The piece's innovations are numerous. For example, Stockhausen wanted to be able to imitate the live percussion with his electronic sounds. To do this, he engaged in an incredibly detailed spectral analysis of the acoustic sound sources — drums, bells and the like — using their characteristics to shape the electronic sounds. The result doesn't attempt to mimic the sounds precisely, but uses their characteristics to come up with electronic doppelgangers for them. His distillation of the character of these timbres of metal, skin, and wood into electronic sounds remains incredibly impressive considering the means at his disposal, and anticipates the sound-shaping techniques that would help form much of the electronic sound palette before the advent of sampling.

Kontakte also further explored the directionality of recorded sound, this time combined with live performers. A four-channel tape with four loudspeakers allowed Stockhausen to pass his sounds around and across the audience in an elaborate and dramatic use of acoustic space that might be seen as an early precursor to surround sound.

Today, to hear *Kontakte*, even if only in its stereo reduction, is to marvel at its sonic complexity. The detail and intricacy of its sound world is stunning, even to ears accustomed to the limitless possibilities of modern sampling and synthesis. Try to imagine how the piece might have sounded to young musicians of the late '50s or early '60s, and you get some understanding of why Stockhausen began to attract attention from across the wider musical world.

His 1967 work *Hymnen* (Anthems) was particularly significant in this respect. A nearly two hour-long work for tape, *Hymnen* begins with scattered fragments of short-wave radio public broadcasts, which are gradually joined by recordings of various national anthems from around the world, as well as synthesized electronic sounds. The piece slowly evolves in to a sort of hallucinatory collage, with the radio broadcasts, national anthems and electronic sounds weaving in and out of one another. With its trance-like sound world and leftish political overtones, *Hymnen* cast its spell far outside classical music circles — in fact, of all Stockhausen's electronic works, it seemed to become the one pop musicians became most often enamoured of. Indeed, by the mid-'60s, many of the innovative popular musicians of the era were beginning to take note of the possibilities that Stockhausen's work seemed to unveil. And among his admirers were the most popular pop musicians of all: Paul McCartney and John Lennon.

A Growing Influence

It was perhaps fitting that the Beatles chose Stockhausen as one of the few musical figures to make the cover of *Sergeant Pepper's Lonely Hearts Club Band* in 1967, since it was the album that decisively shifted the artistic emphasis for bands out of the concert stadium and into the recording studio. 'Strawberry Fields Forever', with its abundance of overdubbing, tape delays and Mellotron flutes, is often quoted as showing Stockhausen's influence, and Stockhausen has said that John Lennon and he spoke often on the phone during that period. Despite the composer's oft-stated antipathy for popular music, he would go on to describe Lennon as "the most important mediator between popular and serious music of this century".

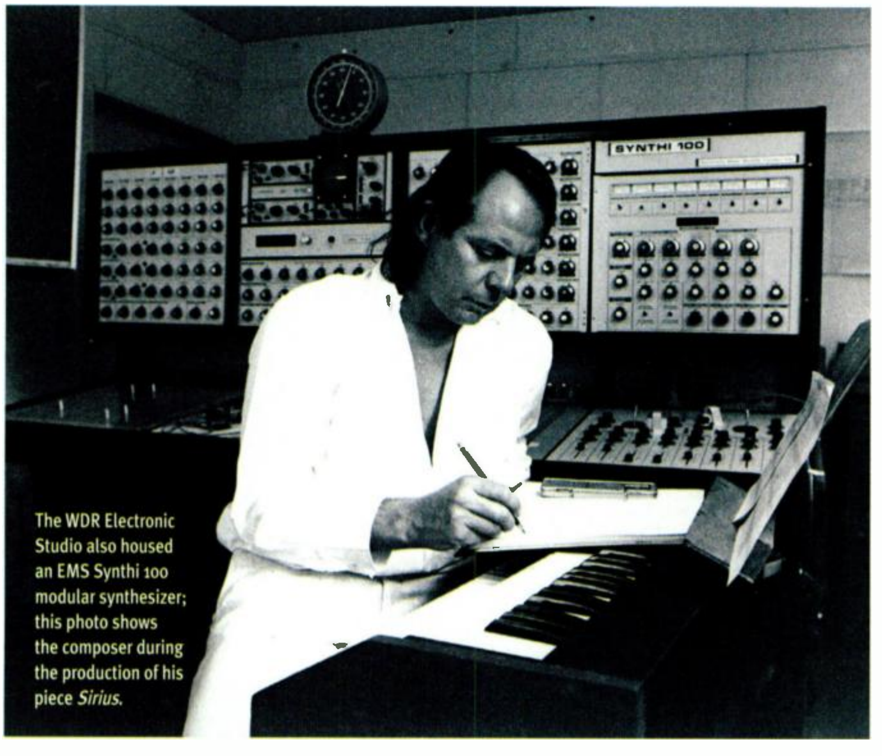
In Germany, meanwhile, so-called 'Krautrock' was beginning to make an appearance, typified by bands such as Kraftwerk and Tangerine Dream. Their music was formed from an eclectic fusion of British/US rock music and the influence of works by electronic music pioneers, principally Stockhausen himself. Indeed, Irmin Schmidt and Holger Czukay of the Krautrock group Can were in fact students of Stockhausen at university in the mid-'60s.

His influence was also felt in the world of progressive jazz. Miles Davis, whose later albums make extensive use of studio techniques, paid homage to Stockhausen's influence in his works. In his autobiography, he wrote that "I had always written in a circular way and through Stockhausen I could see that I didn't want to ever play again from eight bars to eight bars, because I never end songs: they just keep going on. Through Stockhausen I understood music as a process of elimination and addition." The collage-like quality of music from the 'Electric Miles' period was said to stem directly from his reaction to *Hymnen* and several of Stockhausen's non-electronic pieces.

Throughout the '70s, a string of artists including Brian Eno, Pink Floyd and Frank Zappa would acknowledge Stockhausen's influence on their increasingly innovative work. By this time, of course, the liberation of electronic music was well under way; the increasing availability of commercial synthesizers and advanced recording equipment helped to deliver electronic musical creativity to a much wider constituency — who wasted no time in taking it in more readily accessible directions.

More Helicopter!

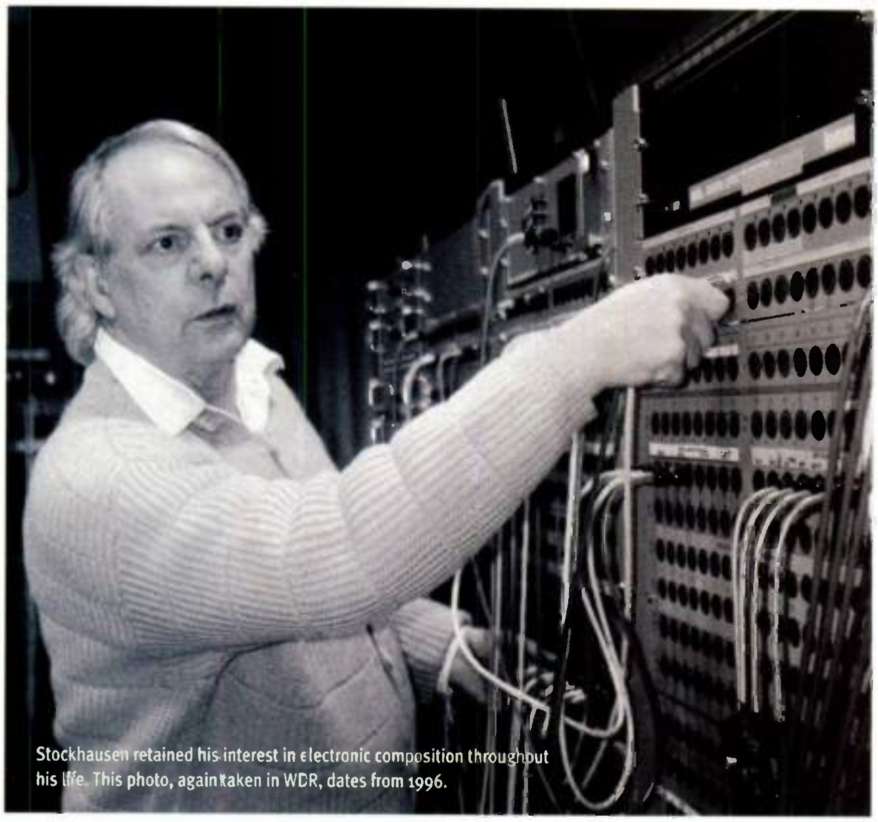
Stockhausen continued composing with electronics throughout his life, and in his later years he could often be found at electronic music festivals across Europe, attending performances of his own pieces or overseeing new works. His passion for



The WDR Electronic Studio also housed an EMS Synthi 100 modular synthesizer; this photo shows the composer during the production of his piece *Sirius*.

Further Investigation

If you want to find out more about the life, work and opinions of Karl Heinz Stockhausen, a good starting point would be a visit to his web site (www.stockhausen.org). This offers a range of resources, including an on-line shop selling CDs, scores, books and videos.



Stockhausen retained his interest in electronic composition throughout his life. This photo, again taken in WDR, dates from 1996.

pushing the envelope never seemed to dim; perhaps the most striking work of his later years is the *Helicopter String Quartet* of the mid-'90s, a piece which called for four helicopters, a string quartet, and swathes of loudspeakers, televisions and audio processing equipment. Electronically blending the music of the string quartet with the rotor noise of the helicopters, the piece seemed conceived to prove Stockhausen's theory that "any sound can become music if it is related to other sounds".

It's unlikely, of course, that the sampled beats of hip-hop or the studio experiments of rock musicians, or indeed the "post-African repetitions" of Aphex Twin, were what he had in mind in saying this. But whether he intended it or not, his was the path that helped lead the way. SOS

Sample Clearance

The *SOS* Guide To Copyright Law On Sampling

Using someone else's recording in your music without permission can lead to disaster. We explain the ins and outs of copyright law, and guide you through the process of clearing your samples.

Richard Salmon

To sample or not to sample? This is the question many a DJ, producer, and songwriter must grapple with on an almost daily basis. Sampling is fun, and in the era of the ubiquitous digital audio workstation, very easy to do. But is it always a clever move from a legal and business perspective? In this article we'll consider how to go about sampling within the law, how to avoid getting sued, and consider some of the pitfalls of falling foul of copyright law.

What's The Problem?

Sampling involves the incorporation of another sound recording into your own new record. A producer may sample an underlying element in a record — for instance a string or bass line, perhaps borrow a drum loop, or even lift several bars wholesale from a classic soul record — and write a chart-friendly melody over the top.

The creative act of sampling is nothing new. Much of the Beatles' late-'60s output owes a great debt to the sound-collage and tape-splicing artistry of production legend George Martin. Nor should sampling be a worry, when the primary source of the sample is self-created. Be it a vocal drone, birdsong recorded and cut up into your dance tune, or as in the case of the Nile Rogers' inspired vocal stutter, 'No... No... Notorious'. Rodgers had sampled Duran Duran's vocals and edited them into an immediate radiotastic hook, as he'd already done on 1984's pitch-shift sampled intro to 'The Reflex'. Modern-day producers such as Timbaland and Pharrell Williams aren't averse to incorporating their own homegrown beat-box elements into major airplay hits. The Williams-produced 'Rock Your Body', for instance, is liberally peppered with Justin Timberlake's own down-the-mic percussive elements.

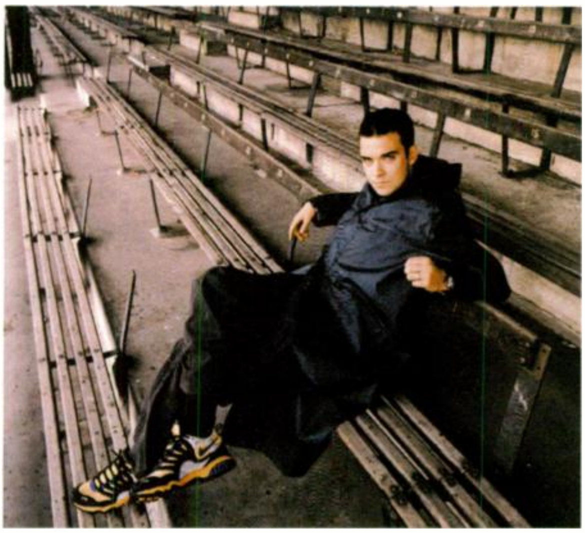
The legal headache, as far as the producer, artist or songwriter is concerned, stems from using another person's original sound recording without prior permission, since this constitutes copyright infringement. The act of sampling without permission infringes copyright in three distinct ways. Firstly, it is a breach of copyright in the original sound recording. Secondly, it is a breach of copyright in the underlying music and lyrics, and thirdly, it constitutes an unauthorised use of one or more of the performances in the original work, such as a guitar riff, vocal hook, or drum part. In addition, the moral rights of the original artist may be infringed, if sampling is undertaken in a way that the artist objects to, or if the artist isn't credited.

Sense & Substantiality

In UK law, under the Copyright Designs and Patents Act 1988, in order for infringement to take place a 'substantial part' of a copyright work must have been used. Substantiality in UK law differs somewhat from its US counterpart, the doctrine of 'substantial similarity'. Moreover, US copyright law permits the defence of 'fair use', which has been invoked in a number of recent cases, although not always successfully (see 'Case Dismissed' box).

Regarding the question of whether a 'substantial part' has been copied, the UK case of Produce Records Ltd vs BMG Entertainment Ltd (1999) established that a 7.5-second sample of 'Higher And Higher', a track originally recorded by the Farm and owned by Produce Records, constituted infringement when appropriated by veteran Latino duo Los Del Rio, for their summer hit 'Macarena'. BMG settled the case out of court, thus avoiding trial, with the major label appearing to concede that Produce had an arguable case.

In Ludlow Music Inc vs Williams (2000), a two-line lyrical 'sample' of the song 'I'm The Way', written by Loudon Wainwright III and published by Ludlow Music, formed the basis of a copyright dispute, when Robbie Williams used very similar lyrics in his own song 'Jesus In A Camper Van' — there was no use of the original recording, so the dispute only concerned copyright in the song itself. At considerable expense to the record label, the judge ruled that the Robbie song be removed from all future pressings of his album *I've Been Expecting You.* Robbie also lost out on 25 percent of the publishing income on 'Jesus In A Camper Van' to Ludlow Music, a figure said to be somewhere in the region of £50,000.



Robbie Williams blends into the crowd at a Stoke City home match.

Despite the fact that the judge actually described Loudon Wainwright's own song as a parody of an earlier Woody Guthrie song, he was of the view that the extent of the copying was substantial, "although not by much". Compare the following lyrics and decide for yourself! Loudon Wainright's lyric goes:

Every Son of God gets a little hard luck sometimes, especially when he goes round saying he is the way.

The Robbie lyric went:

I suppose even the Son of God gets it hard sometimes, especially when he goes round saying I am the way.

Over the pond, the recent US decision by the Sixth Circuit in Bridgeport Music vs Justin Combs Publishing (2007), confirmed copyright infringement liability against Sean 'Diddy' Combs and his Bad Boy record label. This case concerned the title track from The Notorious BIG's 1994 album *Ready To Die*, which sampled the song 'Singing In The Morning' by '70s funk outfit the Ohio Players. The Biggie record sampled just five seconds of horns from 'Singin' In The Morning', but very bad boy Diddy had failed to obtain a licence for its use.

The song's copyright owners Bridgeport Music and Westbound Records sued for infringement, with a US jury awarding $733,878 in damages to Bridgeport, and punitive damages of $3.5 million to Westbound. Allowing common sense to prevail, the trial judge overturned the award, ruling instead that Bridgeport should receive $150,000 in statutory damages, with Westbound receiving $366,939 in actual damages. Still, that works out at over $100,000 per second of music — a brief yet very costly mistake!

Dr Dre, another big name from the US urban scene, has also spent his fair share of time in the legal spotlight. In 2003, Indian composer Bappi Lahiri and Saregama India Limited sued Dre and Universal Music for $500 million, over the use of an unlicensed sample on 'Addictive', the debut single from Truth Hurts' album *Truthfully Speaking*. Dre was also slapped with an injunction preventing the continued sale of the record, which by then had already shifted over 200,000 copies.

Dre and his producer DJ Quik had used a sample of an old Hindi song, 'Thoda Resham Lagta Hai', without permission from the Indian copyright holder Saregama. The plaintiffs alleged that in addition to 'borrowing' the distinctive vocals, Dre and Quik had helped themselves to the hook, the melody and the rhythm.

Dre has also recently incurred the wrath of one Madge Ciccone. Madonna's publishing company are up in arms over Dre's alleged copying of her 1983 hit 'Holiday' on the single 'Not Today', featured in the film *Barbershop 2*. They are demanding a refund of £7m from Dr Dre and his collaborators, artists Mary J Blige and rapper Eve. The publishers allege that parts of 'Not Today' include "several obvious instances of reproduction".

Know Your Rights

Not all court cases go in favour of the copyright holder (see 'Case Dismissed' box), but it's always advisable to obtain a licence and permission from the copyright holder before sampling another's work. Moreover, what may be acceptable in one country may constitute an outright infringement of copyright in another, and defences such as 'fair use' are not universally available everywhere. This needs to be borne in mind when releasing records internationally. So which rights should you be concerned with when sampling?

Even if you believe you can process, edit, or otherwise disguise a sample ▶

Case Dismissed

Not all copyright disputes over uncleared samples are resolved in favour of the sample's copyright holders. One example is the US case of Newton v Diamond (2003). In this case the Beastie Boys had actually obtained permission from ECM Records to sample a six-second, three-note sequence from James Newton's flute recording *Choir*. The Beasties then incorporated the sample as a loop into their song 'Pass The Mic',

which featured on the album *Check Your Head*. Unfortunately, the composer of the tune, James Newton, sued, as he hadn't given his permission for use of the underlying composition.

On appeal, the court confirmed an earlier ruling that no infringement had taken place. The court was of the opinion that the use of the sample was minimal, the two records weren't substantially similar, and also that the public wouldn't recognise any appropriation of Newton's composition. (Though it should be stressed that recognition alone is no legal barometer of whether another work has indeed been copied.)

Other defendants on the receiving end of sample infringement claims in the US have been able to rely on the defence of 'fair use'. Fair use is a doctrine not recognised in UK, which permits copying for the purposes of criticism, reporting and review. The aim of the US legislators who enshrined this in law in 1976 was to allow authors to build upon, and transform existing works, but without the requirement of buying a licence to do so. The rights accorded to the copyright holder needed to be balanced with the broader cultural benefits of allowing artists to borrow from, re-work, and comment upon existing works of art. If Andy Warhol could re-work the images of Campbell's soup or Marilyn Monroe, then a fair use defence would argue that today's gangsta rappers should be free to sample their source of musical inspiration to produce new and original work.


2 Live Crew making a statement with their bodies.

The scope of the fair use defence was explored by the US Supreme Court in Campbell vs Acuff-Rose Music (1994), which concerned 2 Live Crew's infamous parody of the Roy Orbison classic 'Oh Pretty Woman'. Rather than dismissing 2 Live Crew's claim on the basis that they'd used Orbison's music for commercial gain, the court looked at the factors of acceptable fair use, ruling that parody constituted a fair use, despite the fact 2 Live Crew had benefited financially.

Guidelines that the court considered in evaluating fairness of the use included the purpose of the use and its commercial potential, the nature of the copyrighted work, the size of the sample taken in relation to the copyrighted work as a whole, and the effect of the sampling upon the market value of the original work. In general, cases of sampling will be more deserving of fair use protection where they represent true creative effort on the part of the producer, and don't threaten the market of the original record. In this case, 2 Live Crew's buying public were considered to be of a sufficiently distinct demographic to Roy Orbison's older fanbase.

in the mix, you still need permission to sample. This means that the producer or artist must first gain sample clearance from the record label for use of the original sound recording and featured performances. Usually, you will also need permission from the publisher(s) for the use of the underlying composition (ie. the words and music). If copyright wasn't originally assigned to a record label or publisher, then you'll need to track down the respective copyright owners — or their heirs, if deceased — and seek their permission instead. Where, for example, a song has a number of co-writers or publishers, this is no easy task.

Alternatively, you could employ a sound-alike company to recreate the sample you're after (see the 'Recreation Grounds' box). In this case you wouldn't be infringing an original sound recording copyright, and would only require one set of permissions from the publisher of the music and lyric.

It's ironic that in an era of rampant piracy and the downloading of 'free' music, those making records still need to clear and pay for samples — whereas the end consumer can enjoy an entire album free of charge. Still, as a business-to-business activity, sampling is a lucrative business for those companies sitting on valuable copyrights, and in corporate-speak, can often give rise to valuable 'synergies'.

Witness the recent release of Sean Kingston's 'Beautiful Girls'. Not only did Sony/ATV-signed writer Jonathan Rotem help write and produce the track, but the song sampled the Sony/ATV-owned Ben E King classic 'Stand By Me'. The fusion of old recordings and modern technology helps publishers to safeguard future revenue streams by creating such chart-friendly hybrid copyrights. Rapper Kanye West has also topped the charts with his latest album offering *Graduation*, but much of the album's royalties will actually find their way into the pockets of '70s acts Steely Dan and Elton John, thanks to West's bountiful sampling of these artists.

Should you fail to clear the original sample before releasing your own record, you may be faced with a number of unwelcome legal consequences. You could be sued for damages for copyright

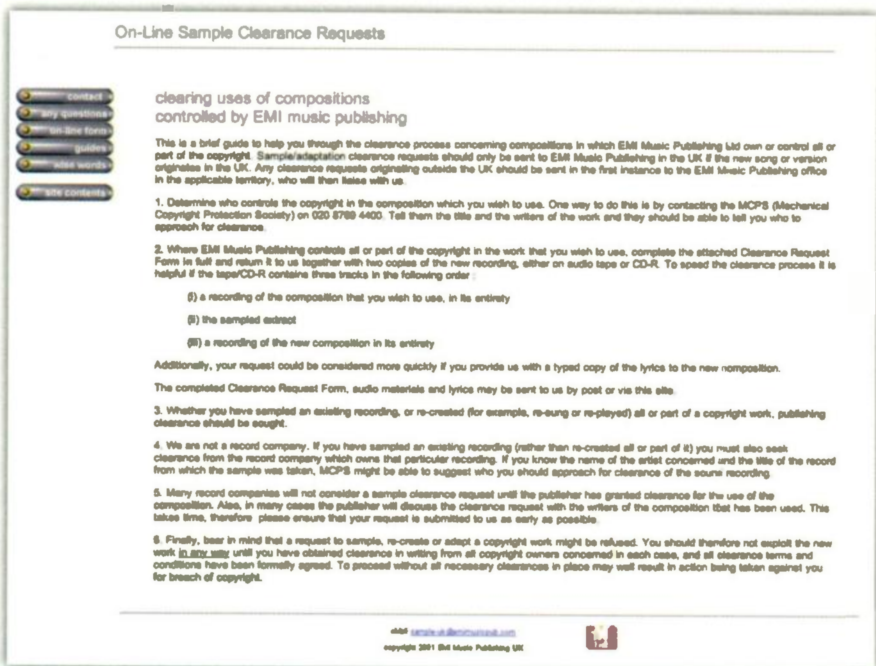

Kanye West, like many hip-hop stars, has used sampling extensively in his work.

infringement and face an injunction stopping you from continued sale of any infringing copies, as well as having to recall and destroy any CDs or DVDs incorporating uncleared samples. Your record label could even lumber you with the costs of this remedial work.

Clear Conscience

If you produce or remix records for other artists, it's usually your responsibility to clear any samples introduced during the recording process. The artist or their label will probably make you contractually responsible for doing so as a pre-condition of accepting delivery of the final record. Where the record label asks or insists that you include a particular sample, you should request that they pay for the related clearance costs. In other cases, sampling will add to the overall recording cost, and may be deducted from your earnings unless agreed otherwise.

If you already have a record deal, then you'll need to address the issue of whether any sample clearance fee, royalty payment or advances paid to third-party copyright holders should be recoupable from your own royalty earnings, or whether the record company should split all or some of the expense. Some labels take the view that sampling costs are part of recoupable recording costs, whereas others take a softer line. However, it's never advisable to ignore the issue and release a record carrying uncleared samples. Further down the line, the record company may invoke the artist's warranty clause in the recording contract and set about recovering sums



On-Line Sample Clearance Requests

contact
any questions
on-line form
guides
site contents

clearing uses of compositions
controlled by EMI music publishing

This is a brief guide to help you through the clearance process concerning compositions in which EMI Music Publishing Ltd own or control all or part of the copyright. Sample/adaptation clearance requests should only be sent to EMI Music Publishing in the UK if the new song or version originates in the UK. Any clearance requests originating outside the UK should be sent in the first instance to the EMI Music Publishing office in the applicable territory, who will then liaise with us.

1. Determine who controls the copyright in the composition which you wish to use. One way to do this is by contacting the MCPS (Mechanical Copyright Protection Society) on 020 8769 4400. Tell them the title and the writers of the work and they should be able to tell you who to approach for clearance.

2. Where EMI Music Publishing controls all or part of the copyright in the work that you wish to use, complete the attached Clearance Request Form in full and return it to us together with two copies of the new recording, either on audio tape or CD-R. To speed the clearance process it is helpful if the tape/CD-R contains three tracks in the following order :

(i) a recording of the composition that you wish to use, in its entirety

(ii) the sampled extract

(iii) a recording of the new composition in its entirety

Additionally, your request could be considered more quickly if you provide us with a typed copy of the lyrics to the new composition.

The completed Clearance Request Form, audio materials and lyrics may be sent to us by post or via this site.

3. Whether you have sampled an existing recording, or re-created (for example, re-sung or re-played) all or part of a copyright work, publishing clearance should be sought.

4. We are not a record company. If you have sampled an existing recording (rather than re-created all or part of it) you must also seek clearance from the record company which owns that particular recording. If you know the name of the artist concerned and the title of the record from which the sample was taken, MCPS might be able to suggest who you should approach for clearance of the sound recording.

5. Many record companies will not consider a sample clearance request until the publisher has granted clearance for the use of the composition. Also, in many cases the publisher will discuss the clearance request with the writers of the composition that has been used. This takes time, therefore please ensure that your request is submitted to us as early as possible.

6. Finally, bear in mind that a request to sample, re-create or adapt a copyright work might be refused. You should therefore not exploit the new work in any way until you have obtained clearance in writing from all copyright owners concerned in each case, and all clearance terms and conditions have been formally agreed. To proceed without all necessary clearances in place may well result in action being taken against you for breach of copyright.

copyright 2001 EMI Music Publishing UK

Major music publishers such as EMI are used to dealing with requests for sample clearance and have procedures in place to streamline the process.

direct from the artist, should the label be sued for copyright infringement.

During recording sessions, producers and artists should keep detailed notes of samples used, along with their source, and their timings on the record. This can be used for notification purposes on delivery of your final mix.

It's also sensible for producers and writers to hire out their services through a limited company, then through an employment agreement with this new company, assign copyright in the songs that they produce. Should they be sued for millions in a copyright infringement claim, they'll then be protected from personal bankruptcy!

In more general terms, in order to clear samples you can either use a sample-clearing company to assist you, or do the job of gaining permissions yourself. Sample-clearance companies such as Sample Clearance Services Ltd (www.sampleclearance.com) can often negotiate better rates than individual producers or DJs. Such companies can sometimes assist ▶

SAMPLE CLEARANCE

in providing legal advice and expertise in dealing with overseas labels and publishers, and may also be able to clear all future uses of the sample. Where, for example, your club tune crosses over, and you find you have a hit on your hands, you would then be free to license the record for film, TV, Internet or advertising usage, without seeking further licences.

Sample-clearance companies usually charge a flat rate: for example, Sample Clearance Services' web site states that their "standard fees for sample clearance are £275-£300 per clearance". Bear in mind that one sample may require two clearances: one for the sound recording and another for the publishing.

Two Steps Forward

Whether you use a clearance company or the DIY route, these steps should be taken:

- The publisher of the original work must be contacted. You'll need to find out who the original writers of the work are, and which publisher(s), if any, represent their interest or share of copyright in the song. In the UK, the MCPS/PRS can help you with this. They operate a vast database of registered works, and also have a sample-clearance team to assist you. Once you know the publisher and authors, you provide them with a copy of your new record, a copy of the original sampled record, and an isolated copy of the sample in question. Providing extra details, including the release label and size of the release, will help them evaluate your proposed use and speed up the process. The publisher is then in a position to consider price, contact the original composer for permissions concerning moral rights, and start negotiations over copyright ownership and royalty splits on the new record.
- The record company must be asked for permission to use the original sound recording. Master rights have their own price tag, and sometimes artists or labels will simply refuse to give their permission to use a sample — and they needn't give reasons. If no permission is given, or the price tag is too high, you needn't abandon your project altogether: as long as you can license the publishing rights, sample recreation companies should be able to construct an authentic-sounding reproduction of the recording. See the 'Recreation Grounds' box for more details.

How Much?!

In the normal course of events, when permission to sample is given there will

Recreation Grounds

If the owners of a sound recording flat out refuse to license your sample, or insist on a ridiculously high fee, you could employ the services of a sample-recreation company to work around the problem. Companies like Rinse Productions (www.rinseproductions.co.uk, or as interviewed in *SOS* September 2003 at www.soundonsound.com/sos/sep03/articles/rinse.htm) and Replay Heaven (www.replayheaven.com) offer to re-record the chosen sample, and can do so to such a high standard that the original version and the new one are practically indistinguishable. These musical skills have helped secure a string of dance hits for labels like Ministry of Sound, who recently benefitted from Replay Heaven's recreation of sections of Steve Winwood's 'Valerie' on Eric Prydz's million-selling dance tune 'Call On Me'.

High-quality re-recordings have the all the hallmarks of the original, but are quicker and easier to clear as there's only the publisher to consider, and no prospect of stalemate over competing interests with the record label. One such example was the recreation of the Human League's 'Don't You Want Me', featured in a Fiat Puto ad depicting a lovers' tiff played out in a Midlands garage forecourt.

Talking of re-recordings, there's been much rumour of late in the press that Wu-Tang Clan have achieved the impossible, and obtained rights to sample the Beatles' 'While My Guitar Gently Weeps', for inclusion on the WTC release 'The Heart Gently Weeps'. This story has garnered welcomed publicity for the group, but the reality is somewhat different. WTC's new track in fact includes re-created or interpolated elements of the Beatles' original. Wu-Tang weren't able to secure rights to the original master recording from EMI records or Apple Corps, and in fact their agreement sees them giving up 100 percent of all songwriting royalties, simply to re-record the Beatles composition. All publishing royalties will go to the estates of George Harrison and John Lennon, Northern Song owners Sony/ATV, as well as to Paul McCartney and the publishers of his share.

Sample-recreation specialists can help in situations where the owner of a recording can't be found, or refuses to license it.

be a fee for the privilege. The value of a sample, as well as the method of payment, will be determined by a range of factors, including:

- The notoriety of the original record and prominence of the sampled work in the new record. Puff Daddy's ode to Biggie Smalls, 'I'll Be Missing You', sampled the worldwide Police smash 'Every Breath You Take', thus sacrificing £500,000 in publishing royalties to its author, Sting.
- The likelihood of your success with your record. The territory, format of distribution, status of the artist and marketing spend all affect how your new version will be perceived, and therefore how much you'll be charged for the sample.
- Contrary to popular myth, samples aren't billed on a per-second basis like some phone calls — nor are they free when under three seconds long. The overall impact of the sample, together with all relevant commercial factors, means that each sample is evaluated on a case-by-case basis.

For the dance producer looking to issue a limited self-release, it's best to obtain a buy-out of all rights in the sample for a one-off flat fee. This would allow the producer to release the record and not incur further expense were the track to be picked up by a major label or licensed on compilations worldwide.

A major artist will be able to charge top dollar for the right to sample their work. They'll probably expect an advance payment running into thousands of pounds, as well as future royalties of approximately 1-5 percent on every record sold. These additional costs should be factored into your budget for the release.

Similarly, a stubborn or opportunistic publisher may demand 50-100 percent of the publishing income for the privilege of using their words or music. Rock band the Verve learned this lesson the hard way, when following the release of *Urban Hymns* in 1997 they were obliged to give up 100 percent of the royalties on album opener 'Bittersweet Symphony' to Abkco Records. The Verve had sampled *The Last Time*, a Rolling Stones / Andrew Oldham Orchestra record from 1965. In the court settlement, entire copyright ownership of the the Verve's song went to Abkco, with full songwriting credit going to Mick Jagger and Keith Richards.

Most publishers are more reasonable

when approached with sample clearance requests. But bear in mind that it can be a time-consuming process, especially if rights holders are based overseas, or where the track sampled has itself sampled another work. It won't be sufficient to gain permission for the second-generation sampled work — you'll also need to clear all original samples. And, of course, where you sample too extensively, it could end up eroding all profits in your track anyway.

Should you fail to clear a sample, or not even bother trying, you could still release your record and hope it goes unnoticed — although you'd be in breach of copyright. But what happens when an underground release becomes an unexpected hit? At this point the original copyright holder will crawl out of the woodwork and demand that you recall the record from the shops and pay damages, and if you're very lucky you'll be able to re-release the record. only with the offending sample removed. Quite apart from the legal nightmare of injunctions, lost profit, and damages claims, the delay alone could cost you sales and your chart position — a fate suffered by Rui Da Silva, who was sued by BMG records and obliged to remove an uncleared sample of Spandau Ballet's 'Chant No. 1', from his dance hit 'Touch Me' featuring Cassandra.

Don't let these cautionary tales put you off sampling, though. Most labels, publishers and artists are only too happy to give their permission to artists looking to re-work their music — for a fee. Moreover, not all unauthorised sampling ends in tears. 'Tom's Diner', an a cappella song written by Suzanne Vega, was known only to fans who bought her 1987 album *Solitude Standing*. Then, in 1990, DNA sampled Vega's voice over a sparse beat-laden track. The results were so popular that Vega and her label decided to issue it as an official remix, achieving worldwide acclaim and a top five single. SOS

Blackstar HT-series

Tube Guitar Pedals

Blackstar Amplification are a UK company set up by a number of ex-Marshall designers. Their first products, reviewed here, are pedals — but the company name provides clues as to their future plans.

I looked at five of their HT range of pedals. Unlike many other stomp boxes that incorporate tubes, these all use the full 300V HT supply, which ensures that the tubes run correctly: they're not just there to provide an orange marketing glow! The main signal path is all-tube, though additional buffering and switching components are used to ensure noise-free switching, and to provide a high-quality buffered bypass mode that enables the pedal to drive long cables.

All the pedals are suitable for use on stage and in the studio, and power comes from an included 16V AC adaptor. Internal circuitry converts this to the necessary voltages to run the tubes and other components.

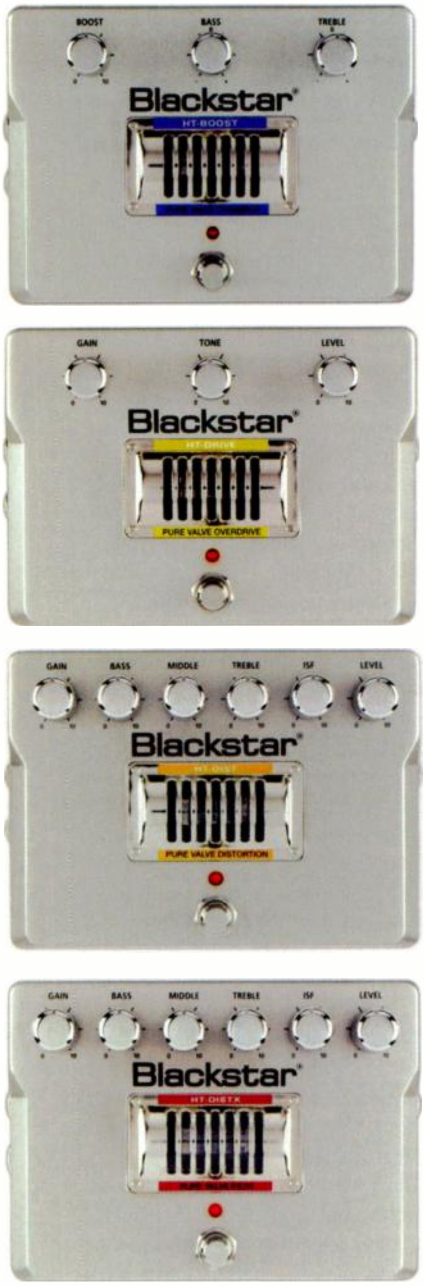
Despite the high spec, the most expensive of these pedals is a modest £129 in the UK, the cheapest being £89 — prices that are made possible by designing in the UK but manufacturing in Korea.

Overview

The simplest of the pedals is the HT Boost, which is designed to act like two extra cascaded preamp tube stages (from one dual-triode tube), boosting your signal and keeping the tube tone intact. It features Treble and Bass tone controls, a Boost control and a bypass footswitch. Like other pedals in the range, it is ruggedly built in a chunky but stylish cast-alloy case with knurled metal knobs and an input jack on the right-hand side. This particular model has two outputs, designated High and Low, to suit both tube and solid-state amplifiers. An LED shows when the pedal is active, and a grille in the centre of the case that allows warm air to escape means you're able to see the tube.

The HT Drive, which offers something a bit more extreme, also has dual, cascaded triode stages, and a tonal range from gentle boost to tube-saturation distortion. Blackstone's A-Class tone control cuts out high end fizziness without dulling the part of the sound you want to keep, and for recording there's also a speaker-emulated output jack, as well as the normal 'to amp' output. The controls will be familiar to anyone who has used a distortion pedal before, namely Gain (drive), Level and Tone. This pedal is actually very versatile, as it ranges from almost clean at one end of the scale to full-on rock overdrive at the other, while the tone control is nicely subtle and very usable (unlike the controls found on many other pedals).

A little further up the scale is the HT Dist, which is an overdrive/distortion pedal with a novel tone-control network that sweeps between a US and UK rock guitar sound. Blackstar call this their Infinite Shape Feature, or ISF. Again, the overdrive is all tube (both stages of an ECC83 dual triode)



The Black Star HT range comprises five pedals. Above from top are the HT Boost, HT Drive, HT Dist and Dist X. The HT Dual is pictured opposite.

and there's a speaker output for recording or for feeding into a PA system. The idea is that the pedal responds like a tube amp, so you can control the amount of distortion by backing off your guitar's volume control. This model breaks away from the 'three knobs and a switch' tradition by also including a full bass/middle/treble passive EQ section along with the ISF tone knob. Gain and Level set the amount of distortion and adjust the output level respectively. You could think of it as similar in concept to an HT Drive but with the extra tone controls.

If 'more' isn't enough, the 'filth monster' of the range is the HT DistX, which can go from mild overdrive to contemporary US-shred, and this again has an emulated speaker output. Its enhanced passive tone controls incorporate the same Infinite Shape Feature and its layout is essentially identical to that of the HT Dist, to which it is also tonally similar, except that there's more drive gain available.

The HT Dual, as its name implies, is a dual-channel pedal designed to put more tonal options under foot. Channel one handles either Clean or Crunch modes, and so can be used for clean boost or for creating moderate overdrive. Channel two offers higher gain and goes beyond crunch up to modern high-gain lead. Its switching operation essentially gives you a choice of three tones (including your amp's own clean sound). Again, there's that ISF tone control to sweep you between US and European rock sounds, and you still get the speaker emulation. There's both a three-band EQ and dual-concentric drive control and output level knobs (for the two channels). Gain 1 also has an associated button switch to set the sound to clean or crunch.

A slightly unusual switching system is employed, with separate footswitches for each channel rather than one switch for bypass and one for channel changing, but some clever logic-switching circuitry makes this very intuitive. If one channel is active and you hit the other, the channel switches to the one you just hit. If, on the other hand, a channel LED is on and you hit the same switch, the light goes out and the unit goes into bypass mode (and vice versa), so you can always get the choice you need with a single switch click.

The Sound

These pedals all sound very musical and have a great dynamic response, just like a real tube amp, which means you can turn down your guitar's volume control to clean up the sound, even when you're using quite heavy distortion.

To work out what they sound like you really needn't look much further than the HT Dual, which is capable of duplicating pretty much all the effects available from any of the other pedals. Channel one is much like the HT Boost in clean mode, and not unlike the more polite end of the HT Drive when you switch to Crunch mode — making it useful for blues or rock/indie rhythm. The main difference is that the HT Dual also has the full four-knob tone-control setup (the tone controls affect both channels, by the way), so there's more flexibility for tailoring your sounds. Channel two of this pedal even gets you into HT Dist and HT DistX territory. Let's have a look at the different sounds available from the rest of this range.

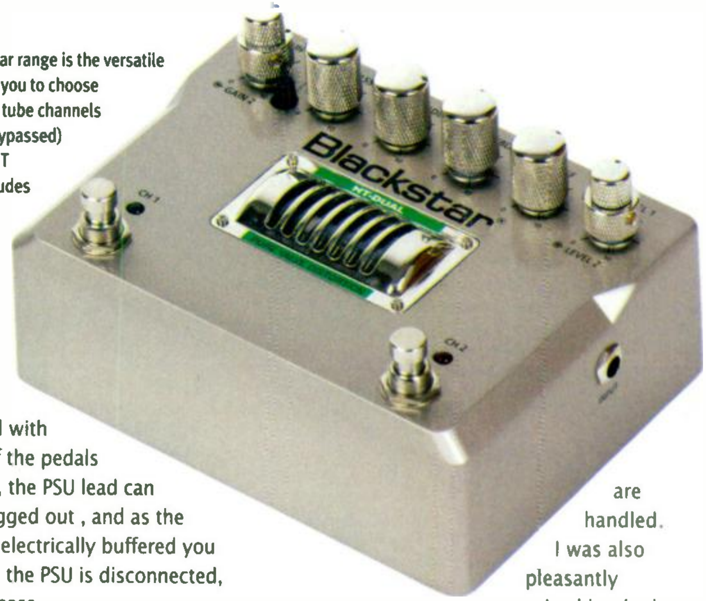
The 'Boost' in HT Boost is exactly what its name implies: the signal gets louder, allowing you to overdrive the input stage of your own amp, so the distortion characteristic you hear depends very much on how your amp behaves. There's a useful degree of tonal control on the HT Boost pedal (courtesy of the Treble and Bass knobs) but the HT Dual is rather more flexible.

The more polite of the overdrive pedals, HT Dist, gets you into blues and classic rock overdrive in a very controllable and convincing way, and working your way up the distortion ladder eventually gets you to the HT DistX, which isn't as savage as its name might lead you to believe. At lower settings it manages a well-behaved crunch, while at halfway up you find yourself in classic rock territory, and with full gain you're ready to shred — but the thing still doesn't run out of control. Those pedals that have three-band EQ behave much like the passive three-band EQ sections you'll find on many tube amps, but the ISF knob is interesting, as it interacts with the more conventional tone controls to change the character of the sound in a way that seems to particularly affect the mid-range. With the ISF control fully clockwise you get that very focused, slightly edgy sound characteristic of US rock, while at the other extreme the sound has a 'spongier' low end and a warmer high end, with plenty of mid-range harmonic squeal. By experimenting with the EQ and the ISF knob, you can coax a lot of classic tones from the box, including some very Tube-Screamer-esque sounds. I tried both single-coil and humbucking guitars and the character of each still comes across nicely, even when you pile on the gain.

The speaker-emulated output makes a fair stab at replicating the coloration of a speaker, and it takes the unwanted fizz out of the high end without making the sound appear dull or lacking in bite. Clearly, it isn't as sophisticated as some of the digital modelling solutions out there, but if you're recording with it there's always the opportunity to add more processing later if you need it. It certainly stands up well in comparison with other stomp-sized speaker emulators and is a very welcome addition.

Bright Star Or Black Mark?

The whole range sounds really good and I'm looking forward to trying one or two of the pedals on stage — but with one caveat: external PSUs are no problem when used with a pedalboard, but if the pedals are used separately, the PSU lead can get knocked or dragged out , and as the bypassed output is electrically buffered you lose all signal when the PSU is disconnected, even if you're in bypass.



The top of this Blackstar range is the versatile HT Dual, which allows you to choose between two separate tube channels as well as the clean (bypassed) signal. As with other HT pedals, it usefully includes a speaker-emulated output.

"The HT Dual is capable of duplicating pretty much all the effects available from the other pedals."

That issue aside, the sound of these pedals is very natural, with excellent dynamic response, and tone controls that give you plenty of usable range without straying into dull or useless fizz territory. Though it may seem subtle at first, the ISF control is very useful for tuning the character of the sound, and the more you use the mid-range control, the more effect it seems to have. There's enough range to go from UK scoop to US modern metal, but I like the way those in-between blues tones are handled. I was also pleasantly surprised by the low noise-floor, even at relatively high-gain settings, and the emulated speaker output is a useful bonus for recording. Best of all, though, is the price —which is lower than that of many solid-state pedals — so be sure to check them out! *Paul White*

SUMMARY

This is a great, versatile series of pedals, ranging from the clean HT Boost up to the hugely flexible HT Dual. They have excellent dynamic control — so you can almost forget the bypass and just use your guitar's volume control to bring the distortion in and out.

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NEWS

Waves have expanded their hardware range for guitarists with the iGTR, a palm-sized guitar effects processor. It's designed to be used anywhere and is powered by batteries, making it highly portable. Connectivity includes a quarter-inch jack input for connecting a guitar, and two mini-jack outputs for headphones, although Waves say that the second output can be used to send a feed to a recorder (or DAW), an amplifier, or a second pair of headphones. There's also an Aux input, so you can plug in an MP3 player and jam along to your favourite tracks.

Knobs and switches on the front panel allow you to select and adjust reverb, delay, chorus, wah-wah, tremolo and phaser, as well as the tone of the amplifier-modelling circuit. The iGTR is available now, priced at £69 including VAT.

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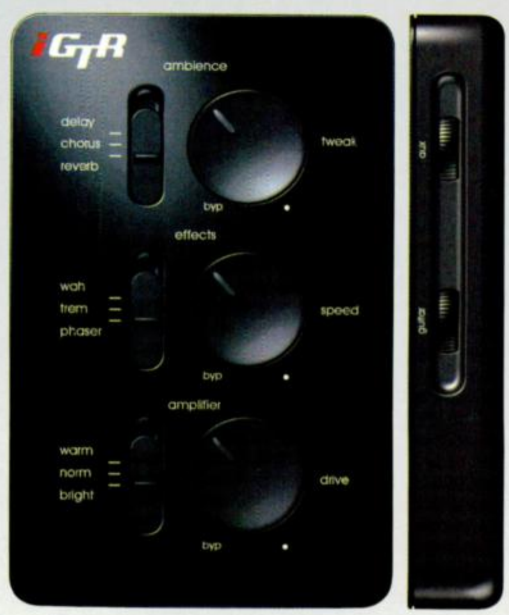


Native Instruments Kore 2

Hardware Controller & Software Host For Instruments & Effects

The latest incarnation of NI's Kore system introduces an updated controller, redesigned and streamlined software and a generous library of built-in sounds. So is Kore all you need to get the most out of your software instruments and effects?

Simon Price

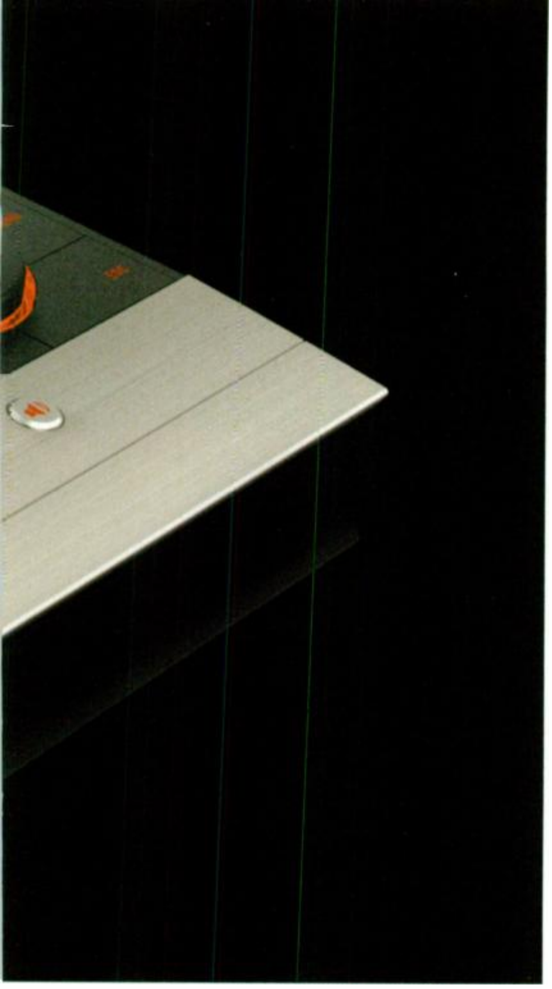
Kore 2 is a combined software and hardware system for hosting and controlling plug-ins, and, as its name would suggest, is successor to Kore 1 (reviewed in the July 2006 issue of *Sound On Sound)*. The software runs stand-alone, or as a VST, AU or RTAS plug-in within your DAW of choice, and promises a range of benefits to studio producers, composers and live performers.

Kore provides a workstation-like front-end for your sounds and effects, allowing you to pull patches from your plug-in collection into a centralised library of 'Koresounds'. You also gain hands-on control of key parameters from a single plug-in interface and hardware device. This lets you abstract yourself from the underlying sound sources, and deal with your sounds as a unified palette.

The host shell provides a structure for layering and chaining multiple plug-ins into larger blocks that can then be used as single plug-in inserts. This brings exciting possibilities for sound design and for saving complex effects chains, or sound 'Multis', in Kore-speak. Kore can also prove useful for musicians wishing to use virtual instruments on stage or in sessions.

New In Version 2...

Kore 2 is a major overhaul of the original Kore, with an updated hardware controller, This, thankfully, no longer needs to be attached to allow you to use the Kore software, which itself has been completely



redesigned. The biggest change is the addition of built-in sounds; you can now make music just with Kore and a computer. Differences between the stand-alone application and the plug-in are gone, meaning you can use the live performance functionality within a host DAW.

New innovations include Sound Variations, which allows you to store up to eight snapshots of the current patch and morph through them. MIDI effects, including a step sequencer, offer enhanced performance possibilities. And my personal favourite: routing of MIDI and audio between objects in the Kore rack turns the system into a versatile synth studio.

New users get the new controller, while upgraders can choose to keep their original hardware for a significant saving (see the 'Upgrades & Pricing' box). The Kore 2 controller looks very similar to its forerunner. The audio controls are gone (the controller no longer doubles as an audio interface), and the buttons around the scroll wheel have a new layout. The glossy plastic areas of the front panel have been replaced with a less attractive, but less fingerprint-prone, matte finish.

In nearly all respects, the hardware problems I described in the Kore 1 review have been addressed. The menu buttons now have a light, clean action with a reassuring click — a massive improvement over the spongy and unreliable earlier ones. The eight controller buttons also click now, although they require more pressure to engage.

The back panel, although devoid of audio connections, still sports useful MIDI In and Out ports for general interfacing duties and, of course, a USB 2.0 port for connecting to the host computer. Note, however, that the Kore 2 controller itself is not a MIDI control surface, and can still only be used to operate the Kore software and hosted plug-ins. Generous provision is made for foot controllers, with two inputs each for pedals and footswitches.

Koresound Packs

If Kore 2 is analogous to a workstation synth, its expansion cards are Koresound Packs, 49 Euro downloads adding new sounds to the Kore library.

Best Of Reaktor is the cream of the crop, featuring great instruments from the Reaktor and Electronic Instruments libraries. I said in the Kore 1 review that Kore could make Reaktor into a much more accessible instrument, and this shows how. Decent use of control assignment is made in most cases, although the Photone-based patches all seem to use the same mapping, so some patches have redundant knobs.

Best Of Massive features 200 patches from Massive's library, which means that it's dominated by top-notch, distinctive synth leads, basses and pads. You might think the control mapping is a bit stingy, featuring just one page per patch. By contrast, the same patches on the full version of Massive have two patch-specific pages, and 16 general parameter pages. However, the controls sensibly follow the Macros from the original Massive patches, so are all directly relevant to the sounds.

Synthetic Drums Reloaded adds 10 kits of electronic and processed drum sounds. The kits are large, packed in across the keys, with different 'sub-kits' in each octave. Although you'll find some solid sounds here, this was the most uninspiring pack, for me. These sounds are feeling a little tired now, and the control assignments are mostly effects and seem to have been an afterthought. I'd stick with the factory kits, as the Drum Machine patch takes care of the TR808, TR909 and CR78, and the Electron and G-Shot kits are at least as good as anything in this extra pack.

The '57 Drawbar Organ deserves special mention for its seriously impressive programming. This pack, a beautifully sampled 1957 Hammond C3, is the only one which features material that is not available in other products. This probably explains why there are a generous number of controls, allowing you much more access to the instrument inside.

Installation Issues

I'm generally in favour of skipping any discussion of installation in a product review — except when it impinges on the user experience. I love NI's synths, and I couldn't be without them, but they do demand a certain level of regular admin. This is especially true when Kore is present, as it needs to co-ordinate with the various plug-ins and sounds on your system.

A fresh install of Kore 2, with no previous NI products installed, shouldn't cause you too many problems. A single DVD installs everything you need, including the Koresound library and the 5GB of sample content used by some of the built-in sounds. To register and unlock the software (with a serial number) and get the latest update, you need to run the Service Center utility which connects to NI on-line. Despite being over 200MB, you should download the 2.0.1 update which adds functionality such as Performance Presets that didn't make it to the first release.

In my case, installing onto a system with Kore 1 and a sprinkling of NI instruments was far from straightforward. Inevitably, numerous plug-in and library updates were needed to make sure everything was happy. Even so, half of my sounds were not seen — a common issue, apparently, and eventually fixed after a trip to the user forum. I also had problems with Kore losing its authorisation and the path to its samples, which required a re-install.

Kore 2 As A Workstation Synth

The most straightforward use of Kore 2 is as a 'workstation' synth, browsing for sounds and tweaking them with the pre-defined controls. This doesn't require ▶

SOUND ON SOUND

Native Instruments Kore 2 **£349**

pros

- Much improved interface and more coherent concept.
- Built-in sound engines.
- Broad-ranging synth and sample-based library.
- Improved hardware controller.
- Sound Variations and morphing.
- Step sequencer and internal MIDI routing.
- Creating Assignments easier.

cons

- Set aside a day to get through the install/update/authorisation process.
- Expensive hardware upgrade for Kore 1 users.
- No built-in audio interface.
- Performance Preset crossfading and sequencing has gone.
- No bank/program select controls for hosted plug-ins.
- Occasional instability.

summary

A great improvement. Despite some niggles, this is much closer to the product we hoped Kore would be. It makes using virtual instruments more fun, and has numerous creative and practical benefits for desktop musicians and producers.

NATIVE INSTRUMENTS KORE 2

The default display, with new customisable Browser. Some attributes lists feature secondary columns to further narrow your search.

much understanding or exploration of the software's internal environment, so the user interface has a strip of buttons for hiding everything that you don't need.

The attractive, clean design of Kore 2 is a class above the first version. Coming from Kore 1, I was initially disorientated by the layout and conceptual changes, and by the fact that everything is icon-based, with almost no text. Luckily, the Info Pane at the bottom of Kore 2's window displays the name and function of every object you mouse over, and provides handy usage tips.

By default, just the Global Controller and Browser panels are shown. The Global Controller mirrors the eight knobs and buttons on the hardware controller, labelled with their current functions. This is the only place where controls are displayed, unlike in Kore 1, where every device in the rack sported its own knobs and buttons. While the old way was potentially useful when using a mouse, it caused confusion about what the hardware was focussed on.

Sharing the Global Controller area is the Sound Variation Matrix. Sound Variations are simply snapshots of all mapped parameters and mixer settings in the patch. Library patches have preset Variations, but you can store your own. As well as using the mouse, you can switch between Variations with the controller's buttons, or morph between them with the knobs. Each knob represents one of the eight Variations, and turning a knob morphs the sound towards its corresponding Variation. This is an ingenious and effective design.

The sound browser has been extensively reworked, and is much the better for it. A key concept in Kore is the classification and searching of sounds by their type and character, rather than by plug-ins (an idea that has been quickly adopted by other developers). As well as user-input word searches, you can select from lists of sound types, modes, timbres

The View From The Stage

Many keyboard players are replacing synth and sampler racks with virtual instruments and a laptop. Kore 2 is well suited to hosting plug-ins for live use, with all the necessary tools for setting up keyboard ranges, MIDI channels and routing, transposition and hardware control. The MIDI Player plug-in can be used for playback, similar to how the Akai S6000 is sometimes used. Kore 2 can now be used inside Ableton Live, without losing the Performance layer, as happened in Kore 1.

Kore no longer has a Live View, instead having a Performance Presets view. Performance Presets are snapshots of the current configuration, including which mixer channels are enabled and disabled. Typically, you'd set up the instruments needed for a gig or session, and create Presets to enable the right channels for each song or section. Kore 1 had the same concept, although a preset stored everything (including audio routing). You can no longer store routing in a preset, which may or may not be a problem for you. On a more positive note, any controller page can now be 'isolated' from the Performance Preset functionality. This would be useful if, say, you had a piano patch you wished to play manually and keep unaffected by PP changes.

Kore 1 had a Performance Presets list you could use to create an automated sequence and set transition rules such as bar/beat quantisation and crossfades. This has gone, replaced by a display of 16 banks of eight Presets. Quantisation of changes is global, and you can't crossfade between Presets. NI say that where crossfading is necessary you should set up Sound Variation snapshots instead, which are similar to Presets anyway. However, this means leaving more channels enabled, so CPU load may be higher.

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Bundle with the Team. The HDSPe PCI Card and the HDSPe ExpressCard are also available with the Multiface II as bundled recording solutions. These new bundles are now available in an attractive priced package that represents a considerable saving on the cost of the individual components.

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Alternatives

Ableton Live's Racks can be used to many of the same ends as Kore 2, as can Reason's Combinator, although both offer less Macro controls and the latter within a closed environment. Logic's library can store instruments and effects in chains. Apple's Main Stage offers a similar solution for live/session work, and has access to Logic's instruments. However, it can't run in a host (or on a PC), and of course there's no integrated hardware. A number of smaller developers have plug-ins for grouping other plug-ins; examples are ART Teknika's Console, and Audiofile Engineering's RAX. Another established software solution for live work is Brainspawn's Forté. For those who prefer not to take a laptop on stage, Muse's Receptor hosts plug-ins in a hardware rack unit.

and so on, producing an ever-diminishing list of sounds tagged with the attributes you require.

The new Browser can be customised by choosing which attribute lists are displayed, or by dragging and dropping existing attributes into a new column to make your own list.

Onboard Sounds

The basic install includes a library of over 900 Koresounds, split roughly into Instrument and Effect patches. Kore 2 is advertised as coming with 'over 500 production-ready sounds'. There are actually 439 Koresound Instrument patches, but I won't quibble, as the Sound Variation snapshots in each patch put the number of actual preset sounds into the thousands.

The library is rich and varied, thanks to the diverse onboard 'sound engines'. Special versions of Massive, FM8, Reaktor, Kontakt, Guitar Rig and Absynth are installed, and used like regular plug-ins within the Kore 2 patches. Unlike the full versions of these instruments, the sound engines don't show up in the plug-in list for use in your own patches, and you can't access their software interfaces. Instead, they are controlled only by whichever knobs and buttons have been assigned in each patch. The extended general control mappings that you get with the full plug-ins are absent, so low-level sound programming isn't possible.

This is fine for finding and tweaking sounds, but if you prefer to build your own sounds from scratch, you'll be looking to buy the full versions of the plug-ins. You will then be able to open and fully edit the plug-ins used in the Koresound library. NI's instruments also come with extensive Koresound libraries, so your Kore database swells considerably as your instrument collection grows.

Rack 'Em Up

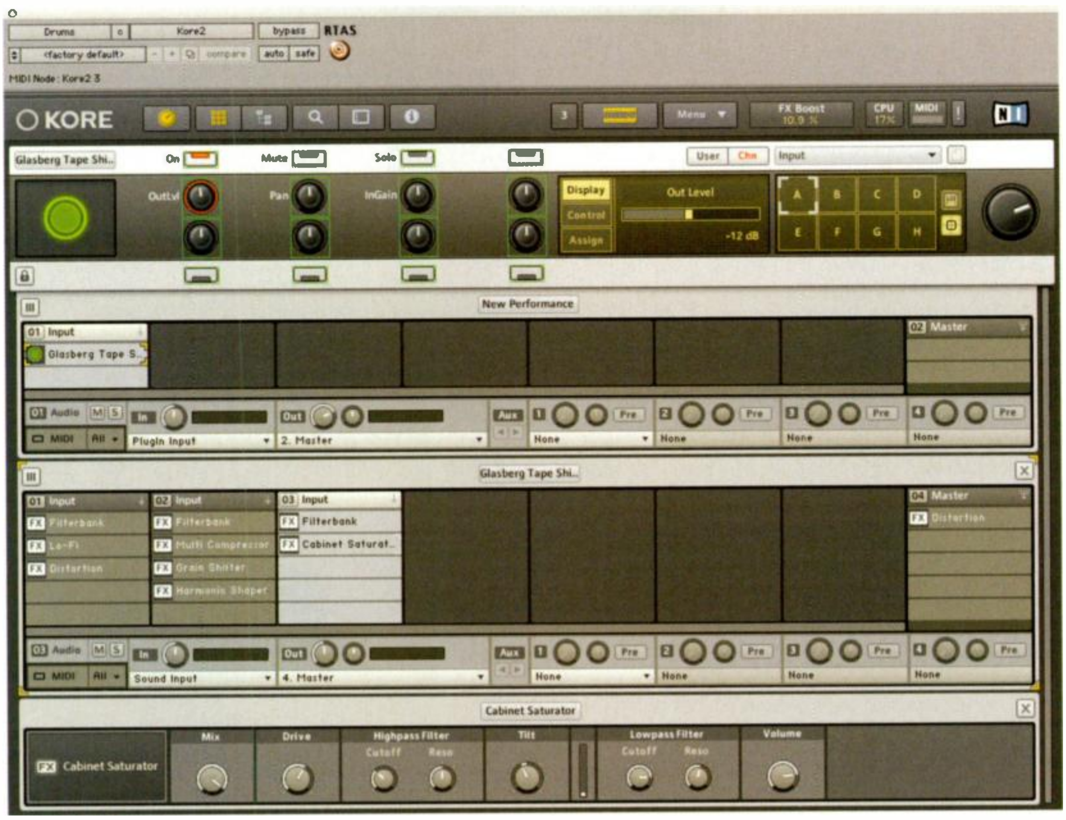
Kore 2 avoids much confusion by removing the difference between the stand-alone and plug-in versions of its software, and presenting all patches in a single page view. There is still a distinction made between a Performance (the top level of any instance of Kore) and the Koresounds and devices contained therein, but this is actually quite arbitrary.

A Performance is basically a mixer, which can contain Input, Source, Group and Master channels. Each channel has controls for routing both audio and MIDI, as well as aux sends, allowing for complex structures to be assembled. Channels have insert slots where you drop the sound sources and effects that make up your patch. The simplest patch would be a single instrument plug-in on a single source channel. Things get more interesting when you put a chain of devices on a channel; for example, an Arpeggiator (MIDI plug-in) followed by a synth (instrument plug-in) followed by a delay unit (internal or plug-in effect).

Adding another channel creates a layered sound, or a split using the Sound Manager view's Channel Mapping area — as in the screen below. By assigning different MIDI channels to the mixer channels, you can create a 'Multi', which could be played back by multiple DAW tracks.

The most powerful aspect of Kore's structure is that any slot in a channel can contain a Koresound instead of an individual plug-in. This Koresound may itself contain an internal structure, with MIDI effects, multiple plug-in instruments, and audio effects.

Selecting a Koresound in any channel of



A large effects patch, with three parallel chains. The lower mixer shows the internal structure of the Koresound patch.

The rear panel of the Kore 2 controller — now without audio connections but with an extra pedal socket.

a Performance now displays that sound's internal structure below the main mixer level. This layer may contain other Koresounds, so a complex sound may have several layers of 'nested' mixers, which will be displayed one above the other.

One of the most elegant functions in the new versions is the 'Save Performance as Sound' command, which packs the entire current plug-in configuration into a Koresound, ready to be slotted into a channel later on.

Hands On

When browsing and loading Koresounds in individual instances of Kore 2, you'll always get at least one top-level page of knobs and buttons that adjust significant parameters of the patch. However, once you grasp Kore's internal structure, you can navigate through it with the hardware, accessing different control pages for each channel and device. The controller displays the structure as a grid and you use the cursor and menu buttons to focus different slots and move between layers.

The touch-sensitive encoders are still one of the highlights of Kore, and now have a new trick. The brightness of the lights around the bottom of each knob now indicates the value of the currently mapped parameter.

The plug-in feels more integrated with the hardware, and it's much easier to tell where you are, partly because the plug-in window updates to show changes of focus from the hardware. You can also scroll through the Browser's results list from the controller, and the on-screen list follows, allowing you to preview attributes. Each instance of Kore in a project has a button that focuses the hardware to that window. You can also switch between different instances from the hardware, although unless you rename each instance they are all called 'New Performance'. Couldn't the name default to the first Koresound you load, unless you choose to change it?

Browsing from the hardware is limited to the results list in the software, whereas in Kore 1 you could navigate a hierarchical list. It takes four steps to go from a control page to changing a patch and back again; it works fine, but it's quicker to use the computer's ▶

NATIVE INSTRUMENTS KORE 2

The built in Step Sequencer plug-in can act as a stand-alone sequencer, or be triggered and transposed from MIDI notes.

cursor keys. On a somewhat related note, the Next/Prev buttons that used to be displayed on plug-in objects are now gone, meaning that you have to open the plug-in's interface to switch patches at that level. However, many VST plug-ins don't have patch-selection controls, assuming that the host will.

Third Party Plug-ins

All NI plug-ins now work with Kore; controller pages have been set up and Koresounds created for all the presets. In fact, Koresound is now the native preset format for most of the NI range.

But what about other plug-ins? Firstly, the factory instruments and effects that come with your DAW (Pro Tools, Logic, Live, etc) will not work because they can only be hosted by their own application. However, any VST- or AU-format plug-in should open in Kore 2. A growing list of third-party plug-ins have pre-defined controller pages, so will work with the Kore hardware without needing you to map parameters.

A notable advance is the ability to batch import sounds from any plug-in that uses the standard VST patch-bank scheme. If you first set up some controller mappings, these will be saved with all the imported patches too. This is a great feature, although it falls foul of Kore 2's lack of patch/bank controls for plug-ins, meaning that you may only be able to grab the default bank.

Pricing & Upgrading

If you're upgrading from Kore 1, moving to the new hardware version costs the full price, but you can opt for a software-only upgrade. Although the build quality of the new hardware is definitely superior, the original controller offers full functionality with the Kore 2 software. In fact, if you are using Kore 1's built-in audio interface, you may be better off staying put.

- **Kore 2 full upgrade: £349.**
- **Software upgrade from Kore 1: £69.**

Summary

Kore 1 showed much promise and was, indeed, useful for live work, but it was complicated and temperamental, and many users found they drifted back to using individual plug-ins in their DAW. Kore 2 is a marked improvement in most areas and I found it much more enjoyable to use. The integrated sound engines are NI's smartest move, as Kore 2 is now useful to anyone, not just those who already have plug-ins. In fact, I had a lot of fun making tracks while restricting myself just to the built-in sounds, drum kits and effects. Sometimes it's a relief to just get on with writing a tune instead of getting bogged down with programming.

A powerful reason to look at Kore is for use as an effects hub. As well as being able to build complex serial and parallel chains for unique effects, you can set up your day-to-day EQ and dynamics chain as a patch, with a simple page or two of hardware controls. With over 30 built-in effects, and many Reaktor and Guitar Rig effects in the library, Kore 2 is a cost-effective way to add a suite of processing options to your DAW.

Kore 2 is probably the most powerful software-based solution for performing with virtual instruments. Although some Kore 1 functionality is lost, the ability to work with Performance presets inside an application like Live is an important step. MIDI effects, and the step sequencer, along with MIDI routing between objects in a patch, let you experiment with Kore as an analogue-style synth studio. And, of course, the Sound Variations give you another way to store sound presets and morph between them.

Despite some false starts, NI now seem to have found their direction with the Kore project If NI synths are an essential part of your music making, as they are for me, Kore is of real benefit and opens new creative avenues. Kore 2, with a laptop and controller keyboard also represents a real and intriguing (if less straightforward) alternative to a hardware workstation. SOS

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Great String

...And In A Hurry!

Whether you're recording real players or using samples, our in-depth guide is your key to polished and authentic-sounding strings.

Mike Senior

There was a time when lush-sounding string overdubs were one of the hallmarks of a big-budget production, and while there's still nothing that quite matches the sound of a real string orchestra, these days you can get amazingly good results on a budget. Pulling off this feat involves a combination of arranging, programming, recording and mixing skills, but most software sequencers provide all the tools you need to have a stab at it, even if you're not confident with traditional musical notation. This article will pass on some practical advice to help you create good-sounding string parts for your track as quickly as possible, whether you're using samples or live string players — or both!

Quick Fixes For String Arrangements

Irrespective of how you plan to realise the sound of your arrangement, there are lots of quick fixes that can help transform rookie string arrangements into something more musically effective, regardless of your facility with notation.

• Aim For The Gaps
Strings can be tricky to pick out in a busy mix, so try to direct string parts into gaps in the arrangement, such as those between lead vocal phrases. If the strings are playing alongside other rhythmic parts, you can make the strings more audible by getting them to switch notes on beats when no-one else does, even if this is simply shifting to a different note-spacing of the same chord. It's also worth looking for frequency ranges where nothing's otherwise happening in the arrangement, and parking the strings there: more audibility, less mix-clutter. On a similar tack, the best arrangements don't have all the parts playing all the time, but rather use variations in the instrumentation to help support the structure of the song.

• Beware Of Footballs
Although there's a lot to be said for keeping things simple, good string arrangements are rarely just made up of block chords, which bore string players to death and are disparagingly referred to as 'football music' by some session players, on account of the rows of identical note values. To avoid this trap for the inner string parts (typically the second violins and violas — see 'Ensemble Line-ups' box, opposite), try giving them some kind of rhythmic figuration, even if this is just a case of them repeating notes, or swapping notes with each other in some kind of rhythm. If you can incorporate some kind of melodic fragment into this figuration at some point where the rest of the track thins out momentarily, there are serious bonus points to be had. You wouldn't have a musical drum part without fills, and it's the same with string parts. For the outer parts (typically first violins and cellos), the best thing is to try to conceive them as melodies in their own right, however basic. If they sound singable, that musical phrasing will add momentum to your track.

• Give Low Instruments Enough Space
You'll usually get more satisfying string sounds if you leave bigger gaps between the low parts than between the high parts. In practice, it's probably easiest to double the cellos at the octave with the double basses most of the time, unless you're after

Parts On A Budget

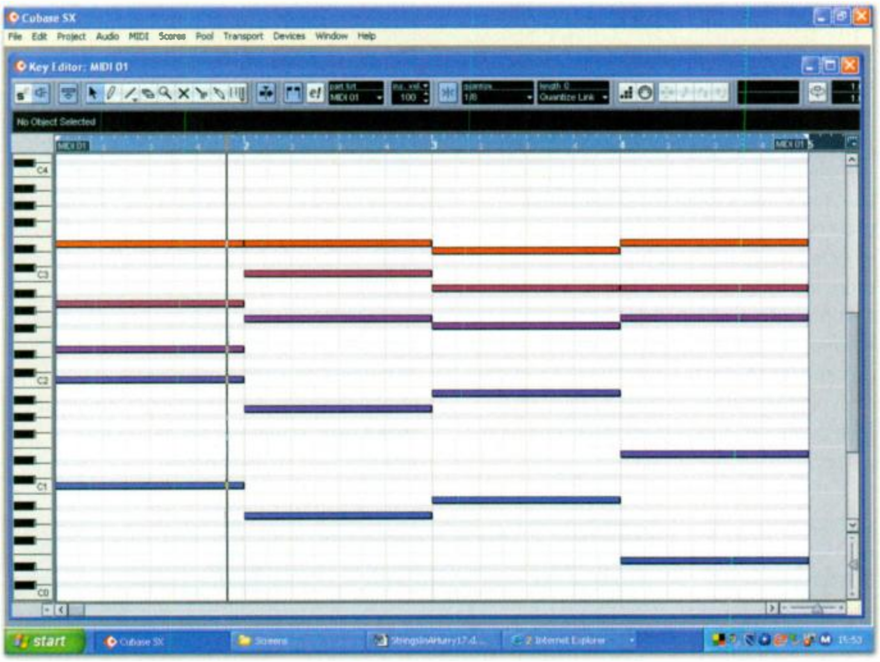
a special effect. It's also not bad practice to give the lower parts enough time for their longer, heavier strings to speak properly, leaving fast runs and figurations mostly to the upper strings.

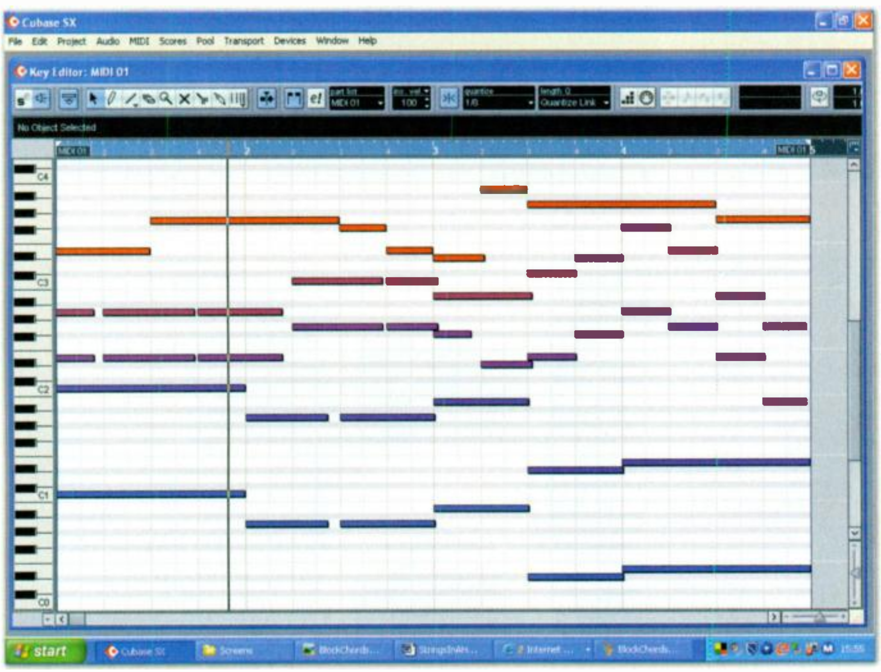
• Fatten Up Solo Lines In Busy Tracks

In busy commercial string arrangements, what sounds like a single string line is often actually harmonised. You can't pick out the harmonies particularly well because they're partially obscured by other instruments, but they nonetheless help give the top-line melody more body in the mix. The simplest way to harmonise a melody like this is to copy the whole melody line; paste it either three, four, eight, or nine semitones below (listening for the one that works best); and then adjust any notes that don't harmonise well to the nearest neighbouring notes that do. Doubling 12 semitones below isn't usually quite as effective on its own and any other interval you try is likely to sound pretty weird.

• String Players Aren't Fingers!

Keyboard players are particularly prone to the mistake of treating a string ensemble as if it's a keyboard patch. The individual players in an ensemble are much more independent than a keyboardist's fingers are, and good arrangements reflect this. Try to push beyond what can physically be played by a single keyboardist, particularly in terms of combining different playing techniques at the same time — how about *pizzicato* cellos and double basses, *col legno* viola trills, and a *con sordino* second-violin melody doubled by tremolo first violins an octave above (if this sounds daunting, check out the Jargon Buster box elsewhere in this article). I'd also suggest that all the instruments playing the same rhythm should be very much the exception, rather





While simple block chords can work well on occasion, there's a lot to be gained in most situations by giving the outer parts a bit of melodic contour, which helps add musical momentum. Some simple rhythmic figuration in the inner parts can also be very effective for increasing energy levels, and you can develop these figurations into little counter-melody fills at the ends of sections as well. All these techniques can be seen in action in the lower screenshot here, which is based on exactly the same progression as the upper screenshot's block chords.

Ensemble Line-ups

A stereotypical string ensemble will be made up of five sections: two groups of violins (first and second) and one group each of violas, cellos and double basses. In terms of relative numbers, more of the higher-pitched instruments are usually used, with the first violins in small ensembles often outnumbering the cellos two to one. For example, a small line-up might have four first violins, three second violins, three violas, two cellos, and one or two double basses. While it has become fashionable to use a slightly more bottom-heavy group in film work, this can work against you if you're producing a string overdub for a rock track, where it may make sense to ditch the double basses and even the cellos entirely. Another thing to watch out for is any situation where just two of any given instrument are playing together, because tuning discrepancies are much more obvious between two players than they are for a soloist or for groups of three or more.

than the rule. A final tip for keyboardists: avoid lots of fast runs of five notes in a row. These might be easy for you to play on a keyboard, but can be a nightmare for string players because they have to keep crossing strings — a reason why amateur string sections so often struggle with Tchaikovksy and Liszt. Groups of four are usually much easier for the upper strings in particular.

• All Players Are Not Equal

If you're arranging for real strings and have any prior knowledge of the calibre of the players, it makes sense to try to adapt the difficulty of the arrangement accordingly. One of the trickiest things on a stringed instrument is playing musically and with a good tone, so it makes sense to reserve your main themes for the strongest players if you have the choice.

If you don't know the quality of the players before the session, then bear in mind that there is a tendency for educational institutions, in particular, to assign their most capable players to the first violin parts, and the less able to the second violins, while persuading some of those who would otherwise be left out of the ensemble to play a bit of viola as a sideline. The result of this is that it's fairly common for these parts to be less well played (and with inferior instruments) by amateur ensembles. Remember that it is perfectly acceptable for the first violin to duck below the second violin in pitch, or indeed for the cello to hop over the viola, if you feel that this will achieve a more satisfactory musical result.

Writing Parts For Samplers & Virtual Instruments

If you're planning to realise your string arrangement entirely with samples, you're spoilt for choice in terms of the number of sample libraries and virtual instruments available. Most are capable of producing truly lifelike results, but usually only under certain conditions, so getting the best out of them involves adjusting your arrangement a little to play to their strengths.

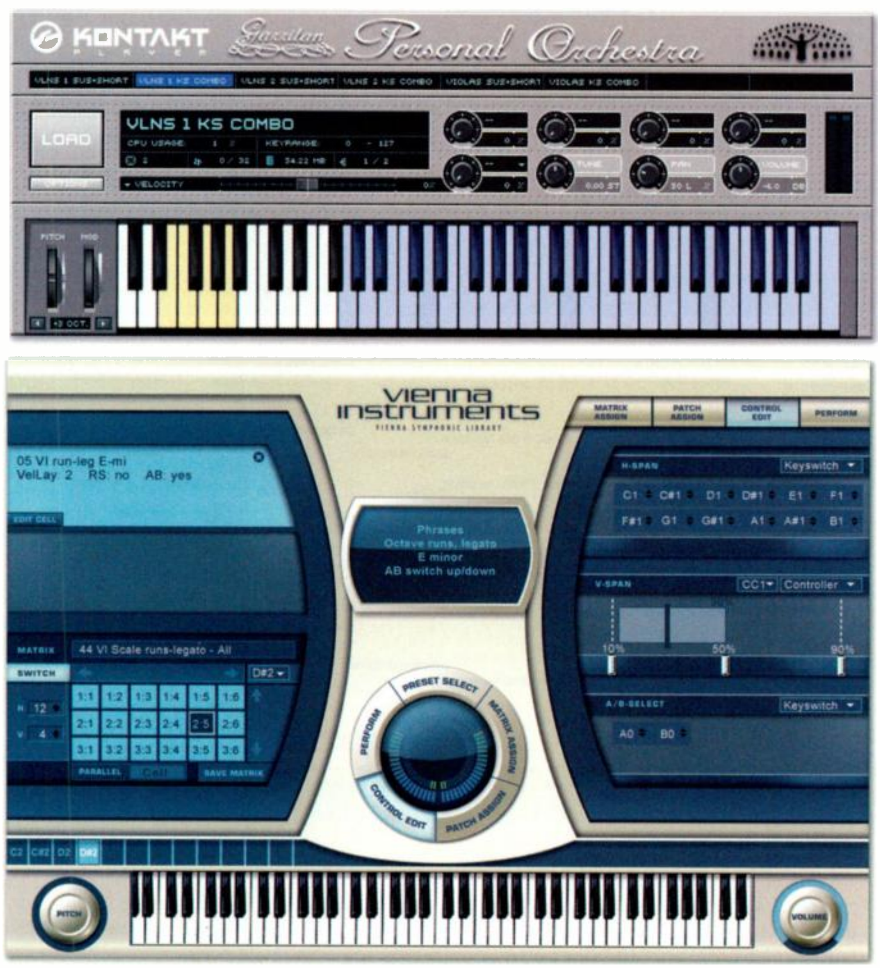
The first thing to realise is that large-ensemble sounds are easier to fake than small-ensemble or solo textures — even state-of-the-art instruments like Garritan's Stradivarius still reveal their computerised undergarments if they're not used carefully. If you have important exposed lines in your arrangement, it will really repay the effort if you record at least one real player and mix them in alongside your samples. This is a tried-and-trusted technique for television and advertising music, and is surprisingly effective at disguising the shortcomings of even budget sampled string patches. I've doubled a real viola solo with a no-frills solo cello multisample from my hardware rackmount sampler (remember those?) without giving the game away, for example.

Other things that really show up the deficiencies of sampled string instruments are fast runs and expressive melodic lines. Both suffer from the inability of most sampled instruments to simulate the effects of slurring — playing every note with a separate bow is pretty unnatural for string players, and even live string players have trouble playing musically unless they can slur some notes together. Good string players will also tend to slide between selected notes in legato lines, breathing more life into a phrase, and this is a feat that few virtual string instruments even attempt to replicate. Again, if you're relying on samples to create your string sound, try to avoid these traps in the arrangement unless you can supplement the sound with a live recorded line or two.

Improving The Sound Of Sampled Strings

The main problem with many sampled string patches is that the sampling process robs the overall sound of some of the complexity you'd get in a real live recording, so anything you can do to reinstate some of that complexity is likely to improve things. Layering several different string patches can help, particularly if you give each a slightly different vibrato depth and rate. You can sometimes get away with less-than-stellar individual patches if you pile up two or three together. What's particularly important, though, is that you try to use fairly contrasted sounds, in order to avoid a phasey combination — many sound modules have a variety of different string patches based on the same raw samples, so try to avoid using those together. A low-level tremolo patch underneath your normal *arco* strings is another way of adding complexity without too many phasing problems.

Layering in a few low-level double-bass pizzicato notes is a useful trick to give a bit of urgency to an otherwise *arco* arrangement, so I'd definitely recommend trying that. Top professional arranger Richard Niles shared another great layering



It's now possible to obtain some very convincing string sounds from virtual instruments and sample libraries, such as those from Garritan and Vienna Instruments, but it can still take a bit of work to recreate the sort of complexities and variations inherent in live performance.

Tips For Recording Strings On A Budget

For detailed information about recording string instruments and ensembles, you can check out Hugh Robjohns' article in the 'Further Reading' box. However, in addition to what Hugh has to say there, I thought it would be worth adding a few little pointers for making the best of the less-than-perfect players and recording situations you're likely to meet working on a budget.

If you're recording a single instrument, the main thing to remember is not to mic too close, because string instruments put out a lot of mechanical noises which can dominate the sound unduly at mic positions closer than a couple of metres away. A bit of room sound isn't a bad thing either, so don't be afraid of using omni mics if your recording room doesn't sound too bad.

For ensembles, the first thing to say is that it's easiest to try to capture the whole sound with a single stereo mic rig if at all possible, and then make up for any balance problems in the old-fashioned way — by routining and respositioning players. Unless you're confident about dealing with the phase relationships between lots of mics, multi-miking makes it trickier to get as good a sound, in my experience. Remember, as well, that you don't have to sit the players the way you would put them on stage, and moving them around a bit can help the balance. In particular, turning the viola section so that it's more on the angle of the second violins can help project their sound better out to the main mics.

Despite my generally recommending sticking with a single stereo mic pair, if you're relying on the strings to supply low end to your mix you'll almost certainly want to put up an additional spot mic on the cellos and double basses as well. This is something Hugh did in the live string session which accompanied the *SOS* May 2006 article, and it's difficult to provide the kind of low end many people now expect in commercial non-classical productions without this extra mic.

Everyone has their own preferred stereo miking methods, but I've found strings (especially violins) to be particularly unforgiving of less expensive cardioid mics, whereas cheaper omnis and figure-of-eights seem to cause fewer problems. For this reason, XY or ORFT wouldn't be my first choice for a budget session. In fact, even with better mics, these setups can emphasise scratchiness amongst any less-experienced first violinists, because of the brighter on-axis response of the mics. I'd suggest using the M/S technique if you can, and if you're using switchable polar-pattern mics as well, changing the pattern of the middle mic gives you a lot of control over the amount of room sound — something that can be really useful in slightly dodgy-sounding venues.

For monitoring purposes, string players usually prefer to wear only one side of their headphones, as this gives them immediate acoustic feedback from their instrument (particularly in an ensemble context) and makes it easier for them to control the sound they're making. Players with little experience of studio recording may not work this out for themselves, so it's worth discussing it with them. If the player is using only one headphone, try to kill the other side of the headphone feed to reduce spill onto the recording, especially if you're using any kind of click track.

tip with us back in *SOS* June 2000: "Whenever I do string arrangements, I will always also use a harp, because harps are very good for that sort of pop 'whoosh' up to choruses."

Unless the string part is very simple indeed, or completely buried in the mix, it's much better to work with separate MIDI channels and sound sources for violin, viola, cello and double-bass parts, rather than just relying on pre-mixed 'ensemble strings' patches. For a start, it lets you have different playing techniques on different parts where necessary, and opens up great textural possibilities, such as high cello melodies accompanied by low-register violins and violas. It also allows you to switch the individual parts into monophonic mode and then use MIDI portamento controls (typically MIDI Continuous Controller numbers five, 65, and 84) in tandem with overlapping MIDI notes to simulate some of the expressive effects of slurring and finger slides.

Irrespective of whether you split the ensemble into separate parts, there are some MIDI programming tweaks worth experimenting with. The first is using a MIDI Continuous Controller to adjust volume levels in real time — CC#7 and/or CC#11 are usually assigned to this as standard in sound modules, but you might need to set it up for yourself in some cases. One of the things that distinguishes real string parts from keyboard string patches is that bows are better at dynamically adjusting playing levels than keyboard keys typically are, and swells are usually an important part of the appeal of real strings. To a lesser extent, using your keyboard's mod wheel (which usually outputs CC#1) to adjust vibrato depth in real time, perhaps only on one of the patches if you're layering, will also make the sound more organic in a way that many musicians are looking for.

If you can get real-time control over the attack and release times of your string samples (often assigned by default to CC#73 and CC#72 respectively), a bit of careful work with these will let you sneak the strings in and out of the texture slowly during your verses and pre-choruses, but then punch through confidently when the chorus riff comes along. In my experience, attack and release times are frequently the secret ingredient that transforms the phrasing of melodic lines into something more like real music.

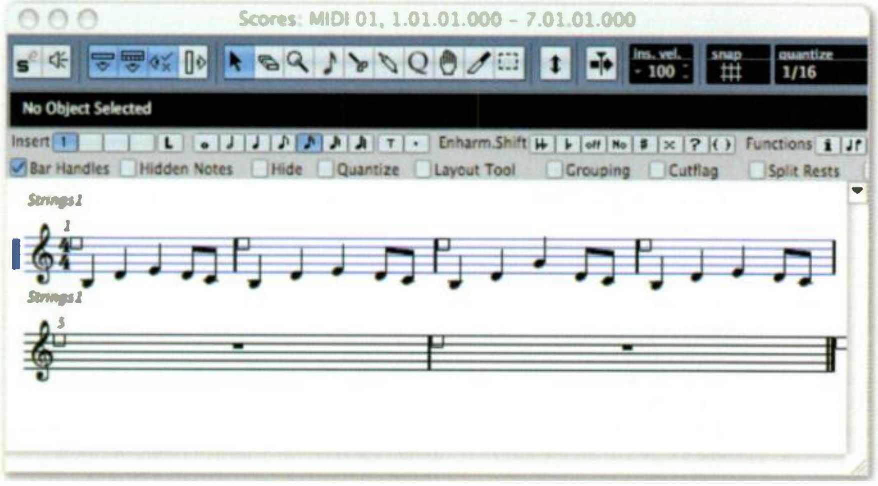
Live Strings: Getting Around The Notation Problem

Nothing beats adding in a few live parts for improving the realism of a whole sampled string arrangement. Multitracking a single violinist playing along with the top line of the arrangement can make a terrific difference just on its own, and this approach can also pay dividends for exposed lines in other parts. However, many home studio owners fight shy of involving live string players because they don't feel confident with creating the notated parts required for all but the simplest of overdubs. Almost all sequencing software has built-in scoring facilities, but what they output by default is often, in reality, practically impossible to play from unless tweaked by someone in the know. Fortunately, for small-scale string recordings this needn't be an obstacle, because the string players themselves can bail you out. Here's what to do...

First, get the string arrangement working to your satisfaction with the sampled sounds running from your sequencer. Any parts you're planning on doubling with live instruments should be separated out onto separate MIDI tracks, and checked to make sure they're in a suitable range. The very highest registers of each instrument are best reserved for special effects, and if you do use them, try to avoid fast runs, which can be very difficult to play without sounding scrappy. Despite what you might find on the Internet regarding the theoretical ranges of string instruments, unless your players are professionals I'd recommend restricting yourself to the following MIDI note ranges (based on middle 'C' being C3), as that will make it much easier for your players reliably to give you a full sound:

- **Violin**: G2 to E5
- **Viola:** C2 to A5
- **Cello:** C1 to A4
- **Double Bass:** E0 to G2 (some double basses go down to C0, but it depends on the specific instrument.)

Once you've checked the ranges, switch over to the scoring window of your sequencer so that it shows the first part, and make sure that it's been given a suitable clef — usually there are built-in scoring templates which can be assigned to each part and which automatically select appropriate clefs, but if you have to select them manually, choose the treble clef for violins, an alto clef for violas, and a bass clef for the cellos and double-basses. Note that double-bass players play an octave below what they see written, so you might need to shift all the notes of the double-bass parts up by 12 semitones if your sequencer's templates don't do this for you in the score window automatically.



If you find notation a daunting prospect, don't worry: you can simply record or program parts as usual and use your sequencer's score editor to communicate the basics to a string player. If you also give the player a paper stave and a pencil, you'll probably find that they're happy to write the part out more fully for themselves.

Finally, quantising the string parts as much as you can get away with (while avoiding actually changing the basic rhythm of the part) is a good idea, as it'll make the printed part easier to read.

And that's all the score editing you should need to do, because although those parts may still be a bit rubbish to play from, you're not actually going to ask the players to do that. Instead, get hold of some pencils and some A4 sheets of looseleaf manuscript paper and ask the string players to jot down a part, based on the raw printout, for themselves to play.

This might seem like an unnecessary annoyance for the players, but there are several reasons why it can work really well. The first is that some string players have a tendency, when presented with printed parts, simply to play what is written, irrespective of whether it's going to sound any good. Putting together their own part based on your printed guide, however, makes it much more likely that they'll be pro-active about discussing adjustments to the part which might improve the sound, either for involved technical reasons, or simply because they are most acquainted with what they can and can't make sound good on their instrument. That said, this way of generating parts works best when there are only one or two players per part — if you're doing any string session larger than that, you'd be best advised to get someone to help you with putting together parts beforehand, to avoid wasting session time.

With pencils already in action, you can also pick the players' brains about which bowings might work best. Faster figurations can often be made much more sweet-sounding when they are judiciously slurred, for example; stamping chords might work better with repeated down bows; and dramatic crescendo swells favour up bows. Expressive bowing is difficult to achieve with samples, so it's one effect which you might as well take full advantage of if you're going to go to all the trouble of setting up a live recording session.

Speaking of crescendos, you'll need to indicate to the players how loud or soft you want different sections to be played, and if there are any important build-ups. The players can then translate your instructions into their parts in the manner that is most useful to them. Furthermore, they'll be able to adjust their bowing to make sure they have enough bow to really let rip when you need them to.

"It's much better to work with separate MIDI channels and sound sources for violin, viola, cello and double-bass parts, rather than just relying on pre-mixed 'ensemble strings' patches."

One final thing I always insist on is that everyone writes bar numbers into their parts (to match up with the bar numbers in your sequencer), as this makes rehearsing and recording takes a lot quicker and easier.

Session Direction

You've done the arrangement. You've organised your string session. You've got notation in front of each of the players. So it should all be downhill from here, right? Well, not quite...

If you were working with top pros, you could pretty much sit back and relax, knowing that they'd be doing everything possible to make you (and them) sound good. However, most budget string-sessions rely on the services of amateur players who, as much as they're a whole lot better at playing their instruments than you are, can still often benefit from a little objective musical direction. So an important job you need to do during the recording session (in addition to handling your studio gear) is to listen critically to what is coming through the monitors, to be sure that it's all sounding the way you want it to.

There are lots of things to be on the look-out for, with less experienced players in particular. First of all, just as with guitar parts, check that the strings are in tune — this is especially important if there are any prominent open-string notes in the arrangement, as players are unable to adjust these by ear while playing, in the way they can with stopped notes. Don't be afraid to ask players to tune individually if necessary, as some amateur string-players won't concentrate enough on it otherwise. Even if the instruments are perfectly in tune, you still need to keep an ear out for dodgy tuning, much more than you might expect to with fretted stringed instruments such as guitars. In particular, listen out for any situations where pairs of instruments are doubling each other in octaves, as this can really expose tuning inaccuracies. Draw the players' attentions to any such doublings, so that they're primed to listen to each other carefully, and encourage the lower part to play a little more strongly than the upper part, as that tends to make small discrepancies less obvious.

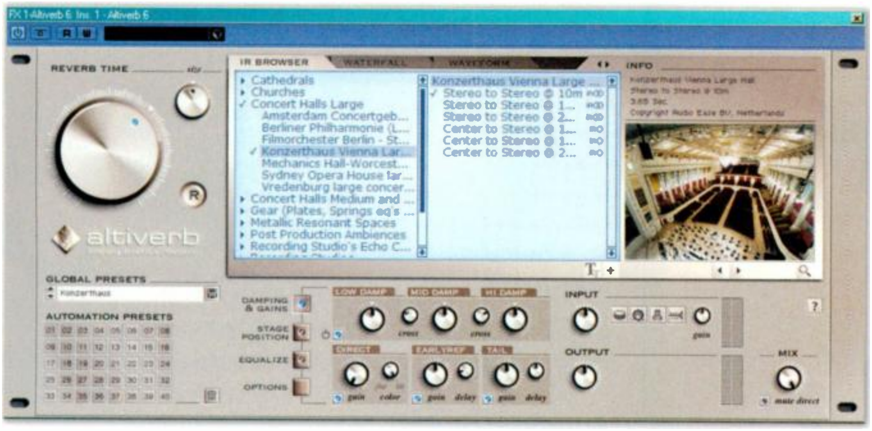
Something I always make a special point of listening for is violinists playing their open 'E' strings. The 'E' string is actually of a different construction than the instrument's other three strings, and as a result has a habit of zinging out rather too piercingly compared with either stopped notes or any of the other open strings — particularly on instruments which don't cost the same as a three-bed semi-detached! Keep a close ear out for these, and if they are becoming obtrusive then talk to the players about possibilities for avoiding them, by playing the note on the neighbouring 'A' string instead. There is almost always a way around playing open 'E' strings — in fact there are apocryphal tales of some professional orchestras fining violinists for playing any! Violas can also exhibit the same problem with their top 'A' string, although

String players usually have a choice of strings upon which to play any given pitch, although this involves moving the left hand from its default position near the tuning pegs. These four pictures here show the same pitch (A#) being played on the four strings of a viola in this way. Although it takes a little more concentration to play with the left hand out of its default position, the other hand positions provide different (and often more suitable) tonalities that sample libraries rarely include.

it's usually offset a bit, in my experience, by the viola's more mellow overall tone.

Fingering Positions & Ensemble Timing

Another important thing to bear in mind is that string players have a lot of choice as to which string they actually play any given note on. For example, the 'B' above middle 'C' could be played on any of the violin's lower three strings, but on the lower strings the player has to move their left hand away from its default location, close to the tuning pegs (this is called 'first ▶



The key consideration when mixing is your choice and use of reverb. A good convolution reverb, such as Altiverb (pictured) can make all the difference, and if you're mixing and matching different sounds, sending some of the signal to the same reverb can help make things sound much more 'together'.

position', and is where playing is usually easiest) and further up towards their nose (into a higher fingering position, which demands a little more concentration from the player). Many string players, left to their own devices, will blithely play your whole chart in first position, because it's easier for them, despite the fact that higher positions often provide a warmer, more evocative tone. So if an important melodic line is sounding a bit thin and/or edgy, try asking the players to play the same notes on the string below, where the necessity of playing in a higher fingering position will frequently round out the timbre.

Fingering positions can also really help to add something special if there are melodic lines that include large upward leaps. The easiest way to play these leaps on a stringed instrument is by crossing over to a higher string. However, if you stay on the same string instead — and do the leap by moving the left hand into a higher fingering position (called 'shifting position') — you get a characteristic little pitch glide which can be very emotive. This is something that characterises good live string playing, and that samples simply can't do for you.

For recording ensembles, timing can be another common issue. Some less confident string players have a habit of waiting a fraction of a second at the starts of sections — until they're sure their neighbour is playing! This can turn what should have been forceful entries into damp squibs. The easiest solution to this problem is breathing. Yes, I know that these aren't wind instruments, but the simple act of everyone taking a breath together on the beat before the entry is usually enough to get all the players coming in firmly on cue. If the ensemble are shifting out of time with each other midway through a section, try to encourage them to move a little with the beat, because the extra visual cues this sets up between them will help here.

Something people commonly want from string parts is intensity and passion, but some classically trained string players can be inclined towards a more measured delivery. A couple of quick ways I've found to encourage more gusto are asking the players to use more bow, or to use more vibrato. A side-effect of the latter request is that most string players will press slightly harder on the string with the fingers of their left hand, giving notes a greater sustain and a clearer sound-quality — great for nice fat pizzicato notes.

Mixing Considerations

The main consideration when it comes to mixing strings, whether sampled or real, is your use of reverb. Many sampled sounds are deliberately much drier than real recorded string sounds, so a good concert-hall reverb (something convolution-based would be ideal here) can enhance the impression tremendously. On the other hand, recorded live strings almost always have some reverb on them from the room they were recorded in, but when you're working on a budget they'll still benefit from something to give them a bit more of a 'big studio' sound. You may find that using a reverb patch with fewer

Jargon Buster

- **Arco:** The normal string-playing technique using the bow.
- **Bowing:** String players can play with the bow by pushing their right hand either away from the instrument (a 'down bow') or towards it (an 'up bow'), and can also choose how many notes to play in a single bow stroke. The specific bowing that they use can dramatically affect the sound, so it is common practice for players to decorate their notated part with bowing indications for the sake of consistency.
- **Clef:** The notation symbol which usually appears at the left-hand edge of each line of traditional musical notation. It is used to indicate which lines and spaces correspond to which pitches.
- **Col Legno:** An unusual playing technique, where the wood of the bow touches the string, giving a thin, ghostly, glassy sound much loved by film composers for sci-fi and horror scenes.
- **Con Sordino:** These words in a printed part indicate to the players to attach a little mute to the bridge which gives the sound a more closed, reedy quality of which I'm a great fan. The mutes are detachable, though, and are easily lost, so if you plan on using mutes for a recording session, be sure to mention this to the players in advance.
- **Fingerboard:** The slightly curved black strip of wood running down the centre of the instrument where the player uses the fingers of their left hand to stop notes, effectively changing the length (and therefore the pitch) of the string.
- **Fingering:** String players can choose which finger to use on which string to reproduce a given notated part, and because there are often several different fingering options for a given piece of notation, players will often write numbers and Roman numerals above the notes to clarify which to use as they read.
- **Legato:** Smooth playing, often using only a few bows for many notes.
- **Monophonic:** Instruments are monophonic if they can play only one note at a time. Although string instruments are not strictly speaking monophonic, playing more than one note at a time (double-, triple-, or quadruple-stopping) is technically difficult to do and you'll almost certainly get nicer-sounding results on a budget session if you avoid it.
- **Open String:** A note played without stopping the string using the fingers of the player's left hand. This gives a more strident sound which rings on for longer.
- **Portamento:** When working with a MIDI instrument in monophonic mode, you can often set things up so that overlapping notes have an automatic expressive pitch-slide between them, called portamento (or sometimes 'glide'). If portamento is available, then there is usually some control over the rate of the pitch slide, via a parameter called Portamento Time.
- **Pizzicato:** Plucking the strings with the fingers of the right hand.
- **Semitone:** The pitch interval between adjacent MIDI notes, for example C4 and C#4.
- **Slur:** A curved line in traditional musical notation which indicates to the string player that several notes are to be played with a single stroke of the bow, giving smoother transitions between notes.
- **Staccato:** Detached, spiky playing style, contrasted to legato.
- **Tremolo:** A playing technique where the bow is moved backwards and forwards a small distance as fast as possible. In ensembles, this gives a shimmering effect, often used to indicate tension or suspense by film composers.
- **Trill:** Rapidly alternating between two closely-spaced notes, using the fingers of the left hand, while slurring them all together with the bow.
- **Vibrato:** An expressive pitch modulation introduced by wobbling the left hand back and forth during playing. String players learn to adjust the nature of their vibrato as each note progresses, for a variety of effects.

Layering Takes To Create An Ensemble Sound

It is very common for people to attempt to create the sound of a larger ensemble by layering together several takes with only a few players, but this usually gives a rather unrealistic kind of chorus effect instead. To avoid this, it's vital that you try to differentiate every take sonically. Many instrumentalists have more than one instrument, for example, so try layering takes of both — a pair of friendly violinists might even be happy to swap instruments between takes. Try getting the players to play some takes in different fingering positions, or *con sordino*. Change mics and mic positions to change the level and nature of the captured room sound, or, as an alternative with a one-person-per-part ensemble, set out chairs for a larger orchestral setup and then record subsequent takes with the players seated in the different chairs.

When you're short of players and you need to overdub, a good trick is to record the different takes in different positions. Keep the mic a fair distance away, so as to capture some room sound, which will help to make the tone different for different takes, thus creating the impression of different instruments and avoiding potential phase problems.

early reflections and/or a longer pre-delay time will help to avoid it conflicting with the ambience within the recording itself.

If you're combining real string overdubs with a sampled backing, you should try to use at least one reverb which is common to both parts, to help them blend together, although I find that you usually need a little extra ambience on one of the parts as well, to sit it with the other. The big problem with adding reverb, though, is that it can quickly clutter up your mix, so be prepared to shorten reverb times, particularly when working with up-tempo tracks, and consider high-pass filtering the effect return — the low frequencies of concert-hall reverbs, in particular, can be problematic in commercial-style mixes. Other than reverb, a little chorusing can help to add complexity, and particularly stereo width, to strings, but if you overdo things you can end up with a sound like a string machine!

The string tracks themselves can easily clutter up a busy mix, even without added effects, and you'd be surprised how severely you can high-pass filter them in many cases — you may just need a slightly higher level of lower instruments in the string mix to help maintain the subjective balance in the remaining frequency regions. The other problem is that strings are often required to switch roles a lot, from background pad to foreground hook, which can make a decent static mix balance impossible to achieve for many arrangements. The most powerful remedy to this is level automation, which lets you dip the levels when the strings are holding background chords, and then just nudge up any little chord changes, fills, or melodies as required. Automated EQ may also help here, to give a little extra brightness and definition against sections of the song with lots of HF-rich instruments such as cymbals and distorted guitars.

Mantovani To Metallica

Whatever style of music you're working in, the desire for a nice, lush string sound seems to crop up once in a while. Hopefully this article has provided you with enough information to achieve reasonable results on those occasions, without having to shell out loads of cash or spend ages wrestling with notation. SOS

Further Reading

• Wikipedia
Wikipedia is a surprisingly good bet if you want to find out a bit more about the workings of the different string instruments. Although people sometimes question the reliability of some of the user-generated content on Wikipedia, the pages covering violin, viola, cello, and double bass seem to me (as a string player myself) to be pretty much on the money.
W www.wikipedia.org

• Rimsky-Korsakov's *Principles Of Orchestration* — Garritan Interactive Version
Here you can find a free on-line version of one of the classic orchestration texts, complete with interactive audio examples realised by Garritan's Personal Orchestra. If you can handle notation and want to hone your string-arranging skills, there's a lot to be picked up here.
W www.northernsounds.com/forum/forumdisplay.php?f=77

• Recording A String Section: Theory & Practice
Recording a very large string section is a task that is beyond the scope of the current article. Thankfully, though, this subject was covered in depth by Hugh Robjohns, back in *SOS* May 2006. As well as taking you through mic techniques, this article also includes some very useful comparative audio examples.
W www.soundonsound.com/sos/may06/articles/recordingstrings.htm

Euphonix MC Pro

Assignable DAW Control Surface

Euphonix's EuCon control protocol first saw the light of day in their high-end digital mixers. Now it has the potential to revolutionise the world of the project studio.

Hilgrove Kenrick

Skywalker Sound, Todd-AO, Digital Factory, Pinewood and Shepperton, Park Road Post: the install list for the Euphonix System 5 console and its variants covers most of the big names in post-production. Synonymous with very high-quality engineering, and chill-inducing price tags, Euphonix are up there with Lawo, Harrison, AMS Neve and SSL in the realms of real pro audio.

Now Euphonix have turned their hands to a pure control surface with no audio pathways, and in doing so, have attempted possibly the most ambitious surface ever devised: one with no ties to a specific platform or operating system, and a truly open-ended architecture. On its own, this controller is known as the MC Pro, but it can also be paired with fader banks and other extras to create the System 5 MC (see the 'Banking On The Future' box).

On most control surfaces, buttons, switches and, to a lesser extent, sliders and knobs have fixed purposes, or a limited set of alternatives. In general, functions are locked to certain buttons, be they for transport controls, track arming or whatever: so to operate the product, the user must conform to the designer's idea of how to work. As mentioned before, they are also commonly locked to a limited number of platforms.

With the MC Pro, Euphonix have torn up the rules for control-surface design, to create something completely flexible. To boil it down, if you want this particular control here, it'll let you put it there. If you want to manage

The trackball and shuttle wheel feature identical layouts of assignable controls. These use Euphonix's smart switches, which are at the core of the MC Pro's power and flexibility.

an operation this way, it'll let you — there are no dictates on how you should work. Although it arrives with a default template installed, the user is free to adapt it to his or her own way of working.

Smart Switches

At the core of the MC Pro's capability are the so-called smart switches. These buttons have little embedded LCDs capable of displaying text and bitmap images, and of turning blue, orange, red or green (or from one to the other to denote on or off). They embody everything the MC stands for — they can display anything, just as they can control anything, and while each has a default command assigned to it, every individual smart switch can be set to do whatever you want. All the sections also have bank controls, a bank being a layer of set functions for the smart switches — change bank, and they all change to a different set of commands. As we go through the surface, I'll describe the default functions of the controls, but keep in mind that any or all can be different if you so choose!

Starting from the left on the lower half of the MC Pro, one finds the left-hand Edit Control, a wonderfully weighty and solid jog/shuttle wheel with four bank buttons to select different control layers on the smart switches. The smart switches themselves are arranged in a blue column of six down the left, and default to zoom and nudging controls. In a row across the top are a further 10 orange buttons, which default to the likes of Trim L/R, move, snip, fades, clip gain and mute.

The middle of the lower half houses a standard QWERTY keyboard recessed into the surface. To the right of this is the second Edit Control. In terms of its visual layout, this is a mirror of the left-hand one, but instead of a jog/shuttle wheel it uses a trackball and cuff surrounded by four click buttons. The smart switches on this side default to green and red transport controls across the top, and blue marker and region selection buttons down the right.

Moving to the upper half of the surface, the left-hand end houses the monitoring controls. This section interfaces with either the Control Room in Nuendo, or the supplied Studio Monitor Express software, giving you hardware control over monitoring levels for three sets of outputs — the top pair scroll through your different monitor feeds, the lower one is fixed to the control room. Each has its own knob and LCD scribble strip, and there are master dim, cut and speaker switch buttons.

Also in the section are master buttons for clearing mutes or solos for the entire project, which usefully light up to denote if either status types are present. Below these is a big Talkback button for — yes, you've guessed it — talkback, either using the tiny internal mic on the top panel of the MC Pro, or a mic plugged into the rear.

SOUND ON SOUND

Euphonix MC Pro £10,281

pros

- Ultimate flexibility.
- Works with virtually any software.
- Simultaneous control of multiple workstations.
- Did I mention flexibility?

cons

- No VST Instrument control in Nuendo yet.
- EuCon licence costs vary from platform to platform.
- The prototype's ability to swap the positions of the jog wheel and trackball has been lost.

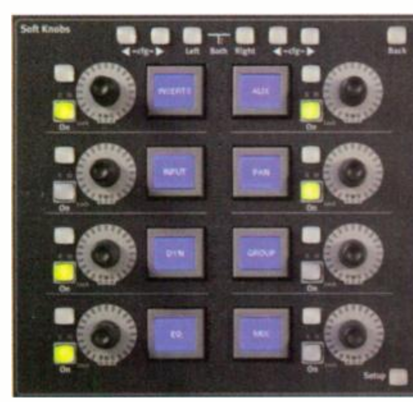
summary

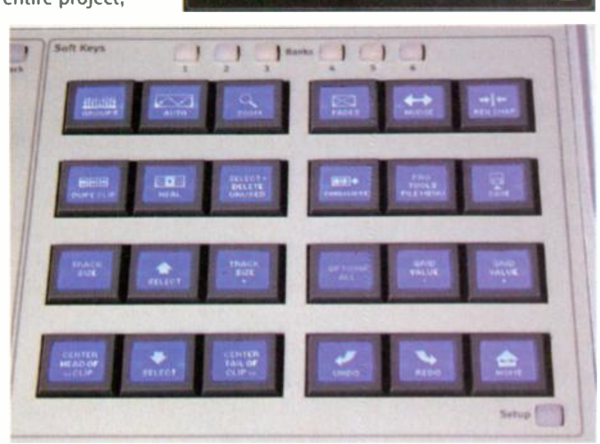
If you want a control surface that will allow you to work the way you want to work, rather than the way the manufacturer thinks you ought to work, this could be the one for you.

Going Soft

Across from the master section is the Soft Knob section, which will be one of the most utilised sections of the surface. Although it's laid out in a square, it's easiest to think of these knobs as being a single channel-strip, as they perform control duties for EQ, dynamics, aux sends and plug-in controls. Each of the eight knobs has its own smart switch, automation read/write and on/off buttons, and the area is capped by configuration and banking buttons. In operation in, let's say, EQ mode, the smart switches change to reflect the frequency or Q they are controlling, with the LED halo around each knob providing a quick-glance overview of its position. Change to Dynamics and they display threshold, release and so on instead. As we'll see later, this area allows you to call up, insert and edit plug-ins. This section also features an Assignable knob — again, more on this in a minute.

Next along is the largest collection of smart switches — 24, in four rows of six — forming the Soft Keys section. Six bank switches across the top give access to a total of 144 commands. As a quick overview, bank number one covers the likes of (create) New, Save and Open, along with window commands to show or hide the mixer, video window and so on. On top of the six banks, some individual commands are further nested, such as the tool menus: select the file button and the switches change to reflect ▶





The Soft Knob and Soft Key sections are where much of the MC Pro's power resides, as these can be assigned to control complex functions or sets of functions within your DAW.

EUPHONIX MC PRO

The touch-screen's default mode is a matrix display of all the tracks within your project.

the commands available to you, just as if you had clicked the mouse on the file header on the toolbar.

Tilted towards the user, the next item on the tour is the TFT touchscreen, which is central to the MC Pro's control capabilities. The main Tracks window gives you a matrix overview of all the tracks in your project, including their solo, mute or arming status, along with names and track numbers. There's also a useful SMPTE and MBT (Measure, Beat, Tick) readout. Touching a track automatically highlights it and brings all of its respective controls to the surface. The touchscreen is also used to program the MC Pro, through simple menus to choose what commands are assigned to what controllers, and what colours, text or bitmap images the smart switches should display. Each separate section of the surface has its own Setup button, which brings up a menu on the touchscreen to edit the controls.

Finally, to the right of the screen is either a bank of four 100mm faders or a pair of motorised joysticks. In the case of the faders, each has six-bar-plus-clip-LED input and output meters alongside, with buttons for arming, channel select, automation and the Euphonix Wave-key which brings that track into focus on the rest of the MC Pro. They're topped off with master solo and channel on buttons. On the right-hand edge are nudge and bank controls for the faders, to switch their focus in banks of four or step along one by one. At the top are the four Workstation select switches, to chose which EuCon-connected workstation you want to control, more on that (again!) later.

To conclude the tour, the rear panel houses the RJ45 Ethernet port and a screw-lock connector for the external power supply. This is a modified PC unit with a short cable for putting it on the desktop. Incidentally, the only fan throughout the MC Pro is in this PSU, and I only found out after asking Euphonix how they'd managed to make something without any! It is that quiet. Handy inclusions are a headphone input jack which passes through to an output on the underside of the front, footswitch jacks for pedal activation of talkback and an XLR for outputting the internal talkback microphone. A PS/2 connector enables you to attach another keyboard, a VGA output allows you to connect a larger external screen, and a single USB socket is used for connecting a memory stick or the like to transfer data and updates.

Networking

Installation is fairly straightforward. The MC Pro connects via Ethernet, and ships with a four-port router which must be connected up to the surface and whichever workstation(s) it's going to be controlling. One point to note is that it must be on its own separate network to anything else you have going on — in my case I used my own network for MIDI-over-LAN and sample streaming — as the EuCon protocol it uses is both high-bandwidth and time-critical, and you don't want anything else getting in the way of the data packets.

To get EuCon running on your DAW machine, an applet is installed into the operating system: Windows XP and Mac OS X are supported in 32-bit configurations only at present, with 64-bit support and Vista compatibility to follow shortly. The final hurdle is the EuCon licence, which comes in several forms; in the case of Nuendo it's a separate licence which is installed on the Synchrosoft dongle alongside the other Steinberg licences. Pyramix users must have Merging's own control protocol, OASIS, installed, as EuCon dovetails with that. For Sonar 7.02 or Logic 7 and 8 users, nothing need be done, as EuCon is already built in to the software. If all you are planning on doing is controlling a HUI-capable application, then no licence is needed, and the basic OS applet itself can be installed on any machine purely for the purposes of keyboard and mouse control if needed.

Flick the switch on the desktop PSU (why oh why won't it reach the floor?), and the touchscreen displays a normal BIOS POST screen, followed by Windows XP loading. A minute or so later, the MC Pro GUI appears on the touchscreen, and a prod of the trackball moves the cursor on your DAW machine. At first, the MC Pro's buttons are largely blank, as we're navigating around Windows XP, but on launching Nuendo 4, suddenly all springs to life, with all the previously mentioned control defaults lighting up across the surface, the track list of your current project filling the touchscreen, and the monitor control LCD telling you the current setting of the control room output.

Smart Work

Using the MC Pro on the default command settings, the logic of it immediately becomes clear. It's an absolute joy to be able to edit fades and lengths, splitting and muting sections whilst zooming in and out where necessary, all without moving hand from the trackball or jog/shuttle. This is truly what control surfaces are for: to put the user back in control, and eliminate the need for endless clicking, menus or tool-type changes just to carry out simple operations.

More logical still are the soft knobs, especially considering the number of different things they can control. Touching an audio channel on the screen brings it up on the knobs and their associated smart switches, offering EQ, dynamics, auxiliaries, inserts and so on. As detailed earlier, pressing on the EQ switch activates your DAW's channel EQ, with the requisite values appearing on the switches. This also opens the EQ window on

The MC Pro has just four motorised, touch-sensitive faders.

EuCon Developments & Support

The Euphonix Control Network (see news item in last month's *SOS*) was originally developed for the company's System 5 consoles, but is now supported natively in several of the top DAWs. At the time of writing, Cakewalk's Sonar 7.0.2 had just joined the list, along with Apple's Logic 7.2.1 and 8, Steinberg's Nuendo 3.21 and 4.1, and Merging Technologies' Pyramix 5.0.12.

A EuCon Client applet is installed on your computer, along with XML templates that create Applications Sets for programs. The EuCon Client also emulates Mackie's HUI protocol, so the MC's faders, buttons and knobs can be used to control applications that don't support EuCon directly, like Pro Tools. Unlike HUI, however, EuCon has inbuilt support for pointing devices and standard keyboards, as well as faders and other specialised controllers. This is what makes it possible to assign an MC Pro button to a whole series of key combinations, menu commands and macros, to create single-button shortcuts for complex actions.

The down side is that some DAW producers charge for EuCon support, such as Steinberg (£611 including VAT) and Pyramix (£176.25 including VAT), while others make it free (Apple and Cakewalk). The Steinberg fee seems rather excessive, particularly given the premium price already charged for Nuendo.

the DAW, so you have two choices of where to look and see what you're doing.

The MC Pro's default Nuendo setup makes it easy to access and edit plug-ins entirely from the controller. When you hit the insert button, you first pick which of the eight slots into you want to place the plug-in, and then the smart switches change to reflect the plug-in hierarchy on the DAW. In my case, this means Nuendo's default listings of delay, distortion, dynamics and so on, but it also picked up my Powercore folder and all the other installed VST plug-ins. The paging buttons at the top are used to scroll through lists of more than eight plug-ins.

Pressing the Dynamics button, for instance, changes the switches to list all the plug-ins that come under that heading. Choosing one results in the relevant plug-in window popping up on the screen, the plug-in appearing in the chosen insert point and, without further ado, all the relevant parameters appearing across the switches numerically, their values indicated by the LED halos around the knobs, ready to tweak away. To do so much without having to move hands around, or switch between keyboard, mouse and controller, speeds up working practices no end. The knobs are touch-sensitive, and capable of measuring acceleration and deceleration in both directions.

The four faders, too, are customisable: you can bank around your project in batches of four, nudge them along one by one or permanently lock faders to specific tracks. In the last case, there's an extra parameter which, again, demonstrates the design ethos: lock to 'attentioned' track. When this is selected, that fader will always show values for whatever track is highlighted in the DAW, so there's no confusion of doing one thing with a mouse or knob and then finding you need to scroll around to find the right track — it's there in front of you at all times. The track metering, although small, is handy for quick reference, especially if your DAW window is busy looking elsewhere.

Turning to Nuendo's Control Room feature, much used by those working entirely 'in the box', the MC Pro automatically picked up all my settings, giving me instant control over control room, studio and headphone feeds, ▶

thus making an additional monitor controller entirely superfluous.

Over and above even the clever little smart switches, there is one single control on the MC Pro that, for me, is the pinnacle of everything Euphonix have achieved with this surface: the assignable knob. Looking rather desolate and unwanted, with its single accompanying smart switch, this little controller languishes in no man's land on the surface itself, and only warrants a few lines in the manual. However, it will control just about anything your mouse pointer hovers over, with the parameter name and value appearing on the smart switch alongside. Press down on the knob, and it locks to that value, so even if you move the cursor away it will still control it. This is particularly handy for things like snap values when editing audio fades or cuts — while hovering around various audio files in a project, I could change snap values with a quick twiddle.

Chopping & Changing

So we've seen that the surface controls an enormous amount, even with the default settings, but what about programming it? Much to my relief, programming is a slice of Victoria sponge, so even the most technically averse can manage it. You simply press the dedicated setup button for the area of the MC Pro you want to change, and then hit the smart switch you want to customise. The switch then flashes and the touchscreen brings up all the options for that key. These are divided into four categories: key, EuCon, bank and MC commands. This is where the real power of the beast lies, as between all of these one can do practically everything except make coffee, and one suspects they're working on that!

Without going into too much detail, EuCon commands are any commands available via direct communication with your DAW. In the case of Nuendo, this list is enormous, and is fortunately broken down into subsections such as track controls, file, export and so on, but suffice to say practically everything you could do with a key command or mouse click is on offer. Bank commands are for quick changes of banks of switches, and MC commands are for surface-specific controls like full layout recalls.

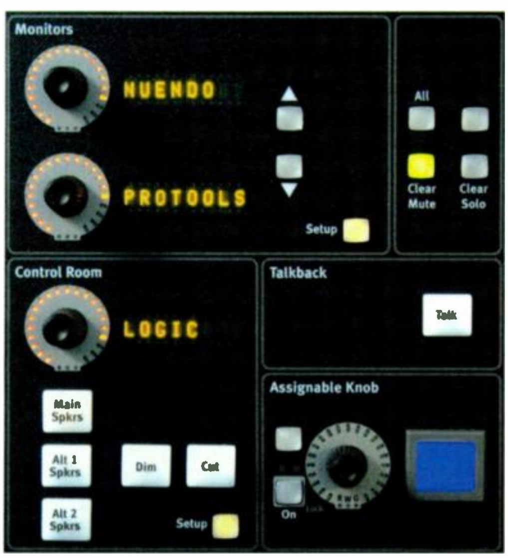
While the EuCon commands already offer a level of customisation and power that most other surfaces lack, Euphonix have topped off the list with the key commands. These can be either single commands, such as Ctrl-V to paste, for example, or strings of commands. It is thus possible to build up complex macro commands, so that a single button-press can set off a string of events which would otherwise take many.

To try and get your head around all the available combinations would bring on a quick one-way trip to the funny farm. Stepping back from such calculations, it's simpler to look at it as truly open-ended architecture — if you want to do something, it'll do it; if you want this command there, then put it there.

What Makes The MC Special?

In terms of bang for the buck, simple comparisons don't always tell the whole story. For instance, the MC Pro has fewer faders even than the Mackie Control, and nothing like as many as the WK Audio ID surface, a Digidesign C24 or the more expensive Icon, but what it does have is those shiny little smart switches. The very fact that the MC Pro is littered with them, and that they're almost infinitely programmable, makes for a very different approach to hands-on control. There are other controllers with jog wheels, QWERTY keyboards and assignable buttons built in, but rarely, if ever, have they all been so carefully integrated as a whole.

Let's give a real-world example. Suppose I have an audio take I want to edit and export in Nuendo. In normal operation I would have to do the following: grab mouse, select snap resolution, move to and select the clip, then right-click (or use the keyboard) to choose trim, slip, split, cut and so on. If I need to zoom in or out, I either have to grab the sliders in the corner of the project window or use keyboard shortcuts to do it bit by bit. Once all of that was nicely tweaked and ready, to export I'd have to go to the file menu, chose export, and then set name, format and bit rates.



The MC Pro itself has no monitoring hardware apart from a headphone amp, but its monitoring section can be used to handle monitoring within a DAW such as Nuendo.

Test Spec

- PC with Intel Core 2 Q6850 CPU and 4GB RAM, running Windows XP Pro 32-bit, with RME HDSP MADI soundcard.
- Tested with Steinberg Nuendo 4.1 and 3.2.1.1153.

Doing the same operation on the MC Pro, I hardly have to move my hands at all — the smart switches over the top of the trackball default to the primary edit controls, so without moving my hand from that area I can do all my trims and fades, or move the clip entirely. To help with such operations I usually have the assignable knob set to snap resolution, so I can alter it on the fly as needed. For zooming, the smart switches down the side of the trackball default to zooming in vertical and horizontal planes, easily adjusted with the ring around the trackball.

Finally, to export, I have a choice — I could either choose the file menu from the block of 24 smart switches and select Export from there, or I could just drop the Export command straight onto a smart switch by the trackball. To take it to the limit, it's even possible to trigger a Nuendo macro bringing up default title, path and format settings for the export, so hitting that key would just execute the export without needing any further intervention on my part.

As with many such exercises, it takes longer to explain than do, but it does serve to show how much can be achieved with little hand movement, and all on the same surface, without having to change between surface, keyboard and mouse. Don't forget that even if you're controlling a HUI application, not a EuCon enabled one, much of this is still possible, because you can still string together key commands or menu operations and assign the results to MC smart switches.

Going Global

Now let's cap it all off with the MC Pro's ace in hand: the four Workstation Select buttons. Thanks to EuCon's network architecture, connecting the optional KVM (Keyboard, Video, Mouse) switcher lets the MC Pro control up to four different workstations simultaneously, even if they're running totally different software! The advantage of connecting the KVM switcher is that your monitor changes too, so you can see precisely what is going on. This is a particularly attractive feature for the post community, as it's not uncommon to have stems from ADR, foley, music and so on being edited on different platforms. Previously you would have had to do all sorts of exports, but now you can deal with them all from one surface, saving lots of time. You could even lock fader banks of a System 5 MC (see box opposite) to specific workstations, so no matter which

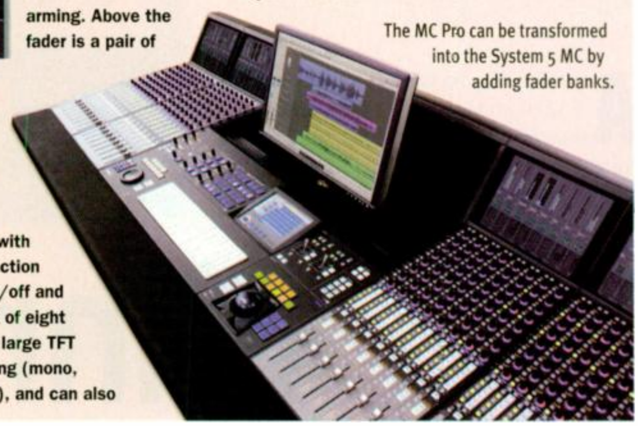
Banking On The Future: MC Pro Expandability

The TFT displays on the System 5's fader banks boast powerful metering features, including the ability to meter surround channels up to 5.1.

The MC Pro is, as this review will make clear, a complete controller in its own right. However, it's also possible to use it as the heart of a much larger control surface by adding optional fader banks. When so equipped, the resulting surface is known as the System 5 MC, to bring it into line with its bigger stablemates.

The eight-channel CM408T fader banks are based on the surface design of the existing Euphonix System 5 consoles, but with newer control knob design and a few other tweaks. (The new fader banks can also be retrofitted to 'proper' 5-series consoles.) Each channel strip comprises a long-throw 100mm fader with input metering and buttons for solo, mute, swap and channel arming. Above the fader is a pair of four-character LCDs naming the channel and its swap (the layer below). Next come the knob function and selection switches, above which are the like of eight touch-sensitive knobs, each with an LCD to denote current function or value, and switches for on/off and automation. Finally, the bank of eight channels is surmounted by a large TFT screen which handles metering (mono, stereo, LRCS or 5.1 per strip), and can also display EQ curves, plug-in parameters, and other channel data — far more comprehensive than just a strip of LEDs.

It gets seriously clever when you're controlling several workstations for a final mix. Conceivably, you could have a Nuendo system for music, Pyramix for ADR and Pro Tools for effects, all in the same rig. Not only will the workstation buttons switch between these, allowing you direct control, but you can mix and match on the fader surface, locking specific channels from each workstation to specific faders. That way, no matter which machine you were switched to control, those faders remain locked to their sources. Very useful if you're pre-mixing the effects and want to quickly pull down the dialogue to check something, as you just grab your allocated master fader for the Pyramix feed.



The MC Pro can be transformed into the System 5 MC by adding fader banks.

machine you were in direct control of, those faders remained constant. It's worth noting that, as EuCon-enabled software includes the likes of Final Cut Pro, you could use the same surface for film *and* audio editing duties.

Flexible Friend

I first saw the MC Pro at Euphonix UK HQ in the early summer of 2006, and despite being in a rather small room kept company by a 32-fader System 5 MC and 32-fader System 5P Hybrid on the hottest day of the year, I was enthralled. I have long been an advocate of open systems, which allow you to work the way you want to work, not the way some manufacturer says you should.

Will the MC Pro make you sound better, get you more work or change the way you work? Unlikely, on any charge. However, it will support you in the way you work, speed up processes and operations, and inveigle itself into your way of operating in such a transparent and seamless way that you will rapidly fall into that age-old cliche: how did I ever work without this! Moreover, it doesn't seem to matter what platform you change to in the future, be it PC, Mac or the same hardware but a different DAW: the MC Pro will continue to do its job, rapidly justifying the investment.

Granted, at first glance the Euphonix MC Pro can, and does, seem expensive for a control surface, especially when it is paired up with the optional fader modules, but when you consider the breadth of its capability with applications the comparisons begin to fail. A 32-fader System 5 MC would be on the same purchase option list as its obvious competitors, the Digidesign D-Command and D-Control, yet as well as boasting better metering and a larger number of controllers, it has obvious advantages for those who are working on multiple platforms. No other manufacturer has yet implemented multi-workstation support as thoroughly or as flexibly as Euphonix.

One curiosity is that during the course of this year, Euphonix have slightly changed the design of the MC Pro. In the original design, both sides had identical wells for the trackball and wheel, which allowed you to choose which side to have the jog and which the ball, and swap them in seconds. As a southpaw, this had instant appeal to me, as I'm sick of right-handed designs. Sadly, the new jog wheel, while very nice to use, cannot be swapped over. Euphonix can supply the MC Pro with jog wheel and trackball on the sides of your choice, but it seems a shame to lose such a handy feature, especially for studios that need to cater for different engineers.

My biggest complaint, as a composer rather than an engineer, is the lack of control for software instruments in Nuendo. After all, many of the plug-in windows I open are instruments rather than exclusively audio processors. I am reassured, however, that this facility is being dropped into a future update, so with a bit of luck it'll be in there shortly. It seems that such things are down to the DAW manufacturer to integrate, not a failing in the EuCon protocol itself.

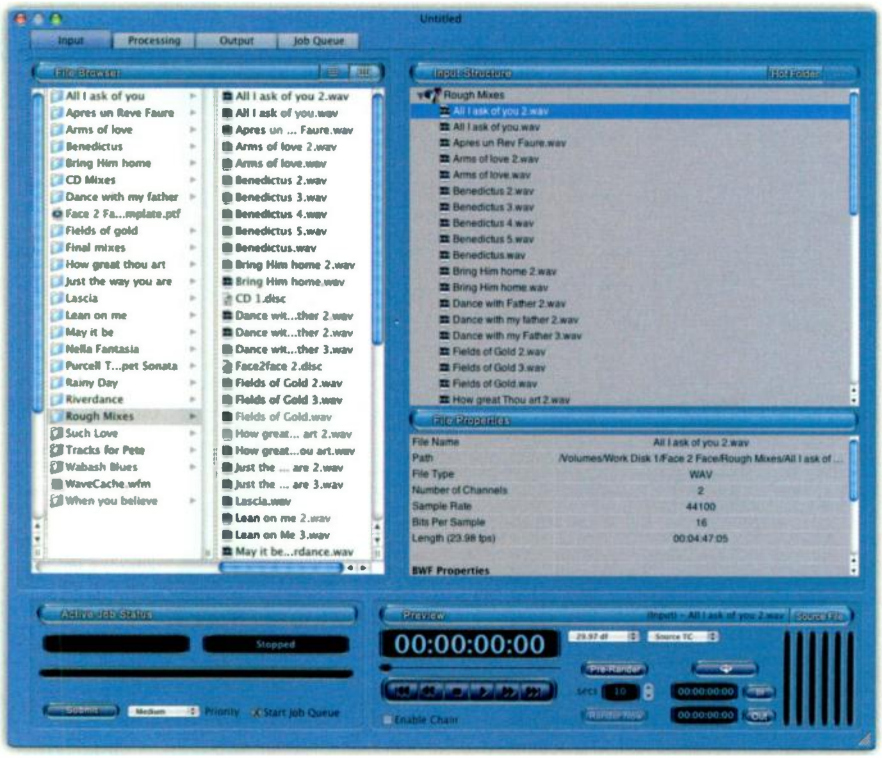
With EuCon itself being recognised by the AES as a control protocol, I suspect we'll be hearing more of it, and indeed as we go to press, rumours abound of EuCon support appearing in virtually every major DAW over the course of the next year. For those lacking enough children to sell into slavery, all is not lost, as Euphonix are also launching two much more affordable project studio controllers derived from the same technology, the MC Mix and MC Control, which will compete directly with products like Mackie Control — see last month's news pages for details. So is the future EuCon-flavoured? I would very much say so. SOS

information

- **£** £10,281.25 including VAT.
- **T** Euphonix +44 (0)20 8561 2566.
- **W** www.euphonix.com

Need to convert files *en masse* into different formats or sample rates, or apply the same plug-in settings to hundreds of files at once? Minnetonka have developed a utility that could take the legwork out of the job.

Mike Thornton



Minnetonka AudioTools AWE

Batch Processing Utility For Mac OS X

AudioTools Audio Workflow Engine (AWE) from Minnetonka Software is described as a batch conversion tool for audio files. However, it does a whole load more than simply converting files from one audio format into another: AWE can do editing, converting, transcoding, encoding, and applying plug-ins. Users specify a set of input files, configure a chain of processors, set parameters for each processor and run the job. All files are automatically processed and placed in the specified output location. Minnetonka claim that "AWE is the industry's first fully automated processing tool for audio assets", and that it has been rigorously tested by "the industry's best and brightest". It's currently Mac-only, but a Windows version is expected sometime in the middle of this year.

Four Steps To Heaven

The four basic steps outlined above each have their own tabs within the AWE interface. Step one is to select the sound files you want to process. You can select sound files by double-clicking within, or dragging them out of, the browser on the left of the window; alternatively, drag them from Mac OS X Finder windows or the desktop. It's easy to audition files using the Transport function in the bottom right-hand corner of the window. I suspect that this application has been written in Java, but it connects to Mac OS at Unix level, so to find connected drives you have to go to the bottom of the browser list and open the Volumes folder; it also displays all the files and folders that Mac OS keeps hidden away. I changed one of my drives while AWE was open and I found that the file browser didn't automatically update. It was necessary to go into the View menu and select the Refresh option to get the file browser to acknowledge the changes.

The Processing tab is where you select the operations that will be performed on the sound files. To do this, you select the desired processors from the list and drag them to the Processing Chain. The interface for each processor will appear when the

Alternatives

The obvious alternative to AudioTools AWE is BarbaBatch from Audio Ease (www.audioease.com/Pages/BarbaBatch4/BarbaBatch4.html), which is similarly priced, and likewise available only for Mac OS. However, the two applications do differ: BarbaBatch supports a much wider range of sample rates and file formats, including lots of obscure telephony formats, but doesn't allow you to apply plug-ins or surround encoders to the audio.

SOUND ON SOUND

Minnetonka AudioTools AWE
£239

pros
- Good support for third-party plug-ins.
- Powerful file re-naming options.
- Excellent sample-rate conversion.

cons
- Incomplete support for interleaving and un-interleaving multi-channel files.
- Limited file-format support.
- Manual not keeping up with developments.

summary
AWE is a powerful batch conversion package with a full range of features, let down a little by the limited range of supported file formats and the handling of multi-channel files.

See It In Action

If AWE sounds as though it might be useful to you, check out the excellent YouTube video showing the basics of how it works: www.youtube.com/watch?v=Fanz77XXKYI.

processor is selected, so you can adjust the processor parameters. I found that some, like Izotope's Ozone 3, didn't fit in the window, so I couldn't get to the section selector buttons, but when I dragged it slightly, AWE turned it into a floating window and I could then get at all the controls in the Ozone 3 window.

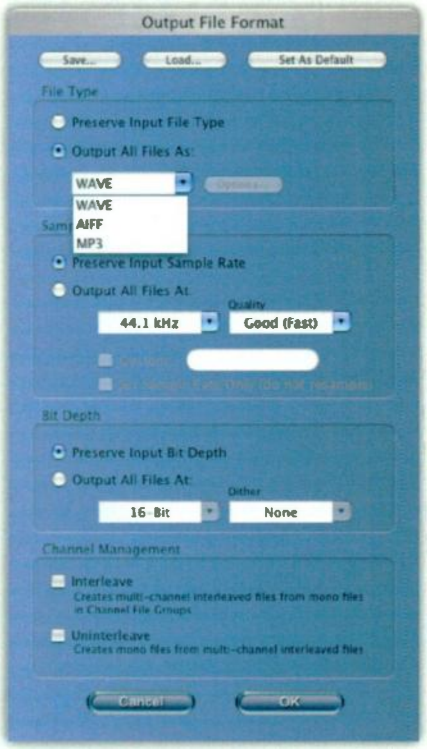
AWE supports its own standard set of processors, any third-party VST plug-ins that are installed on your system (though not Audio Units plug-ins), and optional Minnetonka processors such as their Dolby Digital Processor and Master Bundle. The latter consists of a number of Izotope plug-ins supplied under licence.

If you click on the Output File Format button, another window opens, from which you can set the output file format. I was originally supplied with version 1.2, and was disappointed to find that there were only two output file formats available: WAV and AIFF. I would have expected at least MP3, as well as a range of other compressed formats. Since then, Minnetonka have released v1.3, which does include MP3 export options, along with the much-requested support for Sound Designer II files. However, although sample rates from 8kHz to 192kHz are supported, there are still no 12- or 8-bit output options. The lowest bit depth on offer is 16-bit, an omission which means that AWE may not be a complete 'one-stop shop' for batch file processing for some applications.

Location, Location, Location

The next step is to specify the output filenames and location. AWE can put all the output files into a single folder, or replicate the folder structure in which the original files were contained. The File Name Modifiers section is surprisingly powerful, allowing you to include date, time, job name and the text of your choice. You can also use search and replace to modify any text in the original filenames. The results are previewed in the Output Structure section of the window. The only modification I would make here is that I would prefer a cursor to appear when clicking in one of the text boxes, ▶

VST plug-ins can be used within AWE to process batches of files.



Compared to rival products, AWE supports a rather limited range of file types and bit depths, but sports good-quality sample-rate conversion.

to let the user know that AWE is ready to accept text input.

From this tab, you hit the Submit button. You are asked to give the job a name, and AWE then saves the details of the job in the chosen location and puts the job into the Queue to begin processing. The Job Progress and Job Status indicators keep you informed of the progress of your job as it is being processed. You can queue up as many jobs as you want, and AWE will process them in order, but you need to remember to hit New Job from the File menu. There are Job Queue control buttons to the left, and you can also review the Log for a job to verify that it has completed and processes all files without error.

In Use

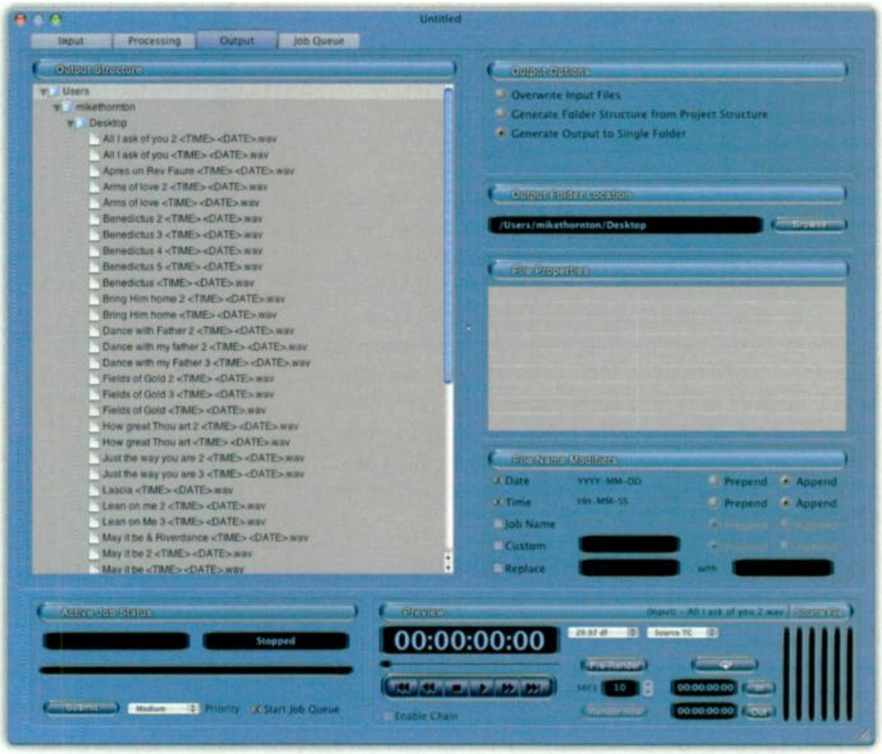
I tried using AudioTools AWE on four different jobs. The first was to process a range of backing tracks using Ozone 3, before adding fades in and out. This AWE did well, although it seemed to be slower than I would have expected.

Next, I tried to create an interleaved surround-format file from a set of individual mono files with suffixes such as '.L', '.R' and so on. It took me a little while to work out how to do this. There is an Auto-Load option in the Preferences, which is designed to load the rest of the multi-channel files when you drop an '.L' file onto the Input Structure, but that on its own didn't seem to do it. It was only after I'd studied the full manual further that I discovered the option to create a New Channel Group. This puts up placeholders in the Input Structure window, and then when I dragged the '.L' file, sure enough, it sensed and loaded the others.

There is an Uninterleave feature which does the reverse, but for some reason it does not break the mono files out with their appropriate letter extensions ('.L', '.R', '.C' and so on), using numbers instead. That isn't insoluble for the odd file here and there, but to have to go through and re-label the file names would be a major pain on a large project.

Hot Folders

An AWE job can be configured to process audio files as they appear in a Hot Folder, rather than the user having to add files to the job manually. To run a Hot Folder job, you configure the AWE processing chain and any output options as usual, and on the Input tab you select the Hot Folder button in the right-hand corner. This button enables you to select a folder AWE is to use as the Hot Folder. Once the job is submitted, AWE will process any file that appears in the Hot Folder. If no new files exist, AWE will wait until one shows up.



You can choose whether all your processed files should end up in the same folder, or whether you'd rather preserve their original folder structure, and the File Name Modifiers section provides extensive control over naming.

Next I tried the MP3 export. This worked as expected, with an excellent range of options, but again seemed slower to process than I would have expected.

Finally, I tried the sample-rate conversion, using some solo piano recordings which would hopefully show up any weaknesses, and was very impressed with the results. The sales literature refers to AWE's "high-resolution sample-rate conversion", and my tests would bear this claim out, even on the lowest-quality 'Faster' setting.

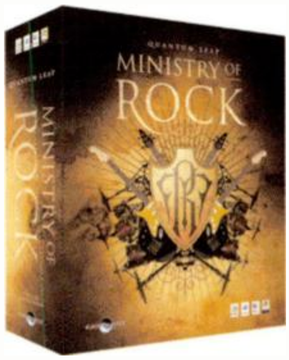
Overall, this is a versatile batch-conversion application that leaves very few stones unturned. The latest version 1.3 adds very welcome support for creating MP3 files, although the manual does not yet cover this feature, so you are left to work out the correct settings to use. The addition of SDII support is welcome, too, but competitors such as Audio Ease's BarbaBatch support a much wider range of file formats, so the lack of this sort of broad format support dents Minnetonka's claim to have created a universal solution. It is also unfortunate that the manual hasn't kept pace with the developments.

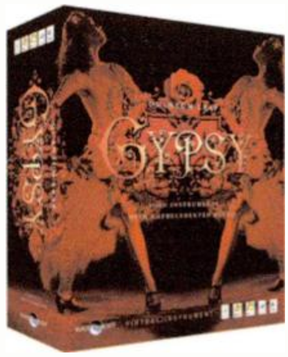
Other than that, though, AudioTools AWE is only let down by some issues to do with multi-channel support. With the ability to use third-party plug-ins and the comprehensive re-naming structure, it's an excellent batch-processing tool for most purposes. SOS

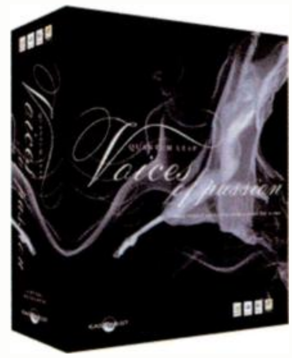
information

- **£** £239; Master Bundle £179. Prices include VAT.
- **T** Et Cetera +44 (0)1706 285650.
- **T** Minnetonka +49 2162 106 2622.
- **W** www.etcetera.co.uk
- **W** www.minnetonkaaudio.com

EastWest 'Play' Instruments



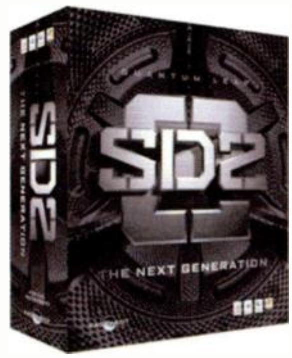




EastWest are a formidable force in sample and software instrument development, with products ranging from hip-hop loops and Latin rhythm sound sets to electronica construction kits and orchestral libraries.

This year marks EastWest's 20th anniversary, having been founded by producer Doug Rogers in March 1988. Back then, EastWest embraced the computer-based musician at an early stage, launching their on-line sales and distribution division Sounds Online (www.soundsonline.com) in the mid '90s. In 2004, they purchased Quantum Leap, a production and sample library-creation enterprise that was founded by Nick Phoenix. Together, the pair continued to develop sample libraries and virtual instruments to suit all genres and formats.

Total prize value £1350



In 2006, EastWest bought Cello studios (formerly the Western part of the United Western Recorders complex on Sunset Boulevard, Hollywood), and renamed it EastWest studios. The complex is now EastWest's US headquarters and, following a redesign of the 'non-technical' aspects of the facility, will be in constant use for EastWest productions.

Soon after the acquisition of Cello, the company announced their new range of sample-based instrument libraries, which are based on Play, their 'advanced sample engine'. Play offers support for 64-bit operating systems, therefore allowing the software to address more RAM than 32-bit systems and process audio with greater internal headroom. The engine can run on a network of machines, meaning that multiple computers can be used to host the samples and, as each instrument is built from the ground up, unique controls and effects are tailored to the particular application. This means that, whether you're playing a viola patch or an accordion preset, you know the software will be optimised to make the most of the sounds.

This month, we've teamed up with EastWest to bring a swathe of great prizes to two lucky winners. Up for grabs are all six titles currently in the Play range, two of which are reviewed in this month's issue of *SOS* (turn to page 72). Fab Four is a collection of Beatles-inspired samples; Gypsy has a multitude of unusual classical instruments to choose from; Voices of Passion is a highly acclaimed vocal sample instrument; Storm Drum 2 comprises a wealth of acoustic drums and percussion; Ministry of Rock contains drums, bass and guitar sounds; and Pianos has a collection of some of the finest piano sounds. In short, winners won't lacking inspiration! The winner in first place will be able to choose four of the prizes from the list, while the remaining two prizes will be awarded to the person in second place.

If you would like to be in with a chance of winning these great 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 *SOS* web site. Please make sure you answer all the questions and complete the tie-breaker. We also require your full address, including postcode, and your daytime telephone number. The closing date for entries is **March 31st, 2008.**

Thanks to EastWest for supplying these fabulous prizes.
T +31 20 404 1687.
W www.soundsonlineeurope.com

questions

In what year were EastWest founded?
a. 1999 ☐
b. 2008 ☐
c. 1988 ☐
d. 1895 ☐

Which studios did EastWest purchase in 2006?
a. Viola ☐
b. Cello ☐
c. Contrabass ☐
d. Harp ☐

EastWest's Fab Four sample-based instrument is inspired by which band?
a. The Rolling Stones ☐
b. Take That ☐
c. The Beatles ☐
d. Abba ☐

EastWest tie-breaker

Many things happened in the '80s, including the birth of EastWest. What was your favourite event of that decade, and why? Answers in 30 words or fewer, please.

....

....

....

....

Name ....

Address ....

....

Daytime tel. no: ....

Email: ....

If you would like to receive more information about EastWest products, please tick or cross this box. ☐

Post your completed entry to: EastWest Competition March 2008, *Sound On Sound*, Media House, Trafalgar Way, Bar Hill, Cambridge CB23 8SQ, England.

the small print

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

individual tracks.

Most of the controls will be familiar to anyone who has used a compressor before, encompassing, as they do, Threshold, Ratio (switchable 2, 4 or 10:1), Attack and Release. The release control includes a multi-stage auto-release function designed to adapt to a

wide variety of programme material. A variable high-pass filter in series with the side-chain input allows the user to reduce the sensitivity of the compression to lower frequencies if required, to stop basses and kick drums causing pumping. One unusual addition is an automatic Fade function that allows the user to set an automatic fade-in or fade-out of up to 60 seconds in duration. The master faders in many DAWs operate pre-insert, so this function allows you to create a post-compression fade that might not otherwise be possible.

This plug-in turned out to be easy to use, and it works particularly well on rock music tracks for adding punch and blending the sounds together. The Mix function is particularly useful in this respect, as it allows low-level signals to be kept up at a sensible level without robbing the louder parts of their dynamics. There is a definite tonal similarity between this plug-in and the bus compressor found in SSL's Duende, but the UAD plug-in has rather more control flexibility.

Maximum Precision

UA's Precision Maximizer is a dynamic processor designed to increase the apparent loudness of a source but without increasing its peak level or killing the impression of dynamic range, as simpler limiting algorithms tend to do. It can operate as a single-band or three-band processor and includes an element of harmonic manipulation as well as limiting. This seemingly applies a combination of tube saturation emulation and techniques used in other UA mastering processes. The aim is to increase subjective loudness while minimising the side-effects of the processing, but at the same time giving the final mix more punch and clarity, which is exactly what it does in practice. A Shape control appears to alter the non-linearity of the compression and soft saturation, and a Mix parameter allows the processed signal to be mixed with the dry input, creating parallel compression.

This plug-in can bring about a significant increase in apparent loudness and density without smothering transient detail, so should prove popular even though what really goes on under the hood is a bit of a mystery!

UA's Precision De-Esser tackles the age-old problem of sibilance. A Threshold knob controls the amount of sibilance reduction, there are two switches governing the envelope attack and release rate of the detector, and a Frequency knob controls the range of frequencies reaching

the detector. This filter can be switched between band-pass and high-pass response and has a wide range — 2 to 16 kHz — which makes it useful not only for tackling vocal sibilance but also the frequencies of over-splashy cymbals or hi-hats. In band-pass mode the width can be varied from just two semitones to 20 semitones. Split mode enables gain reduction to be applied only to the high part of the audio spectrum, but it can also operate as a full-range processor for more traditional de-essing, where the whole signal level dips when a sibilant sound is detected. Personally, I've never liked full-range de-essers, as they tend to make vocals sound 'lispy', so Split mode is the one I'd use for normal vocal treatment. Again, this is a simple and very effective plug-in.

Designer Slopes

Among the well-regarded hardware gear UA have licensed for plug-in emulation is SPL's Transient Designer. I've long been a fan of the original, as it allows the user to modify the attack and release characteristics of a percussive sound without having to worry about threshold levels, as you would with a conventional compressor. This is achieved using something SPL call Differential Envelope Technology, where just two controls allow transients to be sharpened or slowed down, and sustain lengthened or shortened. Attack transients can be amplified or attenuated by up to 15dB, while Sustain can be amplified or attenuated by up to 24dB. Other than that there's only an output gain control, a bypass control and a stereo link switch.

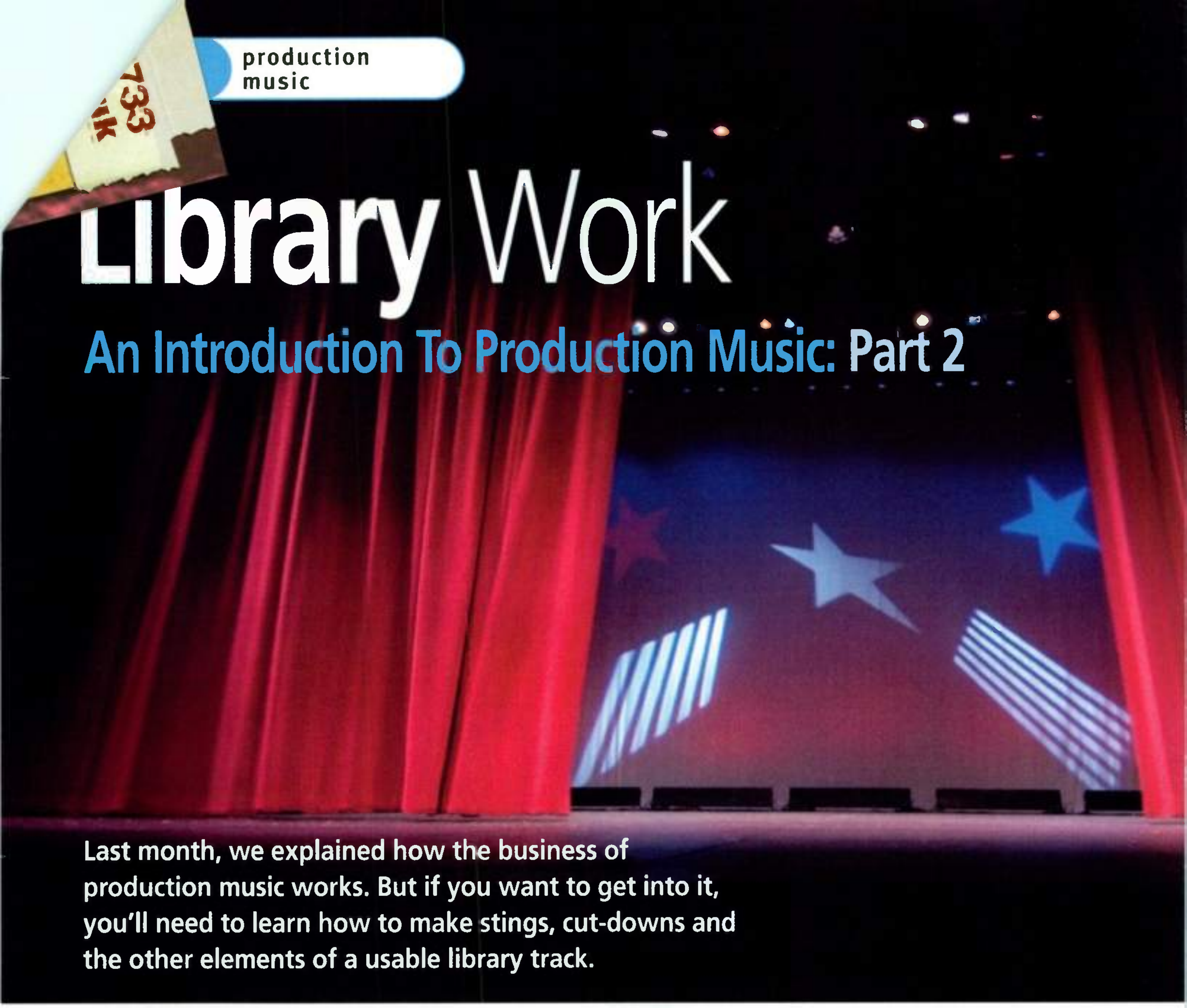
Using the SPL Transient Designer plug-in is simplicity itself, as you simply turn the two knobs until you hear the attack and release character you like. The Attack control can modify a bass-drum sound to give it lots of clicky attack or a more rounded, electronic sound, while lengthening the sustain brings up the natural decay of the drum and also emphasises any room reverb. The plug-in works well both on individual drum tracks and drum mixes, and has applications on other musical sounds, such as acoustic guitar. It's also a great fix-up tool for drying up sounds with too much spill or reverb, where it is far more subtle than conventional gates or expanders. I've been wanting this great tool in a plug-in format for ages, so now I'm very happy, as it performs just as the hardware does.

Lovely Stuff

Universal Audio's plug-in designers just go from strength to strength, and the latest crop are uniformly impressive. Some of the processes offered here can be handled more affordably by competing products without the user hearing a radically different outcome, but the Precision Maximiser is rather special and really works well. I also liked the simplicity and the sonic effect of the LA3A, and it's great to have an authentic emulation of the Transient Designer. And, of course, many laptop users will be over the moon to learn that they can finally join in the fun! SOS

information

- £ Xpander Xpress £779; Xpander Xpert £1099; Xpander Xtreme £1699. Prices include VAT.
- £ Precision De-Esser $99; LA3A $149; SPL Transient Designer, Precision Maximizer and Buss Compressor $199 each; Neve 88RS $299.
- T Source Distribution +44 (0)208 962 5080.
- E sales@sourcedistribution.co.uk
- W www.sourcedistribution.co.uk
- W www.uaudio.com



Library Work

An Introduction To Production Music: Part 2

Last month, we explained how the business of production music works. But if you want to get into it, you'll need to learn how to make stings, cut-downs and the other elements of a usable library track.

Pete Thomas

OK, so you have been briefed to write some production music, or you want to write something as a pitch to a production music company. "What makes good production music, then?", you ask yourself. The answer is almost anything, as long as it's good. As I said last month, in addition to a good tune and very high production values, a spark of originality will often help.

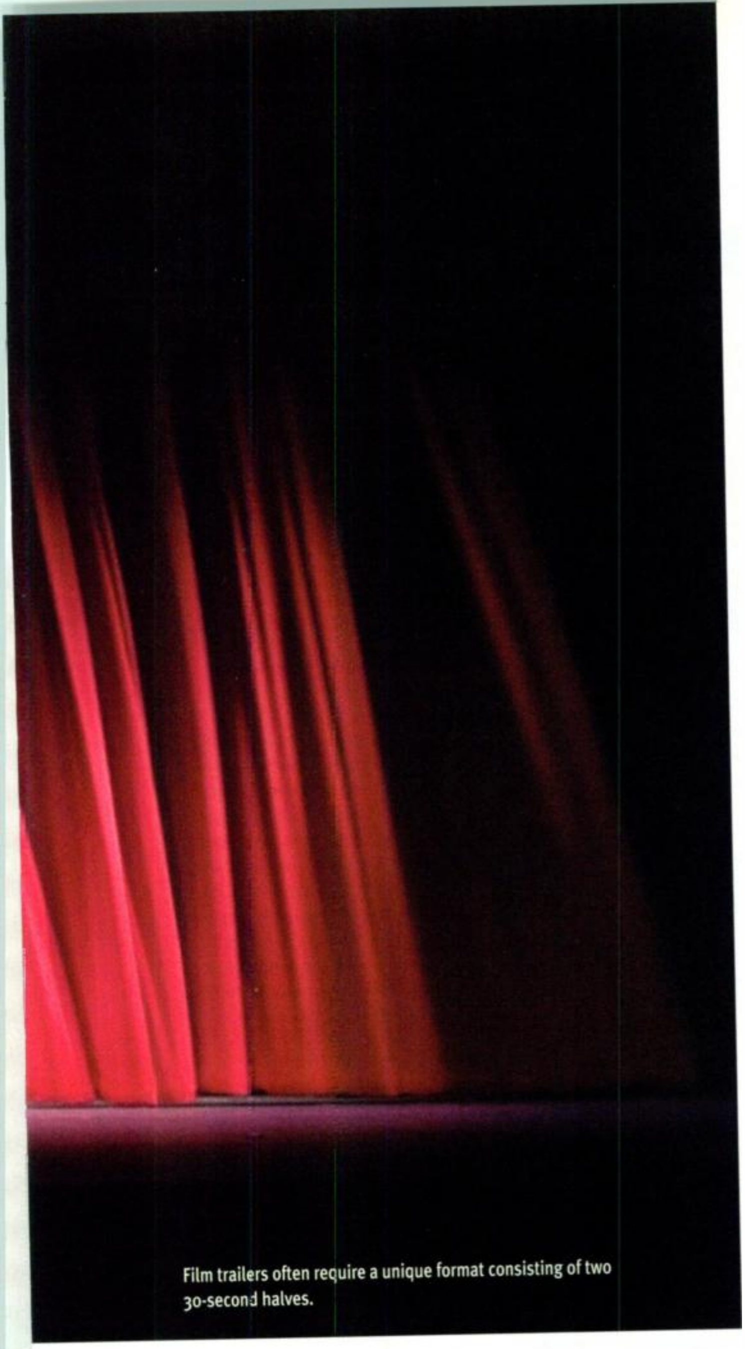
Usually I start out to write a track as if it is destined for a top-selling CD. At this stage I don't worry about specific timings and usually write something about two and a half to three and a half minutes long. I often use a standard form, which introduces a theme, follows it up with variations or solos, then restates the theme, sometimes with a breakdown either after the first theme or before the last.

Last month, we saw that it's crucial to be able to offer versions of your tracks that clock in at exactly 10, 20 and 30 seconds, and most of the technical issues that are specific to production music stem from this requirement. However, at the composition stage, I won't worry about making the tempo suit specific length versions. I'd rather concentrate hard on making the music have the best feel. I have found that if you set a tempo purely to make the 30-second or 20-second version easy to edit, you can end up fighting to get the right groove. The mood I'm in when I first compose a piece plays a large role in determining the tempo, and I find this emotional rather than purely technical connection with the music can really pay off in the long run.

I like to think of myself as owning several hats that I can put on as required; in fact, this approach is good for any kind of music production. So sometimes I have on my trained musician/composer/arranger hat and will be concerned with performance technique, intonation, orchestration. Before too long, however, I will put on my 'bloke in the street' hat and ignore all the rules about music I was taught at college, or all the production techniques I've gleaned from books (yes, even Paul White's Recording And Production Techniques, which is never too far from reach). This hat comes in very handy for indie, pop or folk but it also can work well for jazzy stuff, especially, which benefits from unexpected or out-of-the-mainstream elements. I have arranged a few covers of classical tunes, and once again this approach can result in something a bit different to the vast stock of classical music that already exists in production music libraries. (There are some example tracks on my web site at www.petethomas.co.uk/production-music — the one called 'Nutty Crackers' is a good illustration of this.)

Some Musical & Technical Considerations

It can be very useful if you write into the music plenty of good edit points for the film editor. These can include obvious hits, drum breaks and sudden silences (listen to the audio example called 'Boomtown', which has drum breaks every eight or even four bars).



Film trailers often require a unique format consisting of two 30-second halves.

Key changes and tempo changes are usually a bad idea, as these can make it very difficult to create sensible edits.

As well as full-length versions and various edited cut-downs, there is another format geared specifically for film trailers. This requires a one-minute total length, but with a very definite edit at 30 seconds — in other words, a false ending — so that just the first half can be used. With this format the first 10 seconds can form an intro, with the main theme stated for the next 20 seconds. The second half should be more exciting, building up to the end: think of the typical car chase or rescue.

In addition, I always supply a 'backing track' mix without any lead instruments or vocals. This is useful for programme makers to use as an underscore.

Use of effects on production music need not be different to any other kind of music. I might just have a touch more reverb on production music than I would for a CD, but only if it's a style of music that can cope with it. If you are aiming for something big and brash that might get picked up for a game show or comedy theme it might be appropriate, but I'm always aware of how too much reverb can be disastrous and make the track end up sounding like a bad '80s demo recording. So, if in doubt, just assume you want the same style and quality of sound as if you were making an album track.

If I know the mixes will be mastered, I will probably not put any compression on the final mix, apart from maybe a very gentle touch of multi-band

feature production music

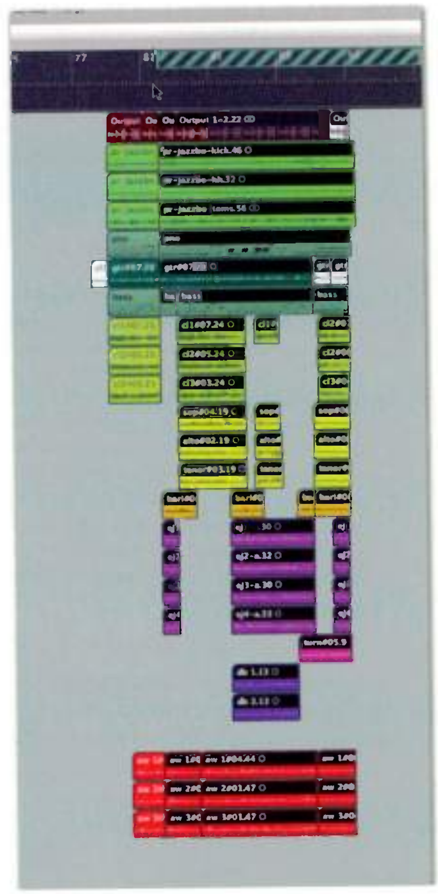
one that will make an easy 30- or 20-second version. For me, this means the music can really live and breathe as a track. The tricky bit is then editing it down to the right length!

When I first started production composition I created a new Logic project for each shorter version. Wrong! If you subsequently need to do a remix or restructure, you have to open each version and apply the changes. So I developed an alternative method that works very well in Logic, and possibly better in other sequencers. As described below, I'm assuming that you are creating your cut-down by removing just one chunk from the middle, but it can easily be adapted to taking out two or more chunks. Note that it can be slightly more complex if tempo changes are involved — I always try to avoid them in production music, as the end user may need to do further edits and tempo changes will make this awkward.

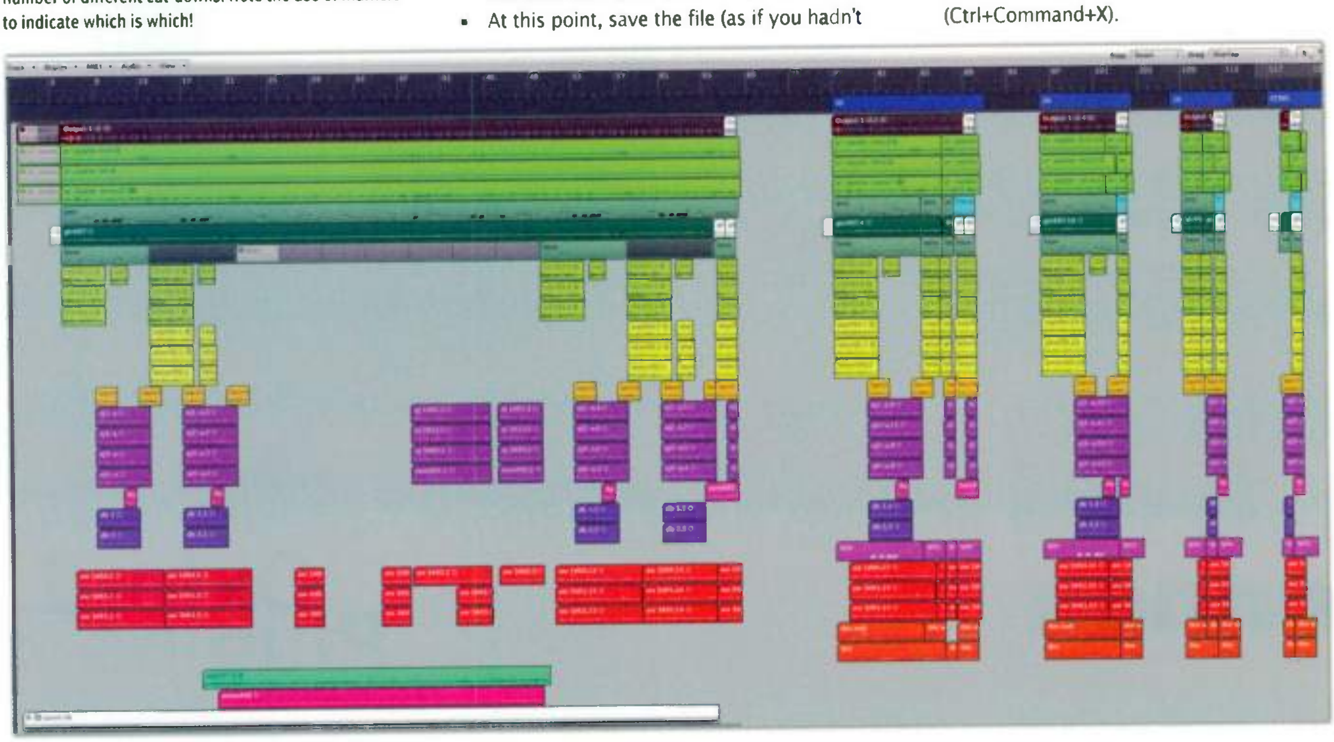
- Create your basic arrangement. This should be the longest version — it's easier to edit stuff out than edit it in.
- Select everything and copy to later in the arrange page, about eight bars after the original end. This is your first cut-down.
- To check your timing, you need to use a SMPTE offset to make the first note or sound of this new version start at 01:00:00:00. To do this in Logic, visit File Menu / Project Settings / Synchronisation, and set the bar to which you have copied the regions in the Bar Position box. You now need to see the SMPTE offset. In Logic 8, Ctrl-click on the transport display and choose 'Use SMPTE view Offset' from the shortcut menu; in pre-8 versions of Logic, you can access the SMPTE offset from the drop-down menu when you click on the small triangle at the right-hand end of the floating transport window. Now for a bit of good old trial and error...

With applications such as Logic, you can define a SMPTE offset such that your cut-down version starts at 'zero'.

- Move the playhead from the start, while watching the transport display, to work out approximately how long your shorter version needs to be. Allow a couple of bars for the final note and reverb tail.
- Work out how many bars you need to subtract from the full version and define a skip cycle of that length in the bar ruler.
- Move the skip cycle so that you get at least your opening theme before the skip area. The music before and after will be what's left after the edit — you should be able to visualise roughly what you want to keep from the beginning and end of the arrangement. Listening with the skip cycle active tells you how the edit might work and you can move it around for the best results. Don't worry too much about audible glitches — you can smooth those out later with crossfades or MIDI edits.
- At this point, save the file (as if you hadn't been doing that every five seconds).
- Select everything again. In Logic, Cut Time & Move By Locators from the Region menu (Ctrl+Command+X).



Once you're happy with the edit sketched out using the skip cycle, you can Cut Time to remove the skipped material and create a shorter arrangement.



For most production music projects, you'll need a number of different cut-downs. Note the use of Markers to indicate which is which!

At this stage you have a very approximate cut-down version. It might be just right, but the chances are you will need to make some changes. Play it from beginning to end and note how close you are to the target timing. I like to use Logic's Giant SMPTE Display option (again, found in the shortcut menu in the transport window). Unless your edit is exactly right — which is unlikely — use Undo to get back to the uncut version, and attempt some of the following adjustments:

- If necessary, add or subtract a bar from the skip cycle to get closer to the target length. Depending on the content, it may be possible to use fractions of bars, but at this stage, aim only for accuracy to within a second or two.
- Moving the cycle zone to different parts of the music, keep trying the cut until you get the best result.
- Tidy the cut by checking individual tracks: you may need to adjust the start and end of regions if notes have been cut into.
- If you need the length to be absolutely accurate, make further adjustments, such as adding or losing a 'pickup' drum beat, roll or sound effect before beat 1, lengthening or shortening the final chord or reverb tail.
- When you're happy with the 30-second version, repeat the whole procedure for a shorter cut-down, either by copying and pasting this cut-down as a starting point, or using the original full-length version again. And bear in mind the following tips:

- Quantise MIDI before doing the cuts, then unquantise and merge sequences. If you don't do this, you may find odd things happening with MIDI notes that are very close to the point at which you cut.
- If you have used a MIDI sustain pedal, convert sustains to actual note lengths, or your edits may cause unwanted sustains which can be very confusing to rectify.
- Try to avoid tempo changes, but if you absolutely must, remember to copy these as well as the regions.
- Use markers to show which version is which, especially if you do alternative versions.

The most important thing is to plan the process from the start — for instance, by making sure the important thematic material is stated at the beginning and the end of the full version.

Always A Sting In The Tail

As well as the shorter versions, you need to supply a 'sting', which is a very short fragment of the tune. Often, the final few notes are the most suitable, but it can also be the opening phrase, or any memorable short hook that serves to identify the tune. Stings can be very powerful. On TV you generally hear them just before and after a commercial break, or during the last few seconds of a commercial (the 'pack shot'). A good sting will lodge itself in the viewer's mind and should summon up the entire theme. It will act as a musical logo and will be identified by the public along with the title graphics of a programme or brand logo of a product on an advertisement.

A strong ending is also important. After years of pop production I had got used to the easy way out and relied heavily on fades to end a tune. With production music, you can't just end on a repeat chorus and fade it out. You can't just end on a repeat chorus and fade it out. You can't just end on a repeat chorus and fade it out...

You must have a good ending. SOS

EVENTIDE TIME FACTOR

a USB port, which allows you to load updates from Eventide's web site via Mac or PC, and a power inlet for the included 9V DC PSU.

Most of the rotary control and switch parameters are pretty intuitive, but it is worth noting that they control different parameters for each delay algorithm. The Depth and Speed controls, for example, control the Wow and Flutter parameters for the Tape Echo model, but for Mod Delay are used to edit the modulation depth and rate. The Xnob is used to change the setting of special parameters for each model, such as crossfade for Digital Delay, bit depth for Vintage Delay, and the tape hiss level for the Tape Echo model.

Three bypass modes are available: true analogue bypass, which is great for simple guitar-to-amp configurations, plus a DSP bypass which routes the signal through the processor but applies no effects, and a third option called DSP+DLY, which applies no new effects to the input, but doesn't cut off any ongoing effects tail when engaged.

Modes

There are two modes, Play and Bank, and you switch between them by holding down the right-hand footswitch ('Tap'). In Play mode, the current effect name is displayed on the screen, and the footswitches are used to control the effect. The left switch acts as a bypass, the centre one switched the infinite repeat function on and off, and that on the right controls tap tempo, or selects tempo values if the Tempo function isn't engaged.

The Bank mode is used to save and access user patches, which are stored as 'banks' and 'presets'. There are 20 presets, organised in banks of two, and the current bank and preset numbers are displayed on the screen. Saving presets is easy: you hold down the rotary encoder until the bank and preset numbers are displayed, turn it to select the location, and press again to save. If 20 presets are limiting, you can back things up by MIDI SysEx, and if you don't want to scroll through all of them, you can limit the number that are available — very useful in a gigging situation.

Loop The Loop

There are plenty of loopers around now (see the 'Alternatives' box below), and I found the Eventide among the easier to use. You get up to a 12 seconds of loop memory in mono. Everything is done with the three buttons, with their functions indicated by small transport icons: round for record, triangle for play and square for stop. Depending what you're doing at at the time, these buttons do different things. For example, pressing the record button while already recording ends the first loop and starts overdubbing another. You also get control over the wet (loop playback) and dry signals. Again, this is all very clearly described in the manual, and in practice it is really easy to use, too.

In Use

I used the Time Factor on a guitar tracking session, substituting it for my Boss DD6, and was instantly impressed. In Play mode (the default mode) the basic controls were all very intuitive, and I was quickly able to dial up the basic delay settings that each track required. The rhythm section had been recorded without a click, and being able to dedicate a switch to tap tempo while leaving others to operate normally was refreshing (the lack of this facility on my other delay pedals has been a consistently source of frustration). Running straight into my amp, it gave out a healthy level and the sound of the digital delay was nice and clean. The same couldn't be said of the Vintage Delay, of course, but I mean that in a good way! I was able to get a very convincing tape delay sound, complete with the effect of speeding up and slowing down the tape where I wanted it.

Though for this session I stuck largely to the digital delay and the tape delay, I had opportunity to experiment with the various different models on guitar and other instruments. The modulation delay and reverse delay in particular appealed to me — particularly for live use, where the expression pedal input comes into its own — but I couldn't find fault with any of the others.

I also tried tracking direct to the desk when recording a synth part, switching the output level accordingly, and again it worked very well. I wasn't using an amp here, but found that I got better results using a dedicated amp and speaker emulation — and it would have been nice to see at least a basic one included in this design.

Back in the studio after the recording sessions I tried the pedal out on a mix as an external send effect in Cubase, where it seemed to perform very well. The ducked delay was a particularly nice effect on both guitar and vocals. The switchable input and output levels mean that it is much better suited than your average guitar pedal for studio work. It was also very refreshing to have so many knobs and dials to twiddle on a studio device — it is the kind of marriage of live versatility and studio quality that you dream of. One thing I've not mentioned is that you can use this pedal as a stereo, mono, or dual-mono effect. You're not able to load two different effects models simultaneously, but you can still run two separate delay lines with different delay and feedback settings, which I found really useful when mixing.

The Time Factor's inputs and outputs are switchable between line and instrument level, and the aux switch and expression pedal inputs add to the versailtity.

Max Factor?

It sounds impressive, but should you buy it? If all you need is a guitar delay pedal or a looper, there are cheaper options. Also, for studio use, the lack of digital I/O may be a limitation for some people. The expression pedal and aux switch inputs, though, and the MIDI connectivity are a Godsend. There's enough hidden depth here to keep you exploring for some time, but you can also get good results very quickly. I'm inclined to agree with the boast on the box that "nothing sounds like an Eventide": the Time Factor is a sophisticated creature with a distinctive personality, and nothing else I know does quite what this pedal does quite as well. It is among the best delay pedals I've used, and is capable of competing with more expensive rackmount effects. I've hung on to this pedal longer than I should have: I'll be sad to see it go and can't wait to try the Mod Factor! SOS

Alternatives

There are plenty of stereo delay pedal options out there at the moment that cover similar territory when it comes to delay effects — the Boss DD6, for example — and there are now plenty of very capable looping pedals out there, such as the Boss RC20. In fact, for the price of the Eventide you could get both these Boss pedals, though you'd still lack much of the Time Factor's functionality, including the expression pedal control. Line 6's DL4 is probably the closest competition, though it is less versatile, particularly in terms of hands-on control.

information

- £ £299 including VAT.
- T Source Distribution +44 (0)20 8962 5080.
- E sales@sourcedistribution.co.uk
- W www.sourcedistribution.co.uk
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URS Classic Console Strip Pro

Channel Strip Plug-in For Mac OS & Windows

Unique Recording Software have modelled a vast array of vintage compressors and EQs and presented the results in a single plug-in.

Sam Inglis

Stand-alone hardware channel strips became big news in the '90s as engineers began to migrate to computer-based systems for recording and mixing. Ditching the big desks and racks of outboard effects in favour of PCs and plug-ins brought undeniable benefits: automation got more sophisticated, track counts grew, space was saved. Yet many engineers felt that something else went by the wayside in the process. Digital audio workstations might have allowed them to use hundreds of EQs and compressors on every track, but those EQs and compressors just didn't sound as good as their analogue counterparts.

The idea behind the channel strip was to put back that missing analogue magic, but in a format that lent itself to the new digital working practices. Thus, a typical rackmounting channel strip would include a high-quality mic preamp, compressor and equaliser, in some cases taken bodily from vintage mixer channel strips. Important signals such as lead vocals could then be processed on the way in to the computer, to give them the sheen and substance that plug-ins weren't able to supply.

Throughout this decade, plug-in manufacturers have been working hard to change the prevailing view of their products as cold, sterile or lifeless, with a good deal of success. Plug-in recreations of vintage analogue processors have become big business, and their quality has improved to the extent that companies such as Neve, API and SSL have allo wed their names to be attached to digital emulations of their classic hardware. And, perhaps inevitably, things have now come full circle, and you can buy plug-ins designed to emulate the hardware that was designed to mean you wouldn't need to use plug-ins...

Consoling Thoughts

All of which brings us to the subject of this review, URS's Classic Console Strip Pro. Like many other plug-ins, its *raison d'être* is the recreation of vintage analogue processing in a digital environment. However, Classic Console Strip Pro goes way beyond the idea of emulating any one vintage channel strip. Instead, it offers a modular design that makes it one of the most comprehensive and versatile processing plug-ins on the market. Classic Console Strip Pro is available in TDM and all major native formats on Mac and PC, and is authorised to an iLok key. Installation is straightforward, although it's slightly frustrating that you need to download and run separate installers for every version you want to install (VST, RTAS, TDM and so on).

The Classic Console Strip Pro window is divided into a number of sections. Some of these can be re-ordered using the Pre and Post switches, with the current order displayed in the Signal Flow section at the left. This also provides a handy way of bypassing individual sections, which can lighten CPU load. Presets saved at the host or plug-in level capture the entire status of the plug-in, but there are also drop-down menus in the individual sections. These allow you to change the model or algorithm that section is using, and in the case of the compressor section, bring up appropriate preset time constants and other settings for that model.

The one fixed element in the signal path is the modelled input stage, which, as you'd expect, is always the first thing your signal encounters. A drop-down menu provides access to a range of algorithms which model different combinations of transformer input stages, tape-head bumps, and tape machine or valve electronics. A gain control lets you choose how hard to drive this algorithm, and an intensity

Alternatives

Perhaps the most obvious alternative in terms of emulating a wide range of vintage compressors and equalisers would be Focusrite's Liquid Mix system, although this, of course, runs on a separate DSP unit rather than as a native or TDM plug-in. You could also investigate McDSP's extensive range of plug-ins, which likewise feature numerous presets dedicated to recreating old and new analogue gear.

slider runs from zero to 200 percent — the halfway point being the most faithful to the original hardware.

A World Of Dynamics

There are three further sections: compressor, filter and EQ. These can be arranged in any order, or the filter section can be placed in the compressor side-chain rather than in the audio path.

The filter section is the simplest, and the only one that does not provide a choice of algorithms. Instead, you get low- and high-pass filters with a fixed slope, both with cutoff fully variable from 20Hz to 20kHz. If you put it in the audio path, it does a good job of cleaning up low-end rumbles or high-frequency noise, but unlike the full-blown EQ section, its sound is transparent rather than characterful. In use, it seemed to me most valuable switched into the compressor side-chain; a gentle high-pass helps stop the compressor pumping on low-frequency sounds like kick drums, while a more aggressive setting can be used to make the compressor act as a de-esser.

The compressor section seems to be partially derived from URS's existing 1970 and 1980 Compressor plug-ins, but they have extended their remit to cover a huge range of vintage units. URS told me that they don't use convolution, and their plug-ins' relatively low DSP or CPU loads bear this out; but whatever technique they are using seems to involve creating several different algorithms for each piece of hardware, at different ratio settings. So, for instance, there are four 'Stress' algorithms (based on the Empirical Labs Distressor?), which default to ratio settings of 3:1, 6:1, 10:1 and 20:1 respectively, and other hardware units are similarly represented four or five times each.

Many of the hardware units emulated here had fixed, stepped or otherwise restricted time constants, so choosing a compression algorithm from the drop-down list also sets these to appropriate values, along with the separate Knee dial. However, the plug-in's interface is the same whatever model you select, and all the controls remain active over their full range; so even if, for instance, you select the 'Stress 3' algorithm, there's nothing to stop you moving the Ratio dial to 20:1 or even higher. Likewise, you can happily use super-fast attack times on models of legendarily slow compressors if you want to. This may displease some vintage obsessives, but as URS's Bobby Nathan points out, the 'original' feature set is not sacred; it was common for studios to modify classic hardware to maximise its flexibility, and URS are merely extending the same principle here.

One thing that initially confused me about Classic Console Strip Pro's compressor section is that calling up a new compressor algorithm also resets the Threshold and Gain Makeup controls. This means that if you use the Threshold control to manage the amount of gain reduction, it's hard to perform A/B comparisons between different compressor algorithms. However, there is logic behind this approach; each compressor preset has been designed to deliver the same amount of gain reduction, so if you want to A/B them, the easiest way is to leave these controls at the default settings and turn up the plug-in's input gain control until you get the desired amount of compression. When you then switch to a new compressor preset, the output level should remain unchanged, even if that preset uses radically different ratio and time-constant settings. The idea, apparently, is that this provides a good way of learning the differences between the different compressor models; once the user is

SOUND ON SOUND

URS Classic Console Strip Pro
£670/£335

pros

- Sounds great.
- Extremely versatile.
- Includes details such as the input stage modelling that are rarely found on competing products.
- The simpler Classic Console Strip plug-in bundled with it is very worthwhile, and efficient in terms of CPU or DSP resources.

cons

- A deep plug-in, with some interface quirks, which takes time to explore fully.
- It would be nice to have a dedicated de-esser section.

summary

If you like the idea of a vintage channel strip in software, there are few rivals that offer as much depth and versatility as this.

more familiar with them, he or she will be more confident about moving away from the presets and making free with the Threshold and other controls. It's an interesting approach, but it takes some getting used to.

By default, an appropriate input stage model is selected automatically whenever you choose a compressor model, but during the course of this review URS released an updated version 1.1 of the plug-in that makes it possible to turn this linking off. This version also introduced authentic auto-release settings for seven of the compressor models.

Strike Up The Bands

The right-hand half of the Classic Console Strip Pro window is devoted to the four-band EQ. Each band allows you to select one of five algorithms; these, again, mimic classic hardware, but have been given dates instead of names. (I'm not sure why URS have to be quite so coy about what they are emulating!) Few of the original hardware units being mimicked here had bandwidth controls, while most had only a few stepped frequency settings (and in the case of the '1951' EQ, no boost circuitry at all), but as with the compression section, the plug-in controls remain freely variable regardless. The Q setting ranges from 3.0 to 0.25, cut and boost run to 15dB in either direction, and the frequency range covers 1.5kHz to 20kHz (HF band), 220Hz to 7kHz (the two mid-frequency bands) and 20 to 500 Hz at the bottom end. The two outer bands can be switched between shelving and peaking operation.

Super Strip

I've been a long-term user of URS's Classic Console Compressors bundle, and especially their A- and N-series EQ plug-ins. In terms of sound quality, the compression and EQ sections of the Classic Console Strip Pro plug-in easily match those earlier products, as you'd expect. In terms of versatility, moreover, they go way beyond URS's previous efforts, although it's mildly annoying that the two mid EQ bands can't be set lower than 220Hz. The compressor algorithms augment the slightly generic 'vintage' feel of the Classic Console Compressors models with a huge range of distinctive and highly usable hardware emulations, including valve, optical, FET and VCA designs, as well as tape compression. I can't think of any other plug-in that models so many different input stages, although Cranesong's Phoenix and Magix's Analogue Modelling Suite offer good alternative takes on the idea. Now that a programme-dependent release option has been added, the only major functional improvement I'd like to see would be the addition of a separate, dedicated de-esser. You can turn the compressor into a competent sibilant remover easily enough by filtering the side-chain, but of course doing so means you can't also use it as a compressor. A dedicated de-essing section could also incorporate more advanced options such as split processing to compress only the offending frequencies.

From a new user's point of view, the inevitable down side to all its flexibility is that Classic Console Strip Pro is not as immediate as URS's individual plug-ins. Until you familiarise yourself with the quirks of the different compressor algorithms, the list is pretty daunting, and the unconventional design approach behind the compressor presets takes some getting used to. Likewise, the simplicity of something like the A-series EQ plug-in is a little lost when you start to introduce Q controls and freely variable frequency ranges. However, there's a good selection of presets that can serve as starting points, and I imagine most users will quickly build up a library of their own.

It's the sound that matters most, though, and if the aim is to create a truly 'one stop' processing shop, Classic Console Strip Pro certainly has the sonic bases covered. Whether you're after bright, in-your-face, screwed-down '80s rock vocals, warm '70s folksiness, characterful '60s colour, or whatever, you'll find it here. It's equally at home on other sources, such as acoustic guitar or bass, and I liked it as a mix-bus processor too; the 'Tape 15ips' preset, for instance, adds a subtle forwardness and bass boost that can really make a mix sound rounded and complete.

In short, then, there's very little to dislike about Classic Console Strip Pro, but prospective buyers might need to think about how well its 'one plug-in to rule them all' philosophy chimes with them. One of the great benefits of mixing in the box is the ease with which you can combine different processors from different manufacturers, and personally, my instinct is often to reach for simpler, single-function EQ and dynamics plug-ins that I know inside out, even though they may not offer the same depth or versatility. What URS have set against that here is the potential to create your own custom 'analogue' console by inserting the same plug-in across multiple DAW tracks, and that will be a mouth-watering prospect to many. If you like the idea of a comprehensive channel processor within a single package, and you're willing to put some time into getting to know it, you should run, not walk, to URS's web site and download the demo of Classic Console Strip Pro. SOS

Stripped-down Strip

Buying Classic Console Strip Pro gets you a free copy of the much more straightforward Classic Console Strip plug-in, which is also available as a separate product. When you haven't the time or the inclination to delve into the complexities of the full Pro version, this could be very handy. It's also designed to have a very low CPU or DSP load (not that the full version is excessive), and to map easily onto hardware controllers for hands-on adjustment.

The compressor section is based on the '1975 VCA' model from the Pro version, including an emulated transformer input stage, and offers basic Threshold, Ratio and Gain Makeup controls, plus a choice of three preset time-constant settings. It can be switched pre- or post- the three-band EQ, which features an equally simple feature set. Low and high filters have switchable turnover points, and the sweepable mid can be set to Sharp or Wide Q.

The three EQ bands are taken from different hardware models, but complement one another well, and the streamlined feature set is well suited to getting good vocal sounds in a hurry. If the full Pro version is beyond your means or seems over-complex for your purposes, it's worth looking at this plug-in (native £97.53,TDM £190.35) as a product in its own right.

information

£ TDM version £669.75; native version £334.88. Prices include VAT.
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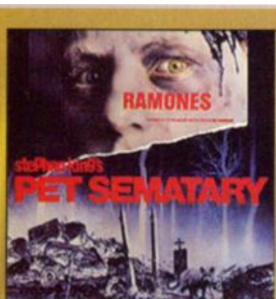
F +44 (0)1440 785845.
E sales@unityaudio.co.uk
W www.unityaudio.co.uk
W www.ursplugins.com

Classic Tracks

The Ramones 'Pet Sematary'

Undisputed kings of the three-chord thrash and arguably responsible for punk rock, it took over 10 years and the theme song to a Stephen King film to secure serious US chart success for the Ramones...



Artist: The Ramones
Track: 'Pet Sematary'
Label: Sire
Released: 1989
Producer: Jean Beauvoir
Engineer: Fernando Kral
Studios: Sigma Sound, The Hit Factory

Richard Buskin

Before them came the proto-punk New York Dolls. In their wake, on the other side of the Atlantic, the more aggressively nihilistic Sex Pistols. Yet, in America, the Ramones were the ultimate punk band, melding a requisite devil-may-care attitude and stripped-down, no-nonsense, four-chord rock & roll with catchy pop melodies and warped bubblegum-style lyrics that appeared to have taken a left turn out of the early 1960s — "DDT did a job on me/Now I'm a real sickie/Guess I'll have to break the news/That I got no mind to lose/All the girls are in love with me/I'm a teenage lobotomy." Sun and surf, girls and glue... The American Dream had been put through the blender.

Hey Ho, Let's Go!

All nice middle-class boys from Queens, New York, Jeffrey Hyman (aka drummer Joey Ramone), John Cummings (guitarist Johnny Ramone) and Douglas Colvin (singer/bass player Dee Dee Ramone) got together in 1974, with Tom Ederlyi (Tommy Ramone) as their manager, and played their first gig on March 30th of that year at New York's Performance Studios. Within a couple of months, Joey had taken over on vocals, Tommy had filled his place on drums, and by the summer the band had secured a residency at CBGB's on the Lower East Side, playing 20-minute sets of rapid-fire sub-two-minute songs. Energetic, unsophisticated and rife with tension, their shows encapsulated the burgeoning punk ethic, and in August 1975, having attracted a regular cult following, the Ramones signed a contract with Sire Records.

The next year was a breakthrough one for the band. In the spring of 1976 their eponymous debut album, recorded for just over $6000, was released to critical acclaim and just failed to crack the Top 100 on the US Billboard chart. Then an appearance at London's Roundhouse on July 4th, second-billed to the Flamin' Groovies, served to kick-start the British punk scene, before a repeat performance by both bands the following month, at the Roxy on Sunset Strip, had a similar effect in LA. A second album, *Leave Home*, was released towards the end of the year, climbed to number 48 on the British chart in the spring of 1977 and spawned the Top 40 UK hit, 'Sheena Is A Punk Rocker'. Nevertheless, even this modest success continued to elude the group in the US, and this remained true through the next two classic Ramones albums, *Rocket To Russia* (1977) and the somewhat more ambitious and textured *Road To Ruin*, which saw Marc Bell (Marky Ramone) taking over on drums for Tommy, who stayed on in a production capacity.

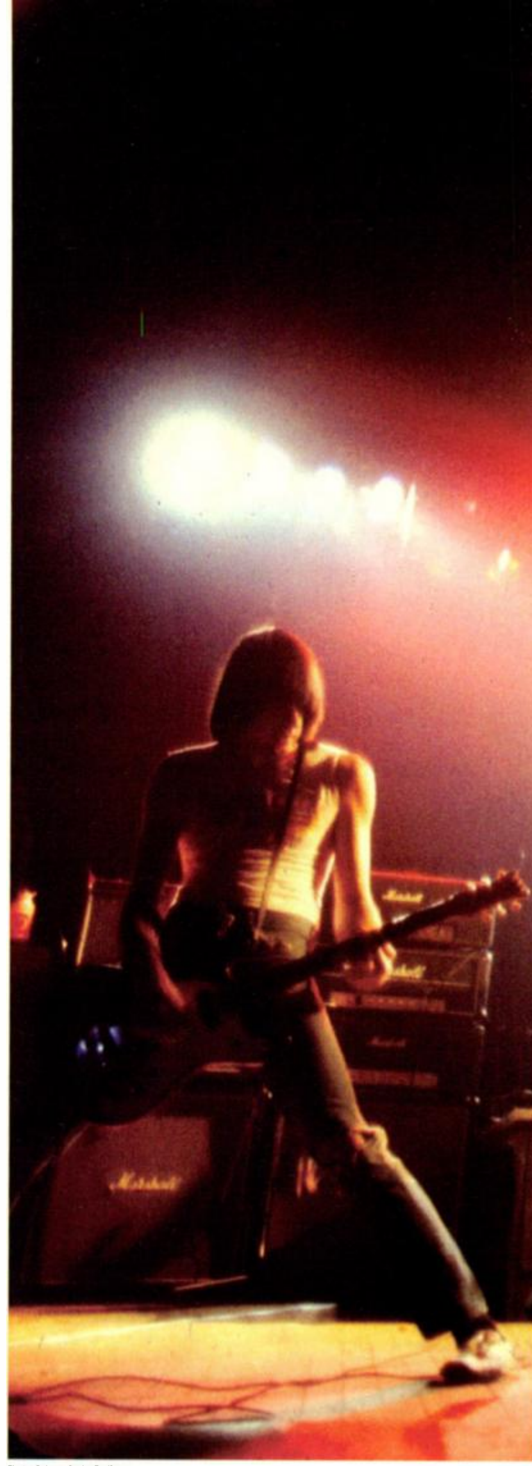
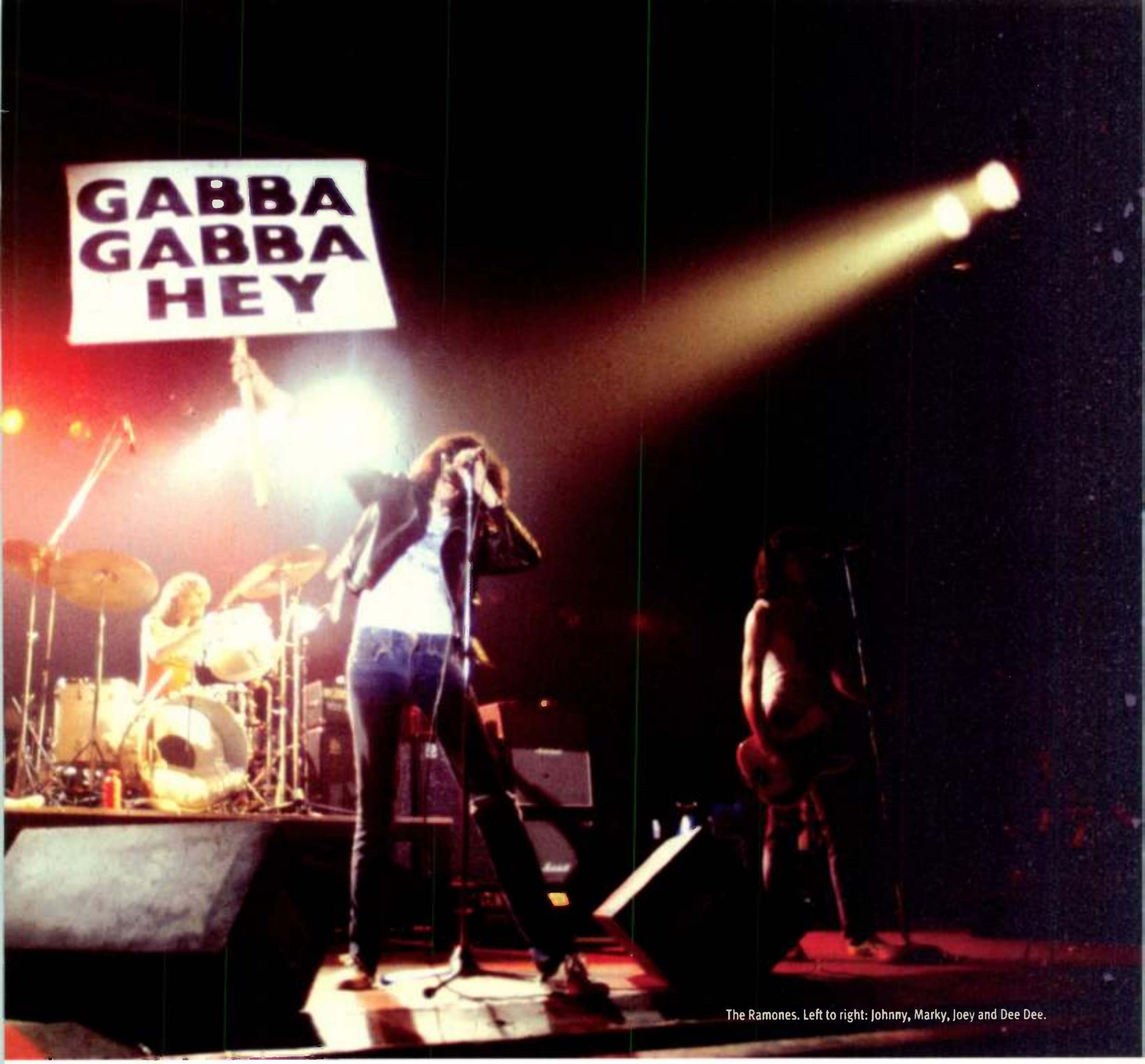


Photo: Roberta Bayley/Redferns

Punk's Not Dead

Although the 1978 demise of the Sex Pistols signalled an end to punk's halcyon period, the Ramones kept going, releasing a live double album of a 1977 performance at London's Rainbow Theatre, and also appearing in — as well as on the soundtrack album of — the Roger Corman movie *Rock & Roll High School*, while embarking on what would turn out to be a traumatic, if somewhat triumphant, collaboration with production legend, Phil Spector. At one point during the sessions, Spector indulged his now-famous penchant for guns by pointing one at Dee Dee and ordering him



The Ramones. Left to right: Johnny, Marky, Joey and Dee Dee.

to play a specific bass riff over and over again. Not exactly conducive to open-minded improvisation. Yet, the resulting album, *End Of The Century*, although moving away from the Ramones' trademark thrash-pop sound, as previously captured by engineer Ed Stasium, towards the tighter one of their producer — and who could blame them for not arguing? — did at least make an impact on the US charts, peaking at 44, while also achieving Top 10 single success in the UK with a cover of the Ronettes' 'Baby I Love You'.

That was in 1980, and the decline in critical acclaim and die-hard following that commenced with *End Of The Century* continued throughout the rest of the decade and a succession of mostly slick-sounding albums, during which Marky quit and subsequently rejoined the group. The one exception, at least in commercial terms, was the theme song to Stephen King's 1989 horror movie hit, *Pet Sematary*, wriiten by Dee Dee and sometime-Ramones producer Daniel Rey. Although dismissed by certain hardcore fans because of its relatively polished, film-oriented patina, this still stands as a bona-fide Ramones classic, one that garnered plenty of US radio play while peaking at number four on Billboard's 'Modern Rock Tracks' chart. As such, it was arguably the band's last hurrah.

"I remember Howard Stern playing it on his radio show while I was driving one morning and I heard him call it his record of the year," says Fernando Kral, who engineered the song at New York City's Sigma Sound Studios alongside producer Jean Beauvoir. "Still, I have to admit I was surprised it did as well as it did. Sure, the production was good and so was the guys' playing, but they weren't giving the fans a traditional Ramones record. A case of 'This is a risky move, let's see what happens.' The safety net was that it was for a film soundtrack, and when the film did well at the box office, the song did well, too."

Bon Jovi, Whitney Houston & The Ramones

Born in Colombia, raised in New York, high-schooled in Florida, Kral began his music business career at the age of 19 in 1981 at an eight-track studio named White Rabbit in Sausalito, California, serving as a gofer, assistant and eventually full-fledged engineer. After a couple of years, on the advice of fellow engineer Tom Lubin, Kral then opted to really learn his craft by interviewing for jobs at the large, Photo: Sigma Sound Studio

Fernando Kral at the SSL E-series desk in the control room of Sigma Sound Studios.

consistent at that time. I mean, we're talking about the Ramones."

To the left of the kit, facing the drummer, stood the DI'd bass player, although in this case it wasn't Dee Dee Ramone, who didn't actually show up for the session.

"He'd been at the rehearsal and the band had sounded tight," says Kral. "But, having written the song with Daniel Rey, he must have thought he'd done his job. So Jean played a guide bass part, which he later re-did, and we also kept that guide part. We EQ'ed it for a lot of bottom and had it in there for low-frequency stuff."

Perhaps Dee Dee already had his mind on other things. Following the summer 1989 release of the *Brain Drain* album, he'd quit the band to pursue a career as a rap artist named Dee Dee King...

"Johnny's guitar setup was to the right of the drums," Kral continues, "with his Marshall stack in a little alcove, gobos all around for isolation, and recorded with a 57 close to the cabinet and either a FET 47 or U87 about three feet back, for a bit of air."

20, 20, 24 Hours To Go

"The start time for most rock & roll sessions was around noon, and we began by setting up the drums, getting a drum sound, getting a bass sound and getting a guitar sound without anyone there. I don't think the band showed up until four or five in the afternoon, at which point we started tracking. They were working from Dee Dee and Daniel's demo, and they'd also recorded all of the rehearsals at SIR, so that's what they referenced for tempo and key, and I don't recall the tracking itself being a laborious process. By six in the evening, tape was rolling, and way before 11 at night we had a master take. No comp'ing, no edits, just a straight take.

"It's not like we were dumping things into Pro Tools and moving them around, and there also wasn't any punching-in on the basic track. Don't forget, Jean Beauvoir is a pretty darn good musician himself, he's got some great music chops, and he knew what he wanted to get out of these guys. What's more, having rehearsed with them beforehand, he also knew what they needed and expected, and whatever coaching needed to be done. In other words, he knew what the headphones needed to sound like for Johnny to play along, and what Marky needed in that regard. He was the fifth Ramone that day.

"The only real problem occurred when the drum tech set up the kit and, because this was an important recording session, decided on his own to change all the drum heads. Everything went fine until he got to a couple of the toms and stripped the lugs. That became an issue. It was so late in the day when that happened that, instead of renting gear from SIR, it was decided that Marky would press ahead with what he had. Well, if you listen to the song, there are really no drum fills until the end, and we even punched in some of those fills. Due to the lugs being stripped, they weren't holding their tuning, so after one pass around the kit it was over. That meant, after we had a solid take and he did the one live fill, we just punched in one or two fills.

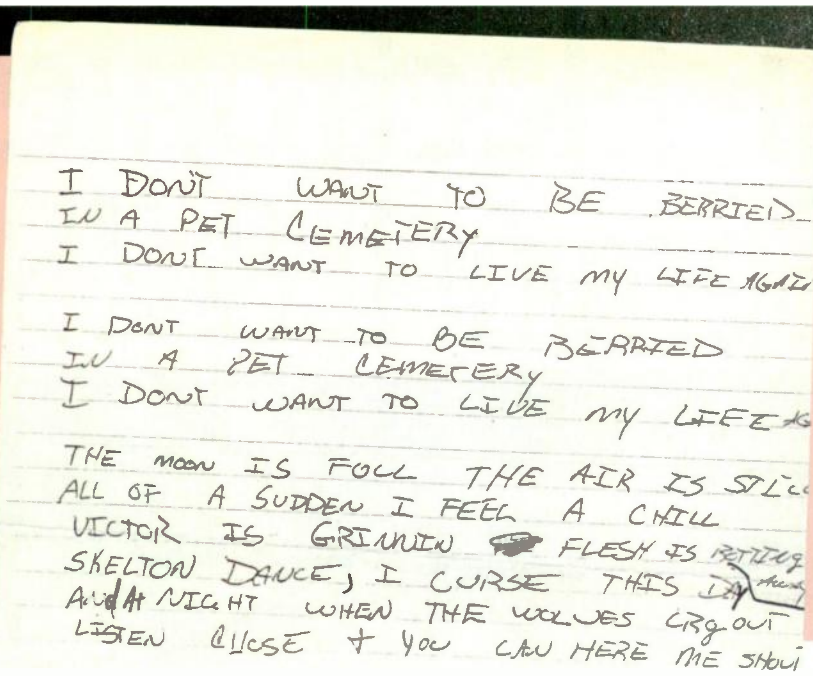
Conscientious, Professional & Just Plain Good To Know (!)

"I was pleasantly surprised by the Ramones. I liked them, I was a fan of theirs, and on this occasion they were certainly disciplined, in that they came in and did the job that they needed to do. There wasn't any kind of fooling around. In fact, Johnny was a really serious character. When he came in, he was focused, he knew what he had to do and he was just spot-on. I remember single-tracking his rhythm part once the final take was established, then double-tracking that part. There was none of that changing his guitar for another sound — he just double-tracked it again that same night, and then boom, gone. With Marky it was the same thing. He was great,

Track Sheet

This is how the track sheet for 'Pet Sematary' looked by the end of the tracking sessions. To the right is a lyric sheet with some of the slightly eccentrically spelled original words to the song.

- **1.** Hi-hat
- **2.** Kick drum
- **3.** Snare
- **4 & 5.** Stereo kit
- **6 & 7.** Room stereo — cymbals and room (near)
- **8.** Room (near and far)
- **9.** Guide DI'd bass
- **10.** Live guitar
- **11.** Overdubbed bass
- **12.** Double-track of Johnny's guitar
- **13.** Daniel's overdubbed guitar
- **14.** Percussion overdub
- **15-20.** Lead vocals
- **21.** Comp'ed vocal
- **22.** Backing vocals
- **23.** Synth
- **24.** Timecode


I DONT WANT TO BE BERRIED
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THE MOON IS FULL THE AIR IS STI
ALL OF A SUDDEN I FEEL A CHILL
VICTOR IS GRINNIN FLESH IS ROTTING
SKELTON DANCE, I CURSE THIS DAY
AND AT NIGHT WHEN THE WOLVES CRY OUT
LISTEN CLOSE + YOU CAN HERE ME SHOU

really communicative, and once the drums were done, boom, gone.

"It was then left to us to keep on working. The objective was to get a decent basic track, work on overdubs through the evening, have Joey come in at night to lay down a rough vocal and feel it out so that there could be more overdubs, and then return on Friday to do some punching-in. Everything went pretty much according to plan. He initially came in at one or two in the morning, and again, he was a sweet guy, a total pro. He listened to what we had, walked around doing his vocal exercises and getting loosened up, and I put up a U87 going to a Neve mic pre, with a Urei 1176 compressor. We worked with Joey until about six or seven in the morning, using about six tracks for his vocals, and then comp'ed a good vocal which we gave him on cassette. After he left, we continued working until 10 or 11 Friday morning, doing more overdubs, including backing vocals by Jean and Daniel [*Rey*]. Then we went home and came back at five in the afternoon for vocal overdubs.

"The idea was that Joey had lived for several hours with the cassette we'd given him, and so now we could punch-in and do any fixes that he thought — or we thought — were needed. There was a very minimal amount of that, maybe just a couple of hours, and once that was done we continued overdubbing through Friday night, including synth pads doubling the guitar in a higher register, as well as some percussion, gated handclaps and Daniel's single-note guitar solo, using a Fender Twin in Studio 5 with the spring reverb. That was the kind of flourish that changed the track from sounding like traditional Ramones, because Johnny could never have played that solo — as anyone who ever saw the band live would know, four chords and that was it! Still, in this case, what Daniel added, driven by Jean's momentum, was being aimed at the film, whereas traditional Ramones would have meant Johnny's rhythm parts being tripled or quadrupled."

Gabba Gabba Hey!

"We finished overdubbing at about five or six in the morning, went home, slept for a few hours, and were back at the studio before noon on that Saturday in order to set up the mix," Kral recalls. "It was one of the fastest mixes. For Jean and Daniel it was their second week working on it, so they knew the song inside and out, and since I had already been doing this for three days there was no guessing as to what was on the tape. We were therefore actually printing the final mixes before five that afternoon; the choice mix, which had more vocal level, slightly more background and a little bit more lead guitar; and the album mix, which had more of a controlled, live band sound. What we didn't do was a separate mix for the film. We just sent them a stereo mix and that's what they used, because it only came in at the end of the movie."

The following day these remixes were handed over to the record company and to the production company. A couple of months went by. Then came the call for a remix, which took place at the Hit Factory on February 16th, 1989.

"In typical record company fashion, they said, 'We love the mix, here are the changes,'" Fernando Kral recalls. "They wanted a louder lead vocal, more of the background vocals — they really wanted to hear those words 'pet sematary' — and while they liked the rhythm guitars they also wanted some more ringing lead parts. Nothing unexpected. Anyway, the session began at midnight, Jean and I were probably ready to start remixing by about three or three-thirty in the morning, and we were trying to beat the clock because Jean had parked his car on 54th Street. As it turned out, the car got towed, but the remix sounded great.

"Looking back, it was a terrific experience for me, doing that song. The band was just so good, even at the rehearsals, and they all seemed to be getting on and really into the whole thing. What's more, they booked one of the best studios in New York, Sigma Sound, which came loaded with top-quality equipment — when you booked that studio, you didn't just get the engineer and the room, you also had access to tons of gear, including three or four different amplifiers, some excellent pianos and some great outboard gear. So, it was a higher-end production for them, and I remember after it was all done and everybody was happy with the record, I heard the comment, 'We ended up spending more than twice the cost of the first album on this one song.'

"Well, that's what happens when you spend a week at SIR, three days at Sigma and another day at the Hit Factory!" SOS

SE Electronics Gemini II & Z5600a II

Valve Microphones

SE have updated their Gemini and multi-pattern Z5600a mics with extra facilities and better technical specs. Are they worthy contenders for space in your mic locker?

Paul White

SE's original fixed-cardioid-pattern Gemini microphone (reviewed in *SOS* June 2004, www.soundonsound.com/sos/jun04/articles/segemini.htm) broke with tradition by offering tube technology in a transformerless circuit, resulting in a microphone that combined the warmth of a tube mic with the airy top end of a solid-state, transformerless model. The vast majority of tube mics have transformer-coupled outputs, due to the need to match the high impedance of the tube circuitry to the lower impedance associated with microphone signals, but that design can soften the high end to a noticeable extent. Sometimes this is exactly what you want, but at other times it would be nice to retain a bit of 'air'. That's just what the Gemini aimed to do, and SE have now updated the concept with a new version, which I'm looking at here alongside another SE tube-mic update, the Z5600a II (first reviewed in *SOS* in May 2003; www.soundonsound.com/sos/May03/articles/sez5600.asp).

Gemini: The Return

You may recall that the first Gemini mic used two dual-triode tubes, hence its impressive girth. Housed in a similarly chunky 80mm x 230mm body, the Gemini II again features dual tubes, this time a 12AX7 for the input stage and a further 12AU7 in the output stage; the latter takes over the role of the more usual transformer. There are no transistors or FETs that I could see, so this appears to be a true all-tube design.

A switchable 10dB pad has been added to the new mic, along with a switchable low-cut filter, and the included PSU is SE's new updated design, with a chunky, brushed-aluminium case and improved electronics for lower noise. The current range of SE tube mic PSUs feature a controlled power-up LED, which blinks for around 30 seconds to give the tubes a chance to warm up, but (as with tube mics in general) it's best to allow the whole mic to warm up for half an hour or more before using it, not only to get the circuitry to operating temperature but also to help drive off any condensation from around the capsule. This particular PSU features no controls other than a power switch, and aside from the IEC mains inlet there's just the eight-pin XLR to connect to the mic and a conventional three-pin balanced XLR output. A screw-locking (both ends) eight-pin cable is included with the kit, along with the shockmount and PSU. The mic itself has a foam-lined wooden case, and all the kit elements fit into a sturdy aluminium carry-case that weighs only slightly less than my guitar combo!

As far as I can tell, this revision uses a similar 1.07-inch, gold-sputtered, centre-terminated, cardioid capsule to its predecessor, mounted inside a large, dual-layer mesh basket on a shock-absorbing support. The twin tubes are fitted in porcelain bases, while the rest of the circuitry is mounted on two double-sided, glass-fibre circuit boards. All the components are discrete and appear to be of high quality, and the standard of workmanship is impressively high. As with the earlier version, there are no clips to hold the tubes in place but they don't seem in any way inclined to work loose. Both of the newly added switches (for pad and low-cut filter, as mentioned above) are sideways-mounted, miniature toggle switches with their functions labelled on the metalwork.

The Gemini II is predictably weighty, so it would benefit from a substantial stand to avoid toppling. At the bottom of the mic is an eight-pin connector for the cable that links the mic to the PSU, and the shoulder of the protruding XLR housing is threaded so that it secures to the shockmount's integral locking ring, allowing the mic to be operated at any angle, including fully inverted. The shockmount itself follows the familiar metal-cage construction, with fabric-covered elastic hoops supporting and isolating the inner section. I had some criticisms of the original Gemini's stand mount, as it was prone to working loose under the weight of the microphone, but this new version has a toothed pivot assembly that locks into place very firmly. The mount also comes with a thread adaptor, so will fit both European-thread and US-thread mic stands.

Alternatives

It's unusual to find a transformerless tube mic such as the Gemini II, and the only other company I can think of that make one are CAD, whose less costly M9 employs solid-state output components in place of the second tube the Gemini uses at the output stage. Turning to the Z5600a II, If you're in the market for one of these the Rode K2 is also worth a look.

Tech Talk

The Gemini II has a 20Hz-20kHz frequency range, but with a gentle hump up at around 10kHz to add an airy presence to the sound. Its sensitivity is quoted as 12.6mV/Pa, and its noise spec, at EIN 12dB A-weighted, is a significant improvement over the 16dB A-weighted managed by the original version. Its maximum SPL for 0.5 percent THD@1000Hz is 135dB, some 5dB better than the Gemini I, and the output impedance is the same nominal 200Ω.

Subjectively, this Mk II version sounds quite similar to the original Gemini, which is a good thing, as I know a lot of users really like the Gemini sound. As I mentioned above, the transformerless design would appear to combine the most recognisable attributes of tube and solid-state microphones, providing the expected tube warmth and low-end weight but without losing the high-end sizzle, as can happen with 'tube plus transformer' designs. Because the presence peak is right up at 10kHz there's no tendency to over-emphasise the harsher upper-mid range, though singers wanting a slightly more rounded top end may want to try using the mic slightly off-axis to soften the tone. I did, however, notice that the basket

Like its predecessor, the Gemini II boasts twin valves, one for the input stage and one for the output stage, which can clearly be seen through the mic's case.

assembly of the mic resonates at quite a low frequency if tapped, resulting in a very audible rumble. Use of the included shockmount, as well as a separate pop shield, is highly recommended.

Because of its high-end clarity, the Gemini II also makes a useful instrument microphone, its capabilities extending from acoustic guitars and other plucked instruments to percussion and wind. If you were thinking of buying a Gemini II as a main vocal mic for your own use, you'd

SOUND ON SOUND

SE Gemini II £899

pros
- Nicely engineered.
- Distinctive tonality that is both warm and airy.
- Comes with shockmount, cable and PSU in a sturdy case.

cons
- Basket assembly could have better damping.

summary
While the Gemini II isn't a cheap mic, it isn't overpriced by any means, given its performance and the fact that it is a tube microphone. Its transformerless tube circuitry gives it a distinctive 'larger than life' sound that is similar to that of the original Gemini, but it has an improved spec and benefits from pad and LF roll-off switches.

need to make sure it suited the character of your voice, as you would with any microphone, but for anyone who already has one or two conventional solid-state and tube microphones the Gemini provides a distinctive tonal alternative and thus would be worth adding to a collection.

The Gemini II isn't a budget microphone — indeed, it costs more than some of the big-name European models, which may give you a hint as to the direction SE are taking for the future. They have a wide range of affordable microphones, and that looks set to continue, but they also have some innovative design ideas that they can apply to more up-market products, and the Gemini II is clearly in that category.

The Z5600a II

Lighter on both the pocket and the wrist than the Gemini II, the Z5600a II is, again, an update of an earlier SE tube microphone, this time the Z5600a. The Z5600a II is packaged in a similar way to the Gemini II, complete with new-style PSU, locking eight-pin cable and shockmount. Unlike the Geminis, however, the Z5600a II is a multi-pattern microphone offering nine pattern variations, going from omni, via cardioid, to figure-of-eight. The rotary pattern-control switch is located on the PSU alongside the eight-pin mic connector and the three-pin XLR balanced output feed.

As the microphone offers multiple patterns, it is equipped with a dual-diaphragm capsule (back-to-back cardioids), again 1.07 inches in diameter and centre terminated. Like the Gemini II, -10dB pad and LF roll-off switches have been added since the last version. Conceptually, this mic is much more conventional than the Gemini II and features a dual-triode ECC83 tube feeding into an output transformer, so it sounds rather more like you'd expect a tube mic to sound, with plenty of body

The Z5600a II kit includes shockmount, power supply, cable and separate wooden microphone box, all inside a larger carry-case. The Gemini II comes similarly packaged.

and warmth and a slightly smoothed-out high end. However, the capsule has a broad presence peak rising towards 10kHz, so this isn't a dull-sounding tube mic.

The Z5600a II's frequency response is specified as 20Hz-20kHz, with a sensitivity of 14.1mV/Pa (0dB=1V/Pa 1000Hz). Improvements to the PSU and internal circuitry have lowered the self-noise to a creditable 12dB A-weighted, while the maximum SPL for 0.5 percent THD@1000Hz is now 135dB. In general, the physical construction of the basket and housing (and the shockmount, in this case) is similar to that employed in the Gemini II and other recent SE models, and comes across as very solid and sensibly engineered. The basket assembly rings slightly if you 'ping' it with your finger, but that doesn't seem to come across in the sound and it doesn't produce the low-end grumble, when handled, that the Gemini does.

Sounds Tubular

Tonally, this mic has much in common with its earlier incarnation, producing a full and generally smooth vocal sound with a bit of lower-mid 'chest frequency' support that may help those with thinner-sounding voices. It does sound a bit lacking in character in direct comparison with the Gemini II, but in reality that is probably because it is intrinsically less coloured. The high presence boost helps keep some 'air' around the top end, but the subjective result is still smooth. As I'd expected, the sound opens up more in omni mode, and although it's not as transparent as a single-diaphragm, dedicated omni model, the Z5600a II works well in omni mode for capturing room ambience or for close-miking acoustic guitars and other instruments. In figure-of-eight mode, the sound is closer to what you'd expect from the cardioid pattern than the omni. In accordance with the laws of physics, both the cardioid and figure-of-eight modes exhibit a proximity effect that exaggerates the low end when used close up, which the omni mode does not. Nevertheless, the tonal consistency between the patterns is pretty close.

At around half the cost of a Gemini II, the multi-pattern capability of the Z5600a II makes it rather more versatile, but it doesn't have the unique Gemini II timbre. It does, however, have a classy sound, especially on vocals, and because it doesn't stray too far from the expected tube mic tonality it may work with a wider range of voice types. It certainly offers good value, and even if you feel a cardioid-pattern mic is all you really need, those additional patterns may well come in useful from time to time. SOS

SOUND ON SOUND

SE Z5600a II £449

pros

- Characteristic warm, flattering tube-mic sound.
- Well built.
- Affordably priced.
- Comes with shockmount, cable and PSU in a sturdy case.
- Multiple polar patterns.

cons

- No obvious cons.

summary

The Z5600a II sits at the more affordable end of the multi-pattern tube-mic market but delivers on both sound and build quality.

information

£ Gemini II £899 ; Z5600a II £449. Prices include VAT.
T +44 (0)1582 470260.
W www.sonic-distribution.com
W http://seelectronics.com

Secrets Of The Mix Engineers: Rich Costey

Iconic rockers The Foo Fighters had a worldwide smash with their record-breaking track 'The Pretender'. Rich Costey was the man behind the mix faders.

Paul Tingen

The Foo Fighters' 'The Pretender' is one of the most successful alternative rock songs in the history of American music. It enjoyed the longest ever spell at number one in the Modern Rock Tracks chart (18 weeks), it was the most played alternative rock song on US radio in 2007, and as this article went to press it had been nominated for three Grammy awards, for Record of the Year, Best Rock Song and Best Hard Rock Performance. The song didn't quite break the hegemony of the all-pervasive hip-hop/R&B genre, only reaching number 37 in the regular Billboard charts and number eight in the UK. However, it did help propel the album on which it features, *Echoes, Silence, Patience & Grace* (also boasting two Grammy nominations, for Album of the Year and Best Rock Album), to the top of hit parades around the planet.

Clearly, the Foo Fighters hit a home run with 'The Pretender'. Whether this is due to the in-your-face arrangements and performance, the exuberant chorus, the political subtext or the confrontational video is a moot point, though one imagines it was a bit of each. The success of the song is impressive by any standard, but neither mixer Rich Costey, nor the band, nor producer Gil Norton, saw it coming. Costey, who has become a highly in-demand producer and mixer in recent years, specialising in artists known for their musical integrity — such as Muse, Mars Volta, Fiona Apple, Franz Ferdinand and Interpol — mixed the whole of *Echoes, Silence, Patience & Grace.*

"While we were mixing all the album's songs through," recounts Costey, "we didn't really have a conversation as to what would be a single. I mixed 'The Pretender' later on, and we knew it was a strong song, but for us it was just another mix."

The Power Of The Bees

Costey was speaking from Avatar Studios in New York, where he had been on mixing duties for most of the winter. The mixer and producer is unusual in that he tends to mix entire albums, rather than just individual songs. "I think I was hired to mix the Foo Fighters' album because they wanted a second, or perhaps rather a third or fourth, pair of ears on the project. There are two schools of thought in mixing, one being that you give the mixer something raw that he cooks the hell out of, and adds loads of herbs and spices to make a dish out of it. I have done projects on which this was the case.

"My work with The Foo Fighters fell into the second category. They had already done a lot of prep cooking and things were in very good condition. So I had to preserve what they had done to a fairly large degree, and this meant that my work mainly consisted of balancing and rides to get the dynamics to

Rich Costey, photographed in New York's Avatar Studios with some of his personal collection of outboard gear. The skull in the background is actually a binaural mic!

happen. Obviously, a certain amount of processing was still necessary, but that was more to enhance what the band, Gil and engineer Adrian Bushby had already done. It's sort of like the mixer's Hippocratic oath — the first rule is Do Not Kill The Patient. I think there are all sorts of people out there who are killing the patient. I try very hard not to add to that problem.

"There were two main challenges in mixing 'The Pretender'. The first was a general one concerning the Foo Fighters, namely their endless walls of guitar overdubs, almost like a swarm of bees. It is something the band goes back and forth about, because Taylor [*Hawkins*] always wants the drums to be louder, and Dave [*Grohl*] prefers the power of the bees. The guitars on 'The Pretender' are quite full on, with countermelodies and so forth, and they all tend to be in the same range, so it gets quite dense. The challenge of this type of mix is to retain the power of the track, yet define a space for everything. Handling the guitar balance was a slight chore, and in comparison the drums and vocals were quite easy.

"The other difficulty with 'The Pretender' was in getting all the dynamics to work properly. The song starts out with a very stripped-down intro — a small string section, guitar and vocals — then ramps up into the chorus, after which there's another breakdown, then it ramps up again and there is a series of breakdowns in the middle eight, including a repeat of the intro, and finally the last chorus with octave guitars coming in, that you really want to be able to hear on top of the wall of everything else. The only way to do this type of mix is to simply do loads and loads of fader rides. I don't think any amount of EQ can substitute for doing rides.

"That's not to say I'm shy on the EQ. I'm not sure where I developed this habit, but I don't cut many frequencies when I mix. I used to cut a lot when I was younger, but eventually I almost entirely stopped doing it. It wasn't a conscious decision; I guess I just don't like the sound. I'll cut a little bit on the drums sometimes, and with vocals I occasionally need to control problem areas, and I also may be quite brutal with synthesizers. But that's about it. I know there are people who cut frequencies to try to layer a track, but look at it this way: if you are conducting an orchestra, can you filter the woodwinds so the oboe doesn't stick out so much? What you do is have the oboist play a little bit more softly, so you can hear the rest of the reeds."

The Importance Of The Intro

"If you look at the 'Pretender' Edit window, you see that it's really neatly organised and ready to be mixed for someone who has never seen the Session before. That's Adrian Bushby's doing. I'm a bit of a slob with my tracks; if you see a Session that I produce it will look like a mess, because I know where everything is. I have assistants that help me organise stuff, and they will also do this when a Session comes in for mixing. I'll then join them when it comes to laying things out on the console. Often I'll also have to sort through the Pro Tools Session anyway, before I get far into mixing, because frequently there are more music tracks than console inputs. I'll spend a while familiarising myself with what's there and developing track-sharing schemes.

"I think on most of the album there's a little bit of guitar track-sharing, as it was mixed at The Pass in Los Angeles on a Neve 8078 console that had 40 full in-line channels with EQ and 32 monitor channels. 'The Pretender' had 70-odd tracks, so if I wanted the possibility of EQ on each track, some sharing of similar tracks was inevitable. I don't mind doing that, but I rarely submix in Pro Tools. You might later find that a viola in the string section is obscured by a baritone guitar and you want to be able to adjust for that.

"After getting a sense of what works and what doesn't work in the rough mix, and what the artist wants, I'll start a mix by pushing all the faders up for the vocals and the rhythm section and so on, to see how different parts are interacting. The lyrics and vocal delivery greatly influence how the track will sound. The dynamics of the vocal absolutely shape everything. The vocal should be driving the band wherever possible. So while mixing it's pretty critical to have the vocals in there early on, otherwise you might find when you push the vocal up later that you have to start all over again.

"I tend to start with getting the tones for the vocals and different instruments right, using various combinations of compression, EQ and distortion, and after that it's just rides, rides, and panning. My strategy usually is to mix in sections. Once the sonic relationships between the instruments are more or less stable, I will start getting the intro to sound great. I try a few different approaches, because with any song, if the intro isn't great it doesn't matter what happens to the rest of the song. That's something I learned a long time ago: if the song is going to work, the intro has to grab you immediately. The importance of this cannot be overstated.

"Once the intro makes sense, you also ▶

have a template for the way the rest of the song works. I'll work on the verse, and then the chorus, and so on, and gradually build up the track. Then I start listening to everything from the beginning in context and make sure all the transitions are OK, and then I try to push it up to the next level. That's all just rides. While I'm working in sections, I listen to everything, not just to individual tracks."

'The Pretender'
Artist: The Foo Fighters
Writer: David Grohl
Producers: Gil Norton and the Foo Fighters

Drums: Quad 8 310, Pultec EQP1, API 550a, Neve 33609, API 2500, Lexicon 960, Urei 1176, Smart C2, SPL Transient Designer, room mics

"I had just gotten some Quad 8 310 EQs before I began the Foo Fighters project, and I found that cranking the shit out of the mid-range with the Quads helped the drums quite a lot. On most of the album the bass and snare drum went through the Quad 8s, with severe amounts of mid-range added. Quad 8 grew out of Electrodyne and was very popular in the 1970s, particularly with film companies. I obtained some channels that came out of the Motown Sunrise console and they're basically three bands of EQ and a mic pre and output level, and the Q gets tighter as you push it up. The Quad 8s are a little bit rounder and warmer-sounding than similar API modules. I also used the Pultec EQP1 and API 550a on the bass drum.

"I had a bunch of different side-chain compressors on the drums that would change from song to song. On some songs it would be the Smart C2, medium ratio and fast recovery, on others the Neve 33609, the Urei 1176, the API 2500 or the Empirical Labs Distressor, and the SPL Transient Designer on toms. You can use the SPL to lengthen the sound of the toms. I don't compress all the drums at the same time, I'll compress individual parts and mix the compressed sound in with the natural sound of the drums. There was relatively little compression implemented on the drums in this song, because the band didn't care for it. They wanted the drums to sound more raw.

"Throughout the album mix I might have used a bit of Lexicon 960 on the drums for reverb, but the room mics — amongst them a heavily compressed foldback microphone — were so good that I tended to use those. I tend to like room mics that are on the darker side. If they're too bright, you can't turn them up very loud because you then also get all kinds of messy cymbal noises. One other thing to note is the drums that come in after the intro of the song. When the whole band is slamming accents like that, you want to make the drums sound very aggressive, and this meant pushing room mics, pushing

A bit of Lexicon 960 reverb was used on drums for the Pretender mix, but Rich Costey liked the sound of the natural reverb on the room mics so much that he tended to use those — including a compressed foldback mic, called 'listen mic' in this neatly arranged Edit screen.

Very few plug-ins were used on the mix for 'The Pretender', but intro vocals were treated to Reel Tape Saturation and Pultec EQP1A, while a vocal overdub at the end of the track was processed with Sound Toys' Echoboy.

One of Rich Costey's racks, containing an impressive array of classic and modern gear, including a Chandler EMI TG12345 Curve Bender EQ; two Mercury EQ-Hi Program Equalizers; Thermionic Culture Vulture; an SPL Transient Designer that was used to lengthen the sound of the toms; an SPL De-esser that was applied to the vocal tracks; a Roger Mayer Model RM58, which compressed the background vocals; a pair of Universal Audio 1176 limiters, which Costey used to make Dave Grohl's guitars "more aggressive"; an EAR 822Q EQ; and a pair of EAR 660 Limiting Amplifiers that came into play for bus compression.

compression mics and so forth. As I said, there were a lot of rides."

Bass: Neve 1073, Pultec EQP, Chandler TG1

"The bass consisted of three tracks: one recorded through an Ampeg SVT amp, one through a 4x12 Marshall cabinet, and one put through a Sansamp. I usually push all three tracks up and adjust their balance if I want different colours. I'll also mix in some side-chain compression, which in this case was a 1073, Pultec EQP, and Chandler TG1. The side-chain will usually be bright as hell and really compressed. One of the interesting things regarding mixing bass is that you may think that the bass sound on its own is fairly bright, but when you add the rest of the instruments you find that the top mid-range of the bass is usually missing. It's sometimes quite shocking to realise how much top end you need to add to a bass to make sure it cuts through a track. If you then hear the bass sound in isolation it may sound pretty uncomfortable, but in the midst of a swirling din of a dense track, that amount of top end usually works just fine."

Guitars: Neve EQ, Urei 1176, Neve 33609, Gates Sta-Level, API 550a

"Dave's guitar tone is a little bit more distorted, with more bottom end in comparison with Chris's. So there is method to their guitar madness, where they try to get everything to fit in with each other. I EQ'ed the guitar parts a little differently on the board to make them fit, and did rides when there were melodies that needed to be featured. Another way of creating separation was through panning. I tend to pan the rhythm guitars hard and the melody guitars slightly inside them. Overall the guitar parts sounded great, though, so they really didn't need a lot of processing. I tend not to compress distorted guitars very much when I'm mixing, because they've usually already been compressed. It's strange, but to my ears, when you compress distorted electric guitars while you're mixing, the whole mix starts to sound overcooked. Of course, it depends on the song; I don't have any rules.

"In the case of 'The Pretender' I had a couple of Urei 1176 compressors on Dave's guitar, to pump them up a bit and make them sound more aggressive whilst barely compressing. I had the Gates Sta-Level compressor on the clean guitar in the intro, with quite a lot of compression and a very slow recovery, plus a 33609 with a very short recovery time on Chris' rhythm guitars, to beef them up a little bit. I worked hard to make the octave guitars in the last chorus stick out. I ended up EQ'ing them with the 550a, in addition to the console, to make them cut through this formidable army of rhythm guitars. The rest was just rides. The stereo phase and flange guitars used in the track already had those effects on them."

Strings: room reverb, Neve BCM10

"There were five tracks of strings and two stereo room tracks for the strings. I know I said that I don't do usually do submixes, but I actually did submix the strings through my Neve BCM10 sidecar, and brought them up on a couple of faders on the main desk."

Vocals: Mercury EQH, UA 175, Roger Mayer RM58, Waves DeEsser, Digidesign Reel Tape Saturation, Sound Toys Echoboy, Pultec EQP1A, SPL De-esser, Neve Portico 5042

"If I recall correctly, the lead vocal is doubled throughout the song. I used a Waves DeEsser plug-in on it, which I'm quite a fan of, as I've not heard a hardware de-esser that is competitive with it. The main signal chain after that was a Mercury EQH tube EQ and then a couple of vintage UA 175s. I hit the background vocals with the Roger Mayer RM58. I haven't mentioned any plug-ins yet, apart from the Waves DeEsser, because I hardly used them on 'The Pretender.' The only other ones I used were the Reel Tape Saturation and the Pultec EQP1A on the intro vocals, and the Sound Toys Echoboy on one of the end vocal overdubs.

"I used board EQ on all the vocals. The Pultec EQP1A plug-in on the intro vocals was to help them cut through. The Reel Tape Saturation was a plug-in that I had just bought. I wanted to warm the vocals up a little bit with it and make them a little bit crisper. Nothing too distorted. The Echoboy plug-in was used on a vocal overdub at the end for a tight delay. I don't tend to use plug-ins that much; they're really not that interesting, in my opinion Finally, I also used an SPL De-esser on some vocals, and the Neve Designs Portico 5042 for a bit of crispness. It has its own sound, and I used it a lot on the whole Foo Fighters album.

"To complete the picture, I used an EMT plate and the room mics for general reverb, and my two EAR 660 limiters were my bus compressors, together with the Manley Massive Passive. We printed the mix to the ATR102, on half-inch analogue tape, from which it was mastered." SOS

Studio SOS

This month we tackle problems that were causing a bass-heavy mix and use coat-hangers to produce better vocal recordings!

Paul White & Hugh Robjohns

Howard Bragen seems to have no trouble coming up with good songs, but he called *Sound On Sound* because he was having problems with his mixes. He said that they sounded fine in his room, but came over as very bass-heavy elsewhere. I suspected that his studio layout and general acoustics could use a bit of help, but when I asked him to describe his room and his equipment setup, he told me that he used a pair of Spirit Absolute Zero monitors. That's a model that can be rather bass light, in a large, untreated room such as Howard's, so I thought this might explain why he was mixing with too much bass end. As Howard was thinking of upgrading his monitors anyway, he went out and bought a set of KRK VXT8s (he chose these because his main musical collaborator also uses them), plus some substantial 1.14-metre stands from Studiospares, but he didn't set them up before the day of our visit.

Howard lives in a flat that is part of a large, converted country house, with very high ceilings (approximately 3.5m). The room he uses as a studio also doubles as a living room and bedroom and is almost square, at roughly 4.75m x 4.25m. Because of the way the furniture was arranged, the Absolute Zero speakers were set up on lightweight Samson stands behind the computer desk, which meant that the two computer screens sitting on the desk were partially obscuring them. This setup is undesirable because sound bounces off any solid object in its path, and in this case both the tonality and stereo imaging of playback were adversely affected. Furthermore, the computer desk was around half a metre to the left of the centre line of the room, and a tall chest of drawers immediately to the right of the desk meant that correct speaker placement was not possible.

We persuaded Howard that we should move the drawers to the left-hand-side wall between the windows, as this would give us space to set up the system more symmetrically — and with the speakers sited so that they would now be firing past the computer screens, and not into the back of them! We also felt that putting some acoustic foam on the expanse of bare wall directly behind the speakers would cut

Right: Howard Bragen's studio as we found it on arrival. His new KRK monitors hadn't been set up and his existing Spirit Absolute Zeros were set up on very lightweght stands and partically obscured by the PC monitors.

Below: We set up the KRKs on much sturdier speaker stands, as well as placing them on Auralex MoPads, which tilted the speakers towards the new monitoring position.

down reflections and improve the stereo imaging. The side walls were quite distant, and because of the location of the windows it would have been impractical to add further acoustic treatment there, so we decided that we would try to find a solution that wouldn't involve treating them. Likewise, the ceiling was so high that it was impractical to treat that — and probably unnecessary too.

Getting Started

After an introductory coffee, we moved the drawers and decommissioned the Spirit monitors, before dragging the computer desk to a central position about 500mm from the wall. We assembled the new speaker stands (which would benefit from being filled with dry sand at a later date, in order to add mass and damping) and used a pair of Auralex MoPads, with their angle wedges in place to direct the new KRK speakers downwards towards the listening position. Normally some Blu-tak is all that's needed between a speaker and a solid stand, but if we'd placed the monitors directly on these tall stands the tweeters would have been above head height, which isn't ideal. While we were rewiring the monitors into the back of Howard's music PC, Hugh rationalised the mains wiring to ensure that we had a 'star' system, with all the audio gear fanning out from a single power point: this is a good idea, as it minimises the risk of ground-loop hum.

Howard wasn't using a separate monitor controller, which meant that level control had to be done in software. When you're working at 24-bit resolution, the drop in resolution as you turn the level down isn't serious, as long as the monitor input-gain trims are set to somewhere near the maximum level you're likely to monitor at, with the on-screen level fader most of the way up.

Sticky Business

As this was a rented flat, we didn't feel we could go sticking Auralex acoustic foam directly to the walls, as removing it can make a bit of a mess. Instead we tried Auralex Temp Tabs, which are essentially self-adhesive Velcro fastenings. However, as self-adhesive tape doesn't stick to foam very well, Auralex also include some small plastic rectangles, the idea being that you glue these to the foam using their own adhesive, then stick the Velcro pads to these. We used the supplied Auralex caulking-gun adhesive rather than the usual spray, and though it was easy to apply, it took rather a long time to set. For the first hour or so it exhibited all the adhesive properties of Marmite on a warm day, so as time was of the essence we took a trip to the local DIY store and bought some Evo Stik contact adhesive. By the time we returned, the Auralex glue was starting to grab, but to make sure the panels stayed up, we added another plastic tab at the top centre of each sheet and fixed this in place with Evo Stik, which did the trick for us. With the Velcro pads stuck in place, and their mating halves clinging to them, we stuck four vertical panels to the wall behind the speakers, centred at roughly (seated) head height. This would cut down on sound being re-reflected from the hard wall behind the speakers, and we hoped that soft furnishings elsewhere in the room would reduce the amount of unwanted sound bouncing around the place. I've since tested the Auralex adhesive and found that although it can be slow to set compared with many instant-grab adhesives (including the excellent Auralex spray adhesive), after a few hours it goes rock solid and is extremely strong.

Fixing foam to walls in rented accommodation is something we're often asked about. By gluing plastic tabs such as Auralex Temp Tabs (pictured) to the foam, you're able to use sticky-backed velcro strips to mount the tiles without causing cosmetic damage.

Testing The New Setup

We weren't quite sure how the room would behave at the low end, as an open ▶

Sending Out An SOS

Howard Bragen uses his studio to produce CDs of his own songs based on a whole range of influences, from blues, country and jazz to latin. He plans to spend a month in the USA in March 2008, working to place some of his material with publishers in order to get coverage from well-known artists. His latest album is near completion and will soon be at mixdown stage, ready for its release in 2008. Watch the news page at www.internetmusic.com as well as Howard's MySpace page at www.myspace.com/howardbragen for release dates.

archway coupled the room to the adjacent kitchen, which was about half as wide but just as high and long as the main room. We tested the system using a Steely Dan track that we all knew, and were pleased to discover that the bass end was pretty even and the stereo imaging acceptable — so we wouldn't need to treat the side walls. To get a subjectively correct balance of bass end, we had to set the speakers in their 'full space' mode, using the rear-panel switches, as the half-space setting sounded a touch bass-light. Because the speakers weren't close to the walls, we felt this was OK.

Next, we put up one of Howard's own mixes, which was indeed a bit heavy in the low bass region, and it also sounded slightly low-mid heavy. We set up a parametric equaliser to see if I could improve on the spectral balance, and I ended up cutting the low bass slightly, dipping the lower-mid at around 350 to 400 Hz to take out a residual hint of boxiness, and adding some 12kHz 'air' by using a Q setting of around 0.5 and a gentle boost.

Though subtle, this clarified the mix and brought it into focus, which Howard liked. Though he was using Cubase as his DAW, he was mixing the outputs via his Yamaha DSP Factory PCI card, which was controlled using XPI software, as he much preferred the XPI interface over the Yamaha original. We couldn't find any way to bypass the EQ other than a section at a time, so A/B testing wasn't as quick and precise as we'd have liked, but it was generally agreed that reducing the low bass and low mid helped achieve a better tonal balance for the mix. However, we stressed to Howard that now the new monitoring system is in place he shouldn't do any overall EQ or compression to his mixes if he plans to have them professionally mastered — which is something that he's seriously considering. It can be almost impossible to undo the effects of EQ and compression, and as the mastering engineer is likely to have more sophisticated processors to do the job, as well as a far more accurate monitoring environment (not to mention years of experience!), it's best to leave everything to them.

Howard usually sings close to his computer, so we improvised an acoustic shield to keep out unwanted room noise, using wooden coat-hangers and Auralex foam.

Vocal Recording

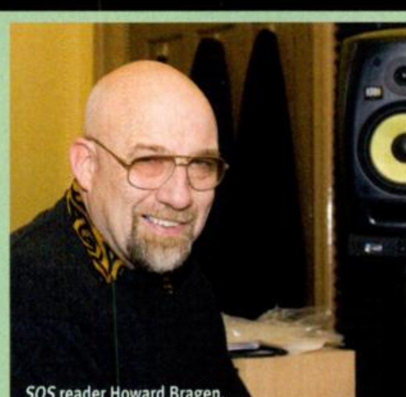
Much of the time Howard records his own vocals while standing close to the computer keyboard, so that he is able to operate the system and sing at the same time. This meant that our usual 'duvet across a corner' trick wouldn't work, so we made up a couple of foam panels that Howard could hang up behind him when singing. These would reduce the room reflections bouncing back into the live side of his AKG C414 XLII microphone, which is normally set for cardioid pattern mode and feeds into the system via a digitally connected Focusrite Twin Trak preamp. The Twin Trak also provided latency-free input monitoring via its headphone output. I needed a simple way to hang up the two panels (a couple of two-foot squares left over from a previous job) from boom mic stands so they could be arranged in a 'V' shape, and arrived at the simple solution of glueing a wooden coat-hanger to the top of each piece, and simply hooking them over the clip adaptors on the mic stands. We had the panels finished in a few minutes and the improvised hanging system worked fine — so I'll probably use that trick again! An SE Reflexion Filter or something similar placed behind the mic would improve the isolation from the room ambience even further, so we suggested that this would be a worthwhile and cost-effective upgrade.

Howard was pleased that we'd been able to make a useful improvement to his monitoring environment without damaging the room or compromising his living environment, and he liked the idea of our hanging panels for vocal recording, as these could be stowed out of sight when not in use. He liked the sound of his new monitors, which seemed to have a well-balanced and reasonably extended low end, and he could see from the way I'd EQ'ed his earlier mixes that the combination of untreated room and the Absolute Zero monitors had been misleading him. SOS

Reader Reaction

Howard: "Working with the new KRK VXT8 speakers that Paul and Hugh set up for me has been a massive improvement. I bought them partly because of a review in *SOS* of KRK's previous model, and because I've used those in the studio in Cardigan owned by Jon Turner. The whole listening experience is less tiring and I feel I can now trust the sound of my mixes to a far greater extent.

"The Auralex foam on the wall has killed some of the sonic 'nasties', and the ingenious solution of hanging the foam pads on coat-hangers from mic stands has helped too, in that I've got fewer unwanted reflections from the room getting back into the mic whilst I'm recording vocals, and I can hear this in the results. I also really liked Paul's EQ settings for the song I played to them and I intend to try mixes using those same (or similar) settings, as well as taking a flat mix to a mastering suite and making comparisons. Thanks again to Paul and Hugh!"



SOS reader Howard Bragen.

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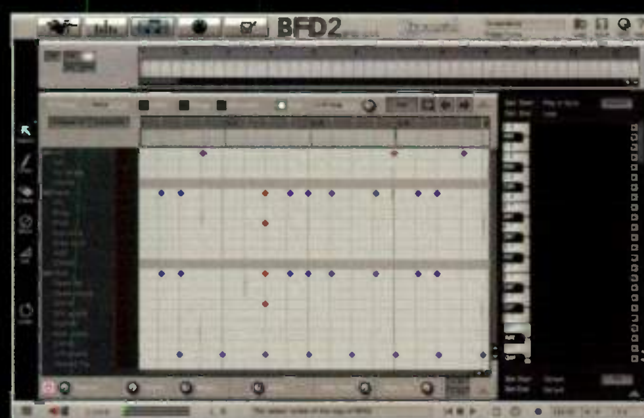

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Way Out Ware Kik Axxe

Software Synthesizer For Mac & PC

Gordon Reid

Way Out Ware have recreated the classic ARP Axxe in software, added a step sequencer and virtual drum machine and given it a ridiculously low price-tag.

Way Out Ware are perhaps best known for their flagship TimewARP 2600, a soft synth based on the ARP 2600. Warmly received in the USA, this didn't have as great an impact on this side of the pond, perhaps because Arturia had already beaten WOW and their distributor, M-Audio, to the market with their 2600V.

Today, WOW are trying a slightly different approach, attacking the low-cost end of the market with a product that is again distributed by M-Audio. Kik Axxe is marketed primarily as an affordable soft synth modelled on the ARP Axxe, but it's much more than that, with programmable percussion sounds and two 16-step sequencers (one for the synth and one for the drums). Nevertheless, let's start, as one always should, at the beginning...

Axxe Facts

ARP introduced the Axxe monosynth in 1975 as a low-cost alternative to the Odyssey and ARP 2600. Despite sporting a relatively limited architecture of a single oscillator, a single filter and a single contour generator, it was a fine-sounding performance instrument, and ideal for newcomers to synthesis. Consequently, it proved to be quite popular, and remained in production until the demise of ARP in 1981.

The synthesizer part of Kik Axxe mimics the original Axxe almost perfectly. What's more, and unlike every other 'soft' recreation of a monosynth that I have reviewed, it's monophonic, which means that it will run on computers that are, by today's standards, of modest power.

For the purposes of the review, I placed my Mac next to my original ARP Axxe, and ran Kik Axxe as a stand-alone application, as a VSTi within Plogue Bidule and as an AU within Plogue Bidule. In each case it worked correctly, which I found surprising for a version 1.0 product. I directed the outputs from the Axxe and the Mac to adjacent channels on my mixer with all EQ and effects defeated, and then attempted to set up the same sounds on each. The results were not identical but, if I ignored the physical positions of the faders (the calibrations of the controls are quite different), I was able to reproduce the sounds of the Axxe on Kik Axxe with reasonable accuracy.

The Tape Delay Effect

Unlike many soft synths, which add numerous effects to the original synth's specification, Kik Axxe adds just one: an emulation of an Echoplex delay unit, called Tape Delay. The mixer section allows you to send the outputs from the synth and drum sections independently to this, and a return control allows you to balance the delays with the original signal.

I don't have an Echoplex with which to compare Tape Delay, but I would be surprised if the original device had the bandwidth of the Kik Axxe effect. Oh yes... and it certainly was nowhere near as noise-free as the Kik Axxe effect. Umm... and you can set the Kik Axxe effect to repeat ad infinitum, but the delayed sound remains undistorted and otherwise unmolested. Oh, OK — the Kik Axxe delay is nothing like an Echoplex (or, for that matter, any other tape delay) but it's useful, so I'm not going to complain!

Oscillator Action

In the quest to discover how alike Kik Axxe and the vintage ARP Axxe sound, I started with the oscillators. Analysis shows that, with the filter wide open, my Axxe's and Kik Axxe's square waves are quite similar to one another and, if you set the controls appropriately, they react similarly to PWM. Likewise, the sawtooth waves are similar at high frequencies. But, as you reduce the pitch, the Axxe's wave becomes rounded while Kik Axxe continues to produce something that is close to an ideal sawtooth. Then, if you then drop deeper into the bass region, both traces become rounded. Despite the visible differences in the mid-range, the timbres of the two waves are similar and, unless you're a real anorak, the discrepancy is (probably) not enough to worry about.

Nonetheless, there's an error in the oscillator that needs to be fixed. I created a sound using the sawtooth alone, played a note, and, while holding that note, increased the level of the square wave. With the square wave at 50 percent of maximum amplitude, the fundamental was eliminated and the pitch of the note went up by an octave. If I increased the square-wave amplitude to maximum, the fundamental reappeared. This suggests that the amplitude of the square wave is double that of the sawtooth, but with the phase inverted, which is not how the original Axxe behaved!

The Filter & Envelope Generator

Using a simple audio analyser, I found that the highest frequency of self-oscillation of my Axxe's filter lies at 13.4kHz, whereas Kik Axxe's goes all the way to 20.4kHz at a sample rate of 44.1kHz. These figures do no more than demonstrate how misleading such figures can be, because I found that the Axxe sounded more 'open' for some sounds, while, for others Kik Axxe was the brighter. In the end, I stopped worrying about it; the responses of the two filters are slightly different, that's all. If I introduced a second ARP Axxe into the equation, all three synths would be different. Nonetheless, there are some tangible differences in the implementation of the filters. In particular, Kik Axxe's LFO sweeps the filter over a far greater range than on the Axxe (which is a good thing) and, of course, the ADSR can sweep the Kik Axxe filter further because of its higher maximum cutoff.

All of which bring us to the envelope generator itself. This appears to be a conventional ADSR, but there is something strange about it. I set up a simple 'square' contour (Attack, Decay and Release to zero, and Sustain to maximum) and there was a massive glitch at the start of each note. A few moments investigation demonstrated that, to eliminate this, I had to set the Sustain level to a little under 90 percent of its full value. This means that the developers have programmed Kik Axxe in such a way that the Attack finishes well below the maximum Sustain level. This is wrong, but you can turn it to your advantage... If you set the Sustain level to be higher than the end of the Attack, the Decay acts like a second Attack stage, rising to the final level. This means that you can create 'A(1)A(2)SR' contours that are impossible to obtain using a conventional ADSR envelope generator.

The maximum Attack time, at six seconds, is as it should be. Furthermore, despite the tool-tips stating that the maximum Decay and Release times are six seconds, they are actually in the region of 18 seconds, again as they should be. Unfortunately, there's a problem with the envelope triggering. The ARP Axxe was, like many synths of its era, low-note priority with multi-triggering, but Kik Axxe is last-note priority with single triggering. This can make a huge difference, whether you play the soft synth from a keyboard or sequence it, and I don't know why WOW made such a basic mistake.

Like the Axxe, Kik Axxe has an external signal input. Unlike the Axxe, however, the soft synth adds a level control, an Envelope Follower that applies the external signal amplitude to the filter cutoff frequency, and a 'Smooth' control to reduce glitching in the follower. This is good stuff. ▶

SOUND ON SOUND

Way Out Ware Kik Axxe **£39**

pros

- Like its inspiration, it's a simple and useable synthesizer.
- The drum sounds and sequencers are more useful than you might imagine.
- The whole package is excellent value for money.

cons

- There's an error in the oscillator when mixing waveforms.
- There are a couple of errors in the envelopes and triggering.
- The on-screen manual is inadequate, and there's no paper manual.

summary

Most soft synths concentrate on power, polyphony, additional modulation capabilities and effects. This means that there is room for at least one that acts as a true monosynth, and that is quick and simple in use. Kik Axxe fulfils this role well, and its integrated synthesizer and rhythm sequencing enhance its late '70s feel in an entirely complementary fashion.

Playback
Readers' Music Reviewed
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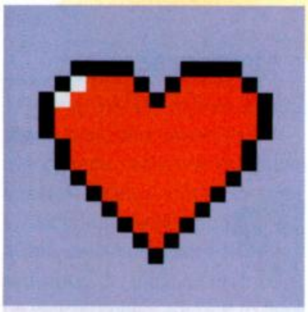


Pixelh8
The Boy With The Digital Heart

If you thought programming music on a Yamaha QY70 or an old Akai MPC was time-consuming, just think how fiddly it must be to do it on a Game Boy! But that's what Matthew C Applegate, aka Pixelh8, has done on *The Boy With The Digital Heart*, an electro-pop album of real songs made up of bleeps and boinks. But, you may ask, why bother? What's the point? I don't want to enter into a debate about why people choose to create music on unconventional platforms, but the obvious retort is: why not?

Musically, the tracks are very clever, and it's not all *Tetris*; there's drums, bass, harmonic and melodic sounds, and they've all been created using what is obviously fairly basic synthesis. All in all, it sounds like synth-pop should, just a bit more computer game-y. One reservation I have is that the songs are quite repetitious. Even though they're mostly around the three-minute mark, they soon get a little tiresome. They are, of course, instrumental, but even so, they could have been more interesting, or just shorter.

Of course, Pixelh8 could introduce some vocals, and this would certainly add interest to the songs (though it might lose him some credibility among Game Boy geeks). I can easily imagine a cutesy female singer like Kate Nash or Lily Allen (when she's not whingeing) singing some fun little lyrics over the top. *Chris Mayes-Wright*

W www.myspace.com/pixelh8

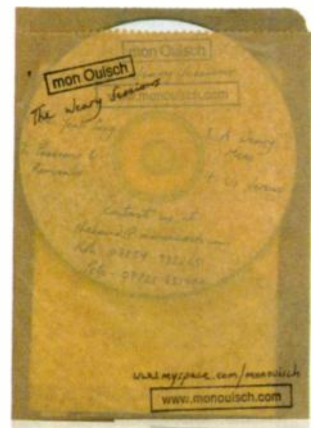
Troy Banarzi
Euphonika

Euphonika might well have been CD of the Month, but for the fact that Troy Banarzi already has an Arts Council grant to fund its production. There's a lot of inverse snobbery and Philistinism around subsidised art, and a tendency to dismiss projects like Euphonika as pretentious or difficult. All I can say is that if you do so, you're missing a treat.

The CD originated as the soundtrack to a "quasi-theatrical sound and vision performance", yet, unlike much 'soundtrack' music, it feels perfectly complete even without the visual or dramatic elements. There's a folk influence and a fairy-tale quality to Banarzi's style of composition; often dark, yet always accessible, it expertly balances a string quartet against other instruments more associated with pop, such as the Hammond organ. Several of the pieces are overlaid with lengthy speech samples, and on the second track, 'Mother Machine', the results are truly spine-chilling, even if some of the others stray slightly too far into schlock-horror territory to be as effective.

It's also beautifully recorded. 'Toy' instruments such as the melodica often come over as irritating novelties, but in this case they convey exactly the right blend of gaucheness and menace. I always like to hear string players who can bring out the potential of their instruments for brutality as well as sweetness and light, and the in-your-face sound suits the music perfectly. In a word: stunning. *Sam Inglis*

W www.banarzi.com



mon Ouisch
The Weary Sessions

Mon Ouisch's four-track demo probably cost them about sixpence to put together, but with the help of a rubber stamp, a paper bag, nice handwriting and a bit of imagination, they've created something that stands out from the crowd. All that time spent in front of Blue Peter obviously hasn't gone to waste.

I was expecting their music to be as minimal and low-key as their packaging, but far from it. Like so many indie bands these days, mon Ouisch hark back to the golden age of punk and new wave. Instead of joining in the collective worship of Gang of Four or Joy Division, though, theirs is an impassioned and intelligent racket that recalls the righteous anger of the Adverts or Stiff Little Fingers. It's still as retro as you can get without actually serving Spangles, but it's refreshing to hear a band set about their instruments with so much urgency and so little irony or self-conscious cool. Oh, and they have some decent songs, too, which is even better. *Sam Inglis*

W www.monoiusch.com

Drum Racks In Ableton Live

A simple drum machine, with individual channels displayed in the mixer.

New in Live 7, Drum Racks takes the Instrument Rack concept in a percussive direction. Read on to see what it can do for you and your drums...

Simon Price

If you caught last month's Live 7 review, you'll already have a good idea of what Drum Racks are for, but we'll start with a quick recap. (There's also an excellent video at www.ableton.com that summarises the features of Drum Racks). A Drum Rack is a specialised form of Instrument Rack, with functionality focused on drums and percussion. Like other Rack types (Instrument and Effect), Drum Racks are shells for building devices from multiple plug-ins. A Rack comprises one or more parallel Chains, each of which is a sound source and signal path. This concept is ideal for building drum machines, because each drum sound can have its own generator or playback device, with its own signal path, filters, effects and so on. All these components are hosted within a single device — the Drum Rack — so that you can treat it as one instrument on one MIDI track.

Using Drum Racks

The screen above shows a simple TR606 drum machine patch made with a Drum Rack. The most obvious difference between Drum Racks and other Racks is the trigger pad section. In this example, each pad plays a single sample, and is triggered by a single MIDI note from a keyboard or pad controller.

The individual channels of a Drum Rack can be controlled in the main Session View mixer. The screen shows how a Drum Rack track can expand to display all the separate Chains in the composite device. From here you can set relative levels and pans for individual drum sounds. Selecting any of the channels in the mixer immediately displays the devices in that chain at the bottom of the screen.

Patterns can be recorded into Clips on the MIDI track containing the Drum Rack patch. However, at a later point you can decide to

Pad Control

Drum Racks have been designed to play nicely with MIDI pad controllers. The four by four grid has been adopted as a standard by all the main hardware units, such M-Audio's Trigger Finger, Korg's Pad Kontrol, and Akai's MPD 16 and 24. I had my eye on Akai's MPK49 (reviewed last month), but its 3x4-pad grid is a deal-breaker since Live 7.

The neatest feature, when using a pad controller, is that if you have one of the directly supported units (which includes the Trigger Finger and Pad Kontrol, as well as Korg's Kontrol 49 and Micro Kontrol), you can select which pads are being controlled from the software and the controller will follow. To the right of the main pad view there's an overview of all 128 pads, with an outline indicating the pads that are currently in view. Moving this square (by clicking or using the mouse's scroll wheel) brings different banks of pads into view. Supported controllers will automatically control the pads that are currently focused. For this to work, you usually need to select the first factory preset scene on your controller.

We'll take an in-depth look at customising MIDI controllers to do this, and all kinds of other nifty stuff, in a future Live workshop.

Drum Rack pads don't just have to be sample-based. In this screen, the Operator FM synth is being used for the snare sound.

strip a sound out of the Drum Rack and treat it as an individual track. To extract a sound, simply click and drag its channel to an empty space in the mixer. A new MIDI track will be created containing the sound's device chain. Any MIDI Clips that were on the Drum Rack track will be duplicated on the new track, with just the notes for that sound.

Sound Sources

An important point about Drum Racks is that they are not instruments in themselves. Each sound must be generated by one of Live's other instrument plug-ins (or a third-party plug-in) hosted in the Rack. When you drop a sample onto a pad, an instance of Simpler (Live's basic sampler) is created on that pad's Chain. However, any instrument plug-in can be used to generate sounds in a Drum Rack. In the screen above, another pad has been added, this time using Live's optional FM synth, Operator. The Operator patch was simply dropped onto the pad from the Browser.

When using synths as sound sources, you need to set which note is triggered by the sound's pad. If you look at the parameters in the Chain list, or in the expanded mixer view, you'll see fields for Receive and Play. Receive is the MIDI note that plays the pad for that chain. Play is the note that gets sent to the instrument on the Chain. By default the Play note is always C3, which will play back any samples at their original pitch. For synths, you need to change this to get the correct note when the pad is triggered.

By editing the Receive fields, you can create Pads with layered sounds. In the screen below, a second snare sound has been added. This was initially dropped onto a spare pad, but the Receive note was changed to D1, the same as the first snare. The D1 pad now displays 'Multi' (this can be renamed), and the pad triggers both Chains at once.

Building A Drum Rack Patch

Unless you have the optional Drum Machines or Session Drums packages, you will only have two Drum Rack patches in your Live library. It's therefore essential to learn how to create your own kits to get the most out of Live 7.

Start by opening the Browser view, selecting the Live Devices tab and opening the Instruments folder. Drag the master Drum Rack (not a patch from its sub-folder) to an empty space in the Mixer. A new MIDI track will be created and the Drum Rack's pads page will appear in the Device View.

Next, add some samples to the pads by switching to one of the three file-browsing tabs and locating a sample you wish to add. In the picture on the opposite page, I've opened a folder of TR606 samples from my library, and dragged them to pads in the Rack. When you move the cursor over the pads, suggested sound types are displayed in the status bar, based on the General MIDI standard drum map.

Essential Selections

After you've dropped some samples onto pads and triggered a few, you'll notice two things: the level is a bit low, and sounds cut off when you release the note. In most cases you'll want to remedy these issues by changing some settings in the Simpler instrument. You can display the Simpler for any Chain by double-clicking it.

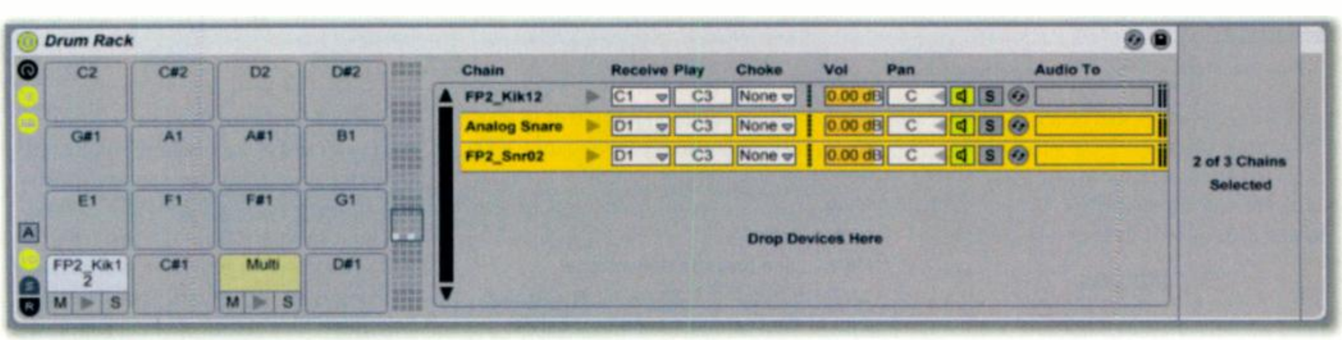
First we'll sort the level. The output level of the Simpler defaults to -12dB. You could just push this up, but a more sophisticated approach is to adjust the Velocity response. In Simpler, adjusting the 'Vel' parameter changes the dynamic range of the sample playback both up and down from the basic Volume setting. At a Vel value of 50 percent, the sample should nearly peak when you hit the keys or pads hard.

Next we'll make sure the entire sample plays back, even from momentary triggers. This is achieved by pushing up the Release knob in the Volume Envelope section. A final adjustment that's recommended for drum samples is to turn the number of voices assigned to the sampler down to one. This makes sure that you don't get multiple soundings of the same sample at once, as you wouldn't with real drums. All these settings can be seen in the screen at the start of the article.

Making these settings each time you add a sample soon gets tedious, so check out the 'Default Drum Drops' box for advanced tips on saving your own default Simpler settings. Once you've dropped a few samples onto your Drum Rack, you can switch quickly between different sounds, to make adjustments, by selecting their pad or by selecting their channel in the expanded mixer view. An even quicker method is to enable the small square 'A' button in the Rack's view selector. This lets you display Chains simply by triggering them from your MIDI controller.

Adding Effects

Once you've got your sounds in place, you can add some effects. You can add separate effects to each sound, add global effects to the output of the Rack, and create Return channels within the Rack. To add an effect to a single sound, simply drag it from the Browser to a pad, or to a channel in the ▶



Pads can be set to trigger multiple Chains, to create layered sounds.

extended mixer view. The effect will be added to the end of the Device Chain. You

Default Drum Drops: A Great Time-saver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Mix Management In Cubase

Crafting the perfect mix is a difficult task, but often all you need is a bit of organisation to make things quicker, easier and more rewarding.

John Walden

Group channels are a good means of regaining some control over unwieldy mixes. By default they're placed together (as shown in the Mixer view here) but you can move them around like other tracks to form subgroups for multi-miked instruments or other complex setups.

I often hear people saying that mixing is a mixture of art and science, and it is true that a good mix demands both creative and technical skills. But you also need a good dose of organisation and administration to get the best out of your mix, so this month, I'll look at how you can configure your Cubase projects to make the technical side of the mixing process more efficient and, as a result, make it easier to focus on the creative side of things.

Perhaps one of the most basic difficulties in organising a mix arises because of the almost limitless track counts that most DAWs offer on a modern computer. With various sections of guitar, synth, vocal, bass, drums and other bits of ear candy each spread over multiple tracks, even the basic task of setting the relative volumes of each instrument within a mix can become cumbersome. Cubase offers all sorts of options for managing the mix of a complex project, but two things are probably key: being organised in the Project window, and making good use of Group channels. So, let's consider what you can do to manage your mix more efficiently...

Project Management

Although it is always tempting to just go with the creative flow when tracking, taking a little extra time to organise your tracks within the Project window can make it much easier to pick your way through the various takes when it comes to mixing. Simple functions such as colour-coding tracks and using folders to keep related tracks together can make a big difference to project navigation. I find that this is particularly helpful when you find yourself having to come back to a project at a later date: the more clearly organised the materials are, the easier it is to recall exactly what stage the project had reached, and how different tracks are routed or related to one another.

Different projects might call for slightly different approaches: an obvious tactic would be to create folder tracks to organise the audio and MIDI tracks, and an alternative might be to use folders for instrument groups — drums, bass, guitars, vocals, and so on. Whatever approach is adopted, folders can provide a useful means of organising the various takes in a project.

Folder tracks can be added to the project via the Project / Add Track menu option. Once created, other tracks (for example, audio or MIDI tracks) can simply be dragged and dropped into the folder from the Track List in the Project window. The folder can then be collapsed, allowing you to hide the tracks contained within them, reducing clutter in the Project window while you shift your focus to another of the elements in your project.

Navigation through extensive track counts can also be made easier by colour-coding your tracks or parts. If you enable the Show Track Colour option (via the blue, green and red striped button located above the Track List), this adds a narrow Track Colour Selector bar to the right edge of each track within the Track List. Clicking and holding the mouse on this bar will bring up a palette of colours to select from. Colour-coding a folder track will apply that colour to all objects within the folder, and this can be useful if you have instrument groups organised by folder (for example, all bass parts blue, drums red, guitars green...). You can also colour individual parts on a track, so you could colour, say, the chorus part of your guitar track differently from the verse part. It doesn't matter what colours you choose, although it does help if you are consistent across different projects.

Count Down

The mix can also be made easier by reducing the number of tracks you have to deal with — and this is where Group channels come into play. For newcomers to multitrack recording, the purpose of Group channels can easily pass you by. In many ways, they are the equivalent of 'subgroup' channels found on many hardware mixers. The outputs from individual channels within the mixer (for example, from audio tracks) can be routed to a Group channel and, in turn, the Group channel is routed to the main outputs of the mixer.

Let's take a simple (but quite typical) recording task involving drums, bass, two guitars and vocals. By the time you have used a multiple-mic setup on the drum kit (for example, separate mics for kick, snare, hi-hat and overheads and a distant 'room' pair), two or three different mics on the guitar amps and perhaps even multiple amplifiers, everyone has overdubbed a few takes, and then you've added several tracks of backing vocals, the track count soon adds up and starts to feel unmanageable.

While it is perfectly *possible* to mix a large number of tracks, it is easy to see how things can become a bit complicated. The best takes for each song section for a particular instrument might be scattered across several tracks and, even with colour-coded folders, there will still be a lot of faders to look after when it comes to the job of mixing.

Track and fader counts can be kept under control in a few ways. One is to use Cycle Recording (as discussed in the Cubase column in *SOS* July 2007) but this will be less practicable if the tracking for the project occurs over several sessions. Another is to use the 'link and unlink faders' option that is accessible by right-clicking on the mixer channel having selected multiple tracks. But things generally get much easier if Group channels are used. For example, imagine a guitar part requires a three-mic setup (for example, one on the grille, one at 12 inches and a more ambient room mic) and is recorded to three mono audio tracks in Cubase. The three tracks can be routed to a Group channel named 'Guitar 1', and stored with the Group track in a folder. The same could apply to a second guitar part.

Prior to Cubase 4.1 there was a limitation with Groups, in that you could only route them to other Groups that were created later (see the Cubase column in *SOS* Dec 2006) — but thankfully, C4.1 removed this limitation. The two 'Guitar' Group channels can be routed to a further 'Master Guitar' Group channel, which can be used to control the overall level of the guitars relative to other elements of the mix, such as drums, bass, synths and vocals.

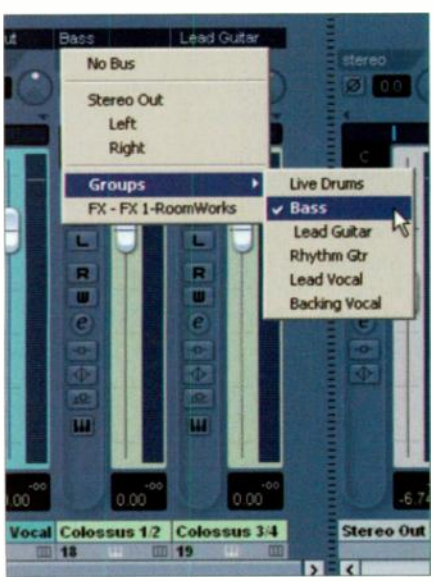
Applying the same approach to each instrument allows a project extending over many tens of individual audio tracks to be focused down to a small number of Group channels. The first screenshot in this article shows a mixer view where a single Group channel has been created for each of the key elements: drums, bass, guitar, lead guitar, vocals and backing vocals — with six faders in all, making things much more manageable. Each of the individual audio tracks has been routed to the ▶

Stem The Flow

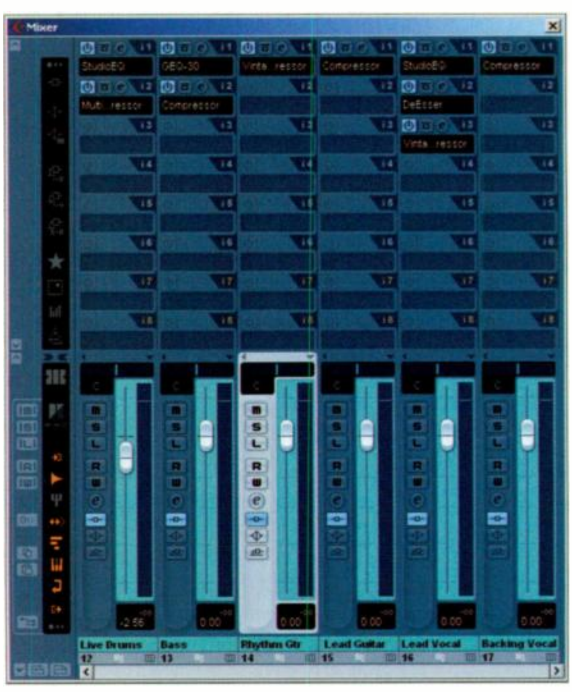
One of the new audio routing possibilities added in the recent 4.1 update was the ability to identify Group channels (and FX channels for that matter) as the input sources for audio tracks. The most obvious application for this is the creation of mix 'stems'. A stem is a submix of a group of related instruments — bass, guitar, drums and vocals might be four such 'stems' that together form the full mix. Such mix stems are often used when composers supply music for film or TV projects, as they give the mix engineer greater flexibility when combining the musical, sound FX and dialogue elements into the final audio mix of the project. For example, with an orchestral score, separate stems might be provided for each of the major orchestral sections — strings, brass, woodwind and percussion. Mix stems are, however, also becoming something that some mastering engineers are requesting when working on straight musical projects, as they can provide them with greater flexibility to shape the overall sound at the mastering stage.

Stems could be created in earlier versions of Cubase, but it involved soloing each Group channel in turn and performing an Audio Export for each stem required. In 4.1, the Input Routing box automatically offers any existing Group channels as a possible input source for an audio track. By configuring a series of stereo audio tracks to be fed from each of your Group channels and putting them all into Record mode, all the stems can be created in real time with a single pass through the project.

Incidentally, 'stems' also provide a useful way of archiving a project. Each major instrument group can be preserved as a stereo audio file. If, at some point later in time, a remix is required, these audio files can be used rather than returning to the original Cubase project. Though this doesn't give as much flexibility as a full project, the stems can be used on any DAW platform, and the approach can be useful where plug-ins or even sequencers used for the original mix have become obsolete.



Audio tracks and VST Instruments can be routed to a Group channel, via the Mixer window (as shown here), the Project window Inspector, or a track's Channel Settings window.



Where possible, it's useful to apply effects plug-ins and EQ at the Group level. Not only will this reduce the strain on your computer, but it also makes tasks like automation much more convenient.

The mixer's Common Panel allows the display of particular track types to be toggled on or off to simply the display. Hovering the cursor over the grey/orange icons will tell you what each one does.

appropriate Group channel via the Output dialogue (either from the Project window Inspector or via the Routing panel available within the mixer). While some balancing of levels using the individual tracks will still usually be required, once this is done, all the key elements of the mix can be balanced from this small number of faders. Incidentally, if you're using a hardware control surface such as a Mackie Control, placing the Group channels together at the top of your track list will generally give you control of these channels without having to scroll through lots of banks of faders to get to them.

As with audio tracks, you can assign the audio outputs of VSTis to Group channels. If they are multitimbral instruments and feature multiple audio outputs, specific instrument types can be routed first to a particular output from the VSTi and, from there, to the most appropriate Group channel. For example, when combining sound sources from more than one orchestral library within a project, the various VSTi outputs could be used to send all the string sounds from the different libraries to a single Group channel, from which the 'master' string level can then be controlled. Similar Group channels could be configured to deal with each of the major orchestra sections.

A Lighter Load

As well as reducing the number of faders you need to worry about, Group channels bring a second form of efficiency. Where you have multiple tracks of the same source, such as several backing vocals sung by the same singer, you often find that you want to apply the same processing to all of them. Group channels have the same capacity for using insert and send-return effects as do individual audio tracks, so it's more efficient to apply a single plug-in at the level of the Group channel than to use individual instances on every source track.

The benefits of this approach can be seen very clearly from our earlier example of the four-piece band, where six Group channels (drums, bass, guitar, lead guitar, vocals and backing vocals) were being fed by over 40 individual audio tracks. In this kind of situation, applying EQ and other processes at the Group channel level makes a lot of sense, as only one chain of effects is required per Group channel rather than per audio track. It's much quicker to set up and change an EQ setting for a single Group channel than for half-a-dozen audio tracks. However, this approach is not always appropriate for all plug-ins and all sources. For instance, you are unlikely to want to apply the same EQ settings to every individual mic in a multi-miked drum kit, while dynamics plug-ins may respond differently to grouped signals than to individual sources.

A New View On Mixing

As with other aspects of Cubase 4's user interface, the Cubase Mixer window can be customised in a variety of ways. This is useful when mixing, as the last thing you really want to deal with is a virtual mixer that requires repeated scrolling across tens of audio, MIDI and other channels (which can easily happen even on multiple-screen setups). If you have your project organised into a relatively small number of Group channels, then the lower portion of the Mixer's Common Panel can be used to toggle off the display of all the other track types.

You can create custom View Sets via the buttons at the base of the Common Panel.

As the Devices menu allows multiple instances of the Mixer window to be open, it is possible to configure one just to show the Group channels while a second may show all channels. It is then easy to move between the two, depending upon whether you need to work with the major mix elements (for example, the overall balance between drums, guitars, bass and vocals) or the detailed balance between different tracks for a specific instrument (for example, the balance between the different mics of your multi-miked drum kit).

Mono To Stereo

Group channels can also be used to apply stereo effects to a mono audio track. If you have a mono track and use a stereo insert effect such as chorus, the spatial element of the effect is lost. However, if you route the mono track to a stereo Group channel and insert the effect within the Group channel, then the stereo effect can be heard in its full glory.

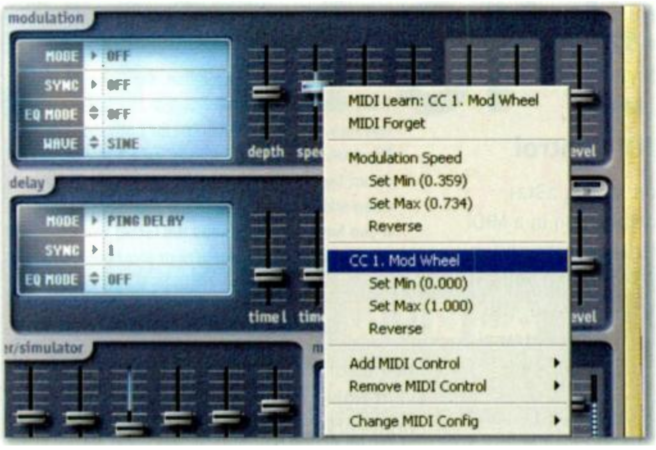
However, perhaps a more flexible way of moving between different views of the mixer is via the Channel View Sets facility. When you've decided on a combination of channels to display, the current configuration can be saved as a View Set, via the buttons located at the very bottom of the Common Panel. Each View Set can be named and then instantly recalled, which is very useful for switching between different elements of the mixer to work on different tasks. If you're able to master the show/hide function, you can even store different mixer views for groups of instruments — for example, you can show only the multitrack drum audio channels.

Templates

It is well worth creating a template project that contains a number of Group Channels pre-configured and ready to roll, along with other elements such as FX channels or VST Instruments that you find yourself using on a regular basis. If you frequently work on material that's based on similar instrument groups (for example, orchestral music or guitar bands), it is also well worth including a series of empty audio tracks all pre-routed to named Group channels. The same might apply if you often process the individual outputs of an instrument like BFD in the same way. While any such template is likely to require modification and additions for each project, it will save a considerable amount of setup time, meaning that the creative urge is less likely to slip out the back door because it got bored of waiting!

In many cases, the mix will still be an iterative process — involving moving back and forth between the (well-organised, folder-based) individual audio tracks and the Group channels. You may be thinking that there's not anything that's radically new or magically creative about folders, colour-coded tracks or Group channels, and you'd be right. But I'm surprised by how little use some people make of these facilities. Getting your material organised and the mixing process down to a more manageable number of faders can make a big difference to the ease with which your initial mix can come together. SOS

USING Z3TA+ FOR EFFECTS



In this example, the modulation effect's Speed control has been restricted to a small range, as shown by the blue line in the fader's 'groove'. The pop-up menu shows that the Modulation Speed range has been restricted, but the Mod Wheel travels over its full range to cover this restricted range, thus providing a type of fine tuning.

▶ As you turn your MIDI controller's hardware knob or slider, the parameter will vary only within the minimum and maximum levels you specified. However, there is a limitation: the new range you've set is not spread across the hardware knob's full range: restricting it doesn't provide 'fine-tuning', but merely ensures that the value won't go beyond the specified minimum and maximum values. (As you've probably figured out, Reverse changes the 'sense' of the parameter. In other words, turning up a control when you have it set to Reverse actually reduces the parameter value.)

If you do want fine-tuning, you can achieve it by setting a 'master range' for the control. The method is very similar to the one just described. First, set the virtual slider (or other control) to the desired minimum value, then right-click on the slider and under the name of the *parameter*, this time (for example, Modulation Speed), click on Set Min. Now set the virtual slider to the desired maximum value, right-click on the slider, and under the name of the parameter, click on Set Max. A blue line appears in the slider's 'virtual groove' to show the range you've specified, and when you assign a controller it will vary the parameter value over the range you specified, but this will be spread over the full rotation or travel of the hardware control.

If a parameter has multiple control sources, you can see the parameters for only one source at a time in the menu that pops up when you right-click on a parameter's virtual slider. However, you can choose which parameter is shown simply by moving the hardware control before you right-click on the parameter's slider.

Real-time Effects

You can also use z3ta+ for real-time processing of whatever input you're feeding into Sonar. The basic procedure is, in an audio track, to first set the audio Input field to pick up the appropriate signal from your audio interface (mic, guitar or whatever). In the audio track's effects bin, insert the z3ta+ audio effects plug-in (or z3ta+ soft synth — and if you're using the soft synth, remember to click on Options and select Audio Input Thru). Now turn on Input Echo in the audio track (the button to the right of the Record button).

You'll now be able to monitor the effects as you play, sing or send in any real-time input. Of course, this works best with low-latency systems!

A Cubase Screen Shot?!

OK, this is a Sonar workshop, but after all z3ta+ is a stand-alone Cakewalk product too, and it has a feature that, ironically, doesn't work in Sonar but does work in Cubase 4: the ability to send a track's audio not just through the effects section, but through the z3ta+ filters and VCAs as well. This is possible because Cubase can assign a MIDI track to a VST plug-in effect (it doesn't just do instruments), so you can trigger the envelope generators from a keyboard.

When you install Sonar 7, it actually installs two versions of the z3ta+ effects section: z3ta+_fx and z3ta+_fx2. The one with the fx2 suffix is the one you want.

In Cubase, insert z3ta+_fx2 into the insert of a VST audio channel (of course, you should also have audio on this track, so that there's something to process). Create a MIDI track and assign its output to z3ta+_fx2. Unless you want to play the synth section at the same time, go through each oscillator and turn each wave to 'Off'. Now your MIDI keyboard can trigger the envelopes; you can play with the filter parameters, drive parameters from LFOs, and so on. Fun stuff! SOS

Z3ta+ inserted in Cubase 4, which can process audio through the filters and VCA, as well as the effects section. In Cubase, an accompanying MIDI track allows triggering of the z3ta+ envelope generators.

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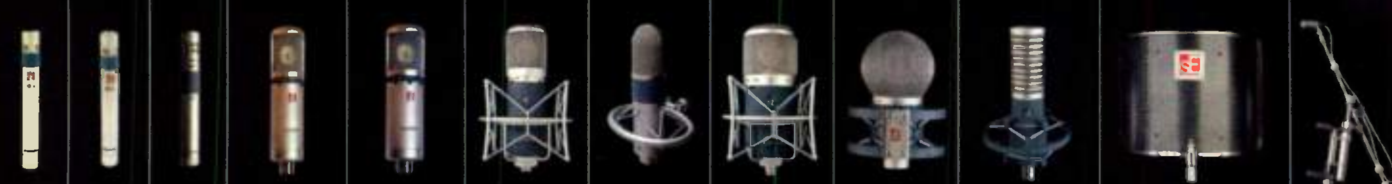

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recording at these new 'guide' Region boundaries. In this example, I've used the snare transients as the guide. First, you need to select all the tracks you want to manipulate and assign them to a Group — but don't add the track that you've just chopped into separate Regions.

Select the track that contains the split Regions and use the Select next Region, Set Locators by Regions and Goto Left Locator key commands to move the Playhead (referred to as the SPL or Song Position Line before Logic 8) to the start of each of the newly split Regions. If you then click on one of the Regions that are assigned to the Group, that will select them all and you can use the Split Regions at Playhead key command to cut all the Regions in that Group. Once all your tracks are sliced as required you can group them to make sure they stay together when you move or quantise them — but don't forget to add the guide Regions to the Group as well.

Update: Logic Pro v8.0.1

There seems to be a law of computing which states; 'for every new feature and bug fixed in a software release there will be a corresponding new bug added and a useful feature removed'. Logic 8's first 'point' release is apparently no exception to this rule. The good news is that Apple are obviously listening to their users — although, at first glance, the 'point-oh-one' appended to the update might fool you into thinking it isn't particularly significant, it actually contains some important changes. The outcry following the implementation of the fixed Transport bar obviously hit home, as there's now a function in the View menu of the Arrange page to turn it off. However, that which the Lords at Cupertino giveth with one hand they taketh away with the other. In adding this feature, Apple have introduced a bug that freezes the Transport bar display — though it can be unfrozen by changing Screensets or jiggling about with the Inspector display View and Restore parameters. Also broken in the latest release is the floating-window Piano Roll link function: it no longer reflects any change of MIDI Region when you change Screensets.

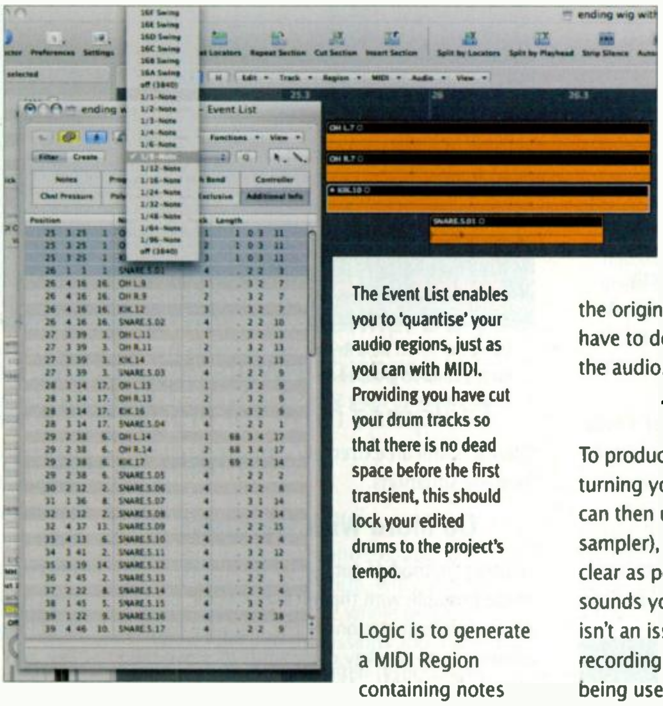
More usefully, Apple have reinstated the ability to cut and paste Audio configurations between Projects which was lost when they removed the old Audio configuration window — which, again, shows they are actually reading all those feedback emails. Frustratingly, though, they haven't also re-introduced the ability to cut and paste configurations between different audio drivers, so importing DAE sessions from Pro Tools is still not possible in Logic 8. It's hard to see why Apple have only gone halfway with this. Perhaps it's an indication that the removal of the Audio configuration window was a much more fundamental change to the program than we first thought. Apple have cunningly added some of the update information into the Logic 8 late-breaking news document (http://manuals.info.apple.com/en/Logic_Pro_8.0_lbn_z.pdf) without actually renaming it!

As far as performance is concerned, I haven't seen any changes on either of my systems. The G5 is still significantly more sluggish than the MacBook Pro. But others have reported both better and worse performance on seemingly similar systems. I think part of the problem is that we're still in a transitional period. People are using G4, G5 and Intel computers along with various revisions of Tiger and Leopard so it's hard to pinpoint where any real issues with Logic 8 lie. It's good that Apple seem to be paying attention to Logic users' issues — but it would be nice if the conversation wasn't always so one-sided!

Quantising Drum Tracks

Once you have the tracks split at the transients, you can either move the Regions to where you want them to be on the timeline using the usual copy and paste techniques, or you can quantise them by selecting them in the Arrange page and opening the Event editor from the main Window menu. Here, you can set the required value and then quantise the Regions as if they were MIDI data. Of course, quantising will only work if your recording has some kind of tempo map assigned to it. If it was played to a click you'll only need to know the bpm values and any tempo changes — which can then be inserted using the Tempo list in the Tempo section of the Options menu or the Tempo Global track.

If the recording was played 'free', you can try to create a tempo map using the techniques described in the *SOS* May 2006 article on tempo matching (on-line at www.soundonsound.com/sos/may06/articles/logictech_0506.htm).



The Event List enables you to 'quantise' your audio regions, just as you can with MIDI. Providing you have cut your drum tracks so that there is no dead space before the first transient, this should lock your edited drums to the project's tempo.

Drum Replacement

Of course, once you have sliced your beats, individual drum hits can be replaced with samples, a common trick that many producers and remixers use these days. There are some third-party drum-replacement solutions available for Logic, such as Drumagog (www.drumagog.com), but there's quite a lot you can do using the tools available within Logic itself. The aim with drum replacement in Logic is to generate a MIDI Region containing notes that mirror the desired transients in the recording. It's usually done with bass and snare drums, but I know a few brave souls who replace hi-hats too. To do this, we need to use Logic's Audio To Score feature, which is located in the Factory menu of the Sample editor. If you just open a recording and apply this process you're likely to be pretty disappointed. Like most of these analysis procedures, the old tenet 'garbage in, garbage out' applies, and you'll need to do some pre-processing before you'll get the results you want. I tend to make a copy of the drum parts I want to work on and paste them past the end of the Song, or into a new Project entirely. I do this by using the Export As Audio Files feature, which is accessed by right- or Control-clicking on the Region in the Arrange page. By doing this you can be sure you don't damage the original recordings, as it's likely you may have to do some destructive processing on the audio.

Transient Designer

To produce the best source material for turning your audio into MIDI data (which you can then use to trigger a drum instrument or sampler), you need to make the transients as clear as possible, while suppressing the sounds you don't want to hear. Audio quality isn't an issue; you're not going to use these recordings in the final mix; they are purely being used to generate data for the Audio to Score feature. For this example, I'm going to use a recording of a snare. As with most of my recordings, there's a lot of bleed from the other drums in the kit, so the first thing to do is use a noise gate. Logic's basic noise gate is perfect for this, as not only does it have controls for setting the threshold, attack and release parameters, it also allows you to set the frequency range to which the gate will respond. I sometimes place a flat EQ with the Analyser switched on, or the Multimeter plug-in, before the gate, so I can see which are

the dominant frequencies of the transients I'm interested in, and then insert these values. You can usually home in pretty accurately on the drum you're trying to isolate. If the beats vary a lot in dynamic range, you can add Logic's Compressor and Limiter to bring them all up to the same level. When you're happy you've isolated your beats, you can then Export the processed recording as an Audio file, by right- or Control-clicking on the Region and then re-importing the processed file back into the Project.

Audio To Score

Once the processed file is back on the Arrange page, double-click on the Region and open it in the Sample Editor. If you've been successful, the waveform should just show the transients you want, with near-silence between them. When Logic performs its MIDI note extraction it will place the Region generated on the currently selected track, which can cause a bit of head scratching if you're not expecting it. I usually create an Instrument track beneath the one the processed file is on and insert the plug-in that will play the replacement sound — in this example, I'm using Ultrabeat.

When you open the Audio To Score window from the Sample Editor's Factory menu you'll see some presets in a pull-down menu. These are a good place to start, but it's likely you'll need to try the process several times with different parameters to generate the results you want. The undo function, and trial and error, are invaluable here.

The Granulation parameter sets the time span of the louder parts and thus the transients. Start with values between 50 and 200 ms and use a short attack time (between four and 40 ms) and short release time (zero to five percent). Leave the Velocity threshold at '1' and set the Basis Quantise to the most sensible value for the material you're using — perhaps 1/4 for bass drum, for example.

Now click on Process. Logic will show a 'sample being processed' display, and a MIDI Region containing the extracted notes will be generated. If you double-click on it to open it in the Piano roll (formerly Matrix) editor, you'll see the individual notes. If the Audio To Score's settings were good, you'll just have notes that fall at the same position as the transients in the original recording. Of course, it's more likely that some spurious data will also have been generated, so you can either edit it out manually or click undo and try changing the parameters.

You'll probably have to drag all the hits to the correct note on which your replacement sample is assigned. To do this, use Select All from the Edit menu and drag the whole lot to the correct note. If you try playing the generated MIDI data and the processed audio together you'll quickly hear and see if Logic has added any notes you don't need. Once you're happy with the MIDI Region, you can copy it to the correct place in the timeline of the original Project. Have a listen to the unprocessed audio recording and the MIDI Region together to make sure they sound satisfactory.

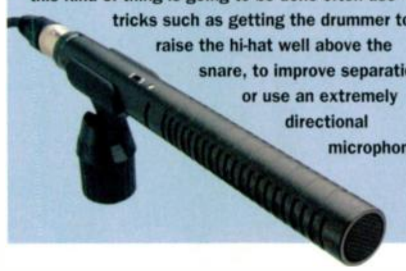
Conclusion

Drum editing and replacement can either be used as creative tools or for improving a recording. The purists may complain, but the listening public are becoming less tolerant of timing errors in their recordings, and even acoustic music can benefit from a bit of editing. Saying this, I wouldn't like to be the person who performs drum replacement or editing on any upcoming remixes of classic Led Zeppelin tunes! SOS

Tricks Of The Trade

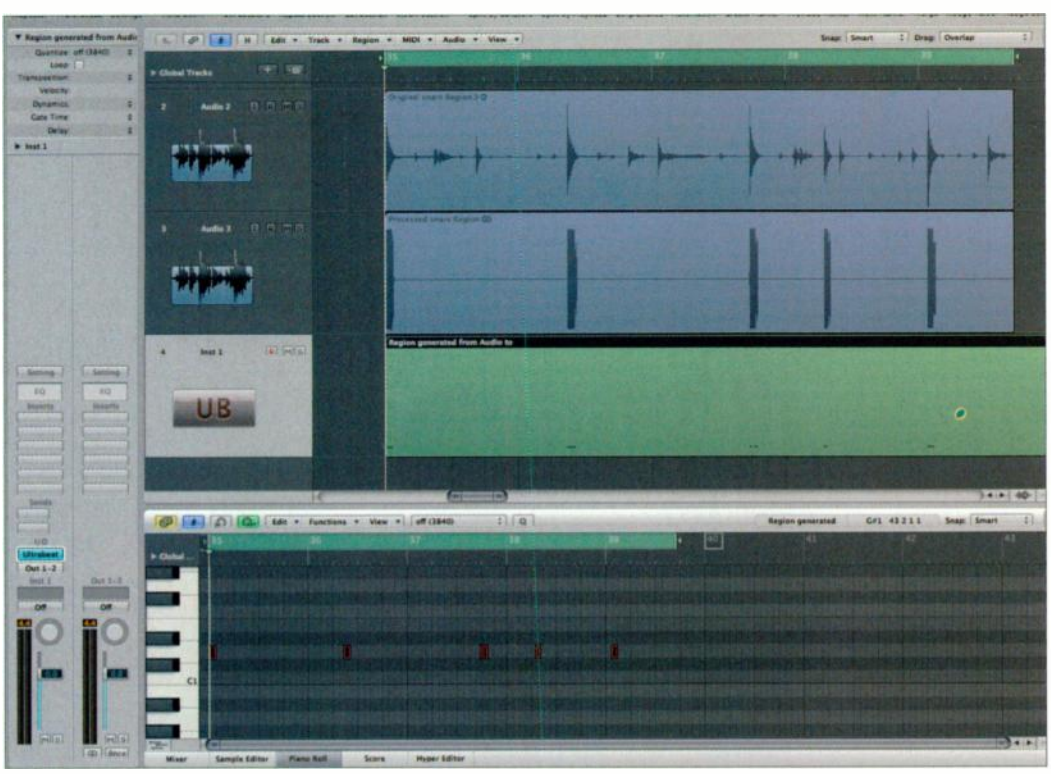
No matter how well recorded a drum track is, you may still want to replace or add another sound in parallel using samples. How easy this is to do depends on the recording. If you are presented with recordings where the snare, bass drum and hi-hat are perfectly isolated, you're not going to have many problems.

In fact, producers who know in advance that this kind of thing is going to be done often use tricks such as getting the drummer to raise the hi-hat well above the snare, to improve separation, or use an extremely directional microphone such as the shotgun type that's a staple in location film work, just to isolate the sound and generate a recording that will be used solely for replacement duties. In this case, the sound quality doesn't matter; only the transients are required.



A colleague of mine uses contact microphones taped to the snare, bass drum and toms to generate a click whenever a drum is hit. He then uses these recordings if he needs to replace the drum sounds. As is so often the case, the more preparation you do beforehand, the easier it is to achieve the results you want later.

However, most recordings will suffer from bleed of one kind or another, which makes the whole process more complicated.



Drum replacement: this screenshot shows the original audio file for the snare track (top), the file that was created to highlight only the transients (middle), and the MIDI region that was created after using Logic's Audio to Score feature which, in this example, is being used to trigger Ultrabeat. The Piano Roll at the bottom of the image shows the MIDI data in more detail.

Three approaches to input monitoring: an Input Monitor-enabled audio track handles the signal from Reason 4 (above), while an aux track is set up to receive the return from a Yamaha REV500 effects unit. Meanwhile, MOTU's CueMix Console (lurking bottom left) provides a zero-latency monitor of a Korg Radias synth.

Robin Bigwood

Listening In

Input Monitoring in Digital Performer

There's more useful stuff than you can shake a stick at this month, including advice on input monitoring, ways to get around latency issues, and news of CD-burning and audio networking applications to make your life easier.

Over the last couple of months we've been looking at monitoring: what it is, and how to make it work for you in Digital Performer with a range of typical studio situations and hardware setups. In this final instalment of our monitoring extravaganza, DP5's Input Monitor function and aux tracks come under the spotlight, and I offer suggestions for incorporating hardware effects units into your DP-based rig.

Not Just For Recording

You could be forgiven for thinking that monitoring is only relevant for recording — specifically, for supplying a headphone backing-track mix to an artist recording a performance. But it encompasses more than this, especially in the modern studio integrating hardware and software synths and effects.

To illustrate what I mean, imagine Reason and DP are running on the same Mac. The Rewire link between the two applications carries audio from Reason to DP, but in order to hear Reason at all some sort of 'open input' is needed in DP. An aux track is often perfect for this, bringing the Reason audio into DP's Mixing Board, and allowing it to be mixed alongside other track types. What the aux track is really doing is monitoring the live signal from Reason, allowing the engineer to hear it while working on other aspects of the mix, just as a singer would hear a backing track as they recorded vocals.

These 'open inputs' are useful for all sorts of things, not just Reason. You could use them to monitor the return from a hardware effects unit, the signal from a hardware synth, or a signal from another Mac arriving digitally via an ADAT connection or across a network. All these need to be monitored as you work on your track, until you choose to record them and capture their signals to an audio track. So what are the options available to you?

Aux tracks: Aux tracks are a long-standing DP feature. You can't record on them; instead they're a lot like mixing desk channels. They can be fed with inputs from your audio interface, DP's internal busses, or the Rewire signal from other software — all of which is useful from a monitoring point of view. What's great about them, too, is that their operation is not affected by DP5's Audio Patch Thru mode — you could have it set to Off and they'd still work. Also, if you have a MOTU audio interface and are using Direct Hardware Playthrough mode, they still allow you to monitor your input through any effects plug-ins instantiated.

Audio tracks: You've always been able to monitor input signals through audio tracks, courtesy of DP's Audio Patch Thru feature, by simply record-enabling them. But in recent versions of DP you have more flexibility, thanks to the Input Monitor function. This means that an audio track can be made to permanently 'patch thru' its input to its output, whether the track is record-enabled or not. Just click the track's Input Monitor button in the Tracks Overview (the 'Mon' column), Sequence Editor (a loudspeaker icon in the track's info pane) or Mixing Board (the Input button next to Rec, Solo and Mute). An Input Monitor-enabled audio track is not quite the same as an aux track, though. First, it does respect DP5's Audio Patch Thru mode, so if that's set to Off the Input Monitor function is essentially disabled. Also, if you're using Direct Hardware Playthrough mode with a compatible MOTU audio interface, Input Monitor will not run the signal through DP (or any effects plug-ins on the track), but instead set up a temporary CueMix zero-latency routing.

Zero-latency hardware monitoring: There's nothing to say we have to control all monitor signals from external hardware with DP. In many cases the best approach is to monitor external effects returns, and especially hardware synths, via a hardware mixer or an audio interface with zero-latency monitoring, as you might when setting up monitoring for a vocal take. The external hardware signals can be incorporated into your control room mix independently of DP, until you want to record them into your track. At this point you just record-enable some audio tracks in DP and route them in.

A Real-world Example

Here's a situation I've mocked up that incorporates all three types of input monitoring, using my own setup of a Power Mac G5 and MOTU Traveler (which has CueMix zero-latency monitoring). I'm running DP 5.13 and Reason 4, and also have some external hardware: a Korg Radias synth and a Yamaha REV500 reverb. This is how everything's co-ordinated (see screen at start of article):

Input: Reason software
Monitoring Method: Input Monitor-enabled audio track.
Description: As it's providing some synth and percussion parts, I want to monitor Reason constantly as I work on my song. Using an audio track with Input Monitor allows me to do this, and when I'm nearing completion I'll record-enable it to record the signals as audio in DP

Input: Radias synth
Monitoring Method: CueMix
Description: As I'm using the Radias to provide the basis of my arrangement, including bass and key rhythmic parts, I want to hear it completely free of latency. Hence I've set up hardware monitoring in the CueMix Console: the Radias signal is not coming into DP at all. But when I've finished my song, I'll record the separate parts into DP on audio tracks for the final mix and any further treatment.

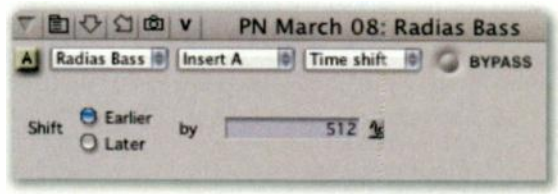
Input: REV500 hardware reverb
Monitoring Method: Aux track
Description: The REV500 is patched into my setup so that it's fed by one of the Traveler's outputs (with a corresponding Aux send in DP) and returns back into a couple of inputs. I'm bringing its signal into an Aux track so I can further treat it with a MAS plug-in. I suffer some latency because of the round-trip out of DP and the Traveler and back into DP, but it's not a problem, as it simply becomes a bit of additional reverb pre-delay. If I need to record the REV500's signal later, I could route the aux track via a bus pair to a record-enabled audio track.

This all works great for me as I'm developing my song. For Audio Patch Thru I'm using the Blend mode, and I've set a 512 buffer size. The mix of hardware and software monitoring presents no problems here. But it's not always so easy...

Dealing With Latency

As we've seen over the past few months, latency can be a problem with certain approaches to monitoring. Musicians generally hate it if the headphone monitor mix of their live performance is anything more than a few milliseconds late — which is why zero-latency solutions like suitably-equipped interfaces and hardware mixers are routinely used for monitoring. But what if you don't have access to one of these, or if, for other reasons, you must monitor your external gear through DP and don't want latency?

In the case of monitoring synths and samplers driven by MIDI, DP automatically sorts out latency issues for software synths it hosts, or those coming into an aux or audio track via Rewire. So even if you use a large buffer size, such as 1024, playback of DP-hosted or Rewire-connected synths will be perfectly in time. But you'll still need to use a small buffer size to get a crisp response when playing them live from your controller keyboard. The same goes for hardware synths monitored in DP, but once their MIDI tracks are in place you can compensate for the latency associated with switching back to a large buffer size by making the MIDI play early. Just put a Time Shift plug-in on each MIDI track and set the track to play early by an appropriate amount. You can start by defining it in samples, to match your buffer size, before fine-tuning further.

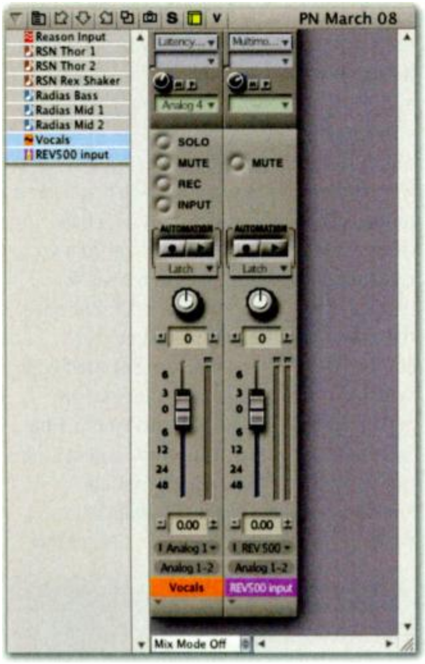


The Time Shift plug-in can help to compensate for monitoring latency with synths driven from MIDI tracks.

It seems such a nice idea, to have your favourite hardware compressor or reverb unit patched into your audio interface, ready to be addressed via an aux send on your audio tracks, and yet latency often spoils the party. In my earlier example, where I'm using my REV500 for some reverb, I can take a bit of latency on its signal because it comes out sounding like pre-delay on the reverb. But for a true processor treatment, like EQ or compression (or if I just don't want any pre-delay on a reverb) any amount of latency is completely unacceptable. Even a few milliseconds could mess up the musical timing of the mix or produce nasty phasing in some circumstances. And a zero-latency monitoring scheme doesn't fix it, because there's latency inherent in the signal passing out of DP. What's needed is some sort of latency compensation.

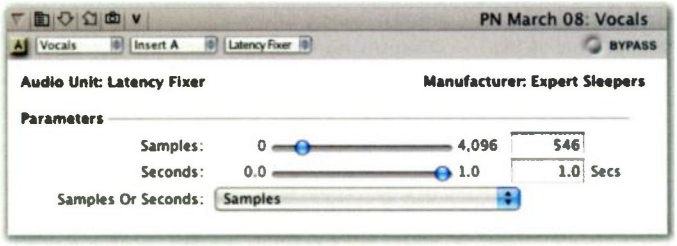
Now, if you're thinking DP5 has built-in latency compensation you're right, but it's only for hosted plug-ins, not external routing. However, a freeware Audio Unit plug-in does exist that does exactly what's needed for external routing latency compensation: it's Latency Fixer, from www.collective.co.uk/expertsleepers and it's a nifty little thing. It works by first reporting a latency (which, using the controls, you set manually in terms of seconds or samples) to DP. DP then compensates for this latency by sending track audio to the plug-in ahead of time, but the plug-in actually applies no

INPUT MONITORING



I incorporate my REV500 reverb unit into my DP mixes by having it patched into my Traveler interface. I can then route track signals to it via an aux send (as on the Vocals track here), and I also need an aux track to handle its signal coming back into DP. Using the Latency Fixer AU plug-in I can completely eradicate the latency that would otherwise occur in this arrangement.

delay at all to the audio passing through it. Consequently, if you place it on an aux track that's being used to feed your audio tracks to an external effects unit, the latency accumulated in the trip out of DP and back in again can be precisely compensated for — after a bit of experimentation. The screens above show how I use it to route audio



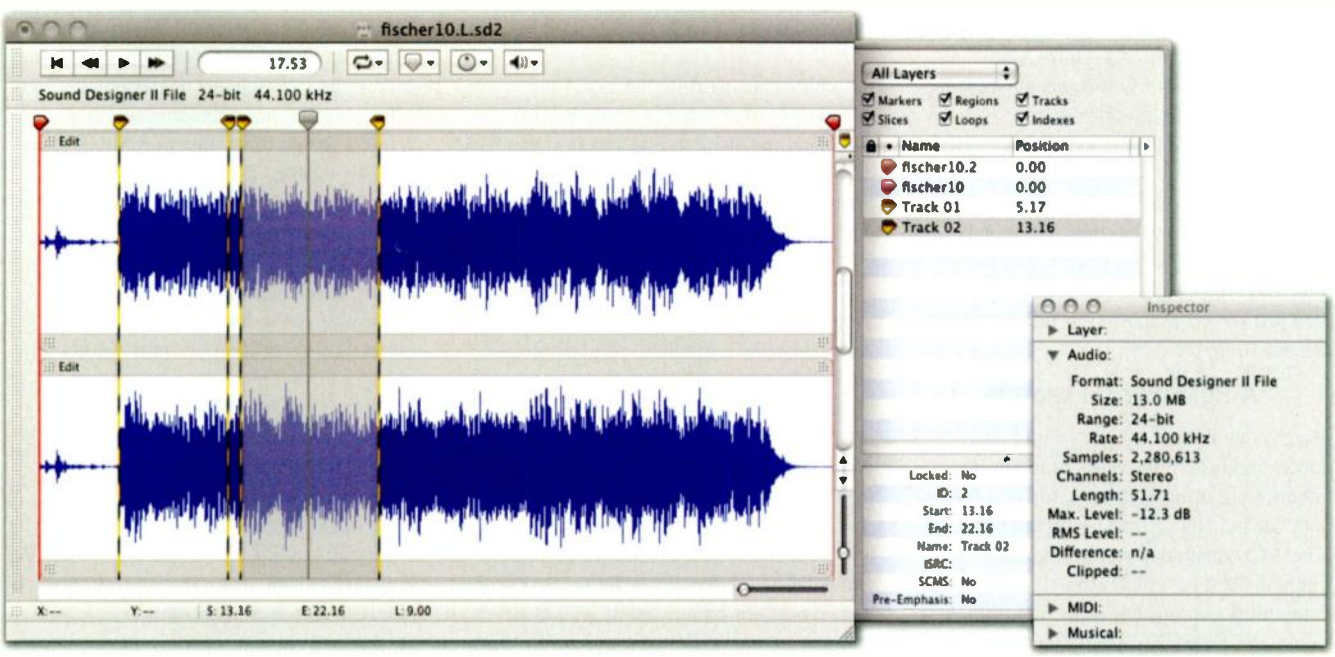
tracks to my external hardware compressor, the output of which comes right back into my DP Mixing Board.

DP To CD?

In the August and September 2007 Performer workshops I looked at some ways in which DP can be used to prepare an album-length project for CD burning. DP's multitrack audio capabilities, flexible effects and automation make it very good for this task, even though it doesn't have any features specifically designed for it. DP can't do CD-burning, so you have to transfer the resulting audio files to another application, and it's here that you can have difficulties. For example, the applications that can work with DP's native stereo audio format, Split Sound Designer II, are either discontinued (Roxio's Jam), not available separately (Apple WaveBurner), don't read region information (i3 DSP Quattro) or are very expensive (Bias Peak Pro and Sonic Studio PreMaster CD). Wouldn't it be great to have an audio editor that could load any audio format, including SD2, correctly read region information, and offer heavyweight editing, dithering, export and burning options? Now, in the form of an updated Wave Editor by Audiofile Engineering, you can.

Wave Editor 1.3 is a thoroughly up-to-date application, utilising OS X's Core Audio features and presenting a slick, customisable user interface. Its really unique feature is a multi-layer (as opposed to multi-track) approach to editing, whereby you can assemble on a time-line different sections of audio at any sample rate and resolution, applying fades, crossfades and other processing on or between each layer. When the time comes for burning a CD or exporting your audio to another format, Wave Editor treats any layers at differing audio specs with iZotope's highly-regarded SRC sample-rate conversion and MBIT+ dithering. For DP users (and others) this approach borders on the revelatory — you'll almost never need to think about dithering your 24-bit projects for CD again, and if you're compiling mixes done at different sample rates you can let Wave Editor deal with that too.

If you like to burn your CDs from multi-region audio prepared in DP (as described in the September 2007 column), Wave Editor (below) offers a straightforward workflow. After opening your multi-region SD2 files, make sure 'Regions' is ticked in Wave Editor's Waveform menu. Then, in the Labels drawer, select all your audio's region labels and right-click (or control-click). Choose Convert To / Tracks, and you instantly have a burn-ready document, with CD track boundaries where your DP region



It ain't all that pretty, but Wave Editor is an affordable audio editor and CD-burning application that complements DP beautifully, and includes heavyweight features like MBIT+ dithering and Disc Description Protocol (DDP) export.

Who's Recommending Soundfly?

Writing the monthly Performer workshop can occasionally feel like something out of a Dickens novel; long, dark evenings spent with just the plaintive chirps of my G5 Power Mac for company, and only a red-hot Macbook for warmth. So it's always nice to get some feedback from readers, and I was especially pleased to hear from Pete Townshend (of the Who fame) recently. He very helpfully drew my attention to a little freeware application I hadn't come across, which might be of interest if you're using (as Pete does) a studio network of Macs to offload various processing and virtual instrument duties. It's called Soundfly and comes from Abyssoft, the company that also makes the superb Teleport 'mouse and keyboard sharing across a network' software. Soundfly exists to send audio from one Mac to another, across an Ethernet, Airport or other network connection. It relies on Cycling 74's Soundflower inter-application audio utility, so you need that installed first. It also utilises two of the Audio Units built into OS X — AUNetSend and AUNetReceive — though you never interact with these directly.

In use, Soundfly is very straightforward. On the Mac from which you want to send audio — perhaps one running some stand-alone soft synths — you run Soundfly. On the Mac that needs to receive this audio you run Soundfly Receiver. Both applications are so simple that normally they don't even have a user interface. But you can force one to appear by holding down the Alt key as you launch each application, and I find it's useful to do. First off, you can configure the audio format used by Soundfly to broadcast across the network. A 'full monty' uncompressed PCM format is selectable, but in case your network can't cope with that much data throughput various compressed formats, like the dependable AAC, can be chosen instead. Some experimentation may be needed to find what works best for your network. I also found I needed to configure Soundfly Receiver on my G5 in order to manually make the connection with Soundfly running on the Macbook. This is straightforward: just select the audio stream from Soundfly in the directory list, and click Connect.

Soundfly won't give you a multi-channel audio connection between your Macs, only a stereo one. Also, the Soundfly connection can't be brought directly into DP's mixing environment on the receiving Mac. But it's still a useful little thing, and across a wired network that isn't bogged down with other traffic it can operate with low latency and dependable audio quality. It can be downloaded free, with the option of making a donation, at www.abyssoft.com. Don't forget to install Soundflower, from www.cycling74.com/products/soundflower, if you don't have it.

boundaries were. What's more, everything a mastering application should be able to edit — CD-TEXT metadata, ISRC and UPC/EAN codes, DDP annotation and PQ subcodes — can all be edited, either by clicking on individual tracks in the Labels drawer, or entering data in the Properties palette. You finally burn your CD by choosing Burn Disc in the File menu.

Wave Editor might not have the instant user-friendliness of consumer-level applications like Toast and Jam, and a read through its PDF manual is a must for the first-time user. But it's hard to imagine a more powerful, flexible and useful audio editor — certainly not one that dovetails so well with DP, and is so affordable. Wave Editor comes as a download from www.audiofile-engineering.com, and costs $250 (but only $200 until 31st March). If you're eligible for educational pricing it's only $100, and a $150 crossgrade is available for owners of most other major audio editors. SOS

Conquering Command

Getting The Best From Digidesign's Command 8

Small and affordable it may be, but Digidesign's Command 8 is also a surprisingly versatile and powerful tool for controlling Pro Tools.

Mike Thornton

I am sure most of those in need of a cost-effective, moving-fader control surface for Pro Tools would immediately think of Digidesign's Command 8 (around £900). This is certainly what I did: I have one beside me now, and most of the time the faders are what I use it for, either when I want to 'feel' a mix or when I need to move more than one fader at a time. It is an approved, supported control surface that won't break the bank. However, since I got it I've discovered that the Command 8 can do much more than merely control channel levels in Pro Tools, so this month we'll be looking at the additional functionality and features that this control surface implements.

A Control Surface With A View

To start with, there are three view modes that can be selected on your Command 8. Home View is the default mode, and is what you will see once you have opened a Session in Pro Tools. In Home View, the track names are displayed on the bottom row of the display and the rotary encoders are set to the Pan settings, which is indicated by the 'Pan L' display on the top row.

When you hit one of the Console View Selectors on the left-hand side of the Command 8, the unit goes into Console View. This lets you see pan position, send assignments and plug-in assignments for all the channels the control surface has access to. When you adjust either the fader or the rotary encoder on a channel, the name is temporarily replaced in the display by the appropriate data value — for instance, the dB setting for the fader or the relative position of the pan controller.

With Pan selected, all the rotary encoders will display and control the pan settings for all the tracks viewed on the Command 8; this is the default setting in Console View. For stereo tracks this mode will display Pan L, and if you want to access the Pan R setting you will need to press the Pan/Meter button, which is just under the Control monitor section. Note also that pressing a track's Select button in this mode will select the corresponding track in the Edit and Mix windows, making it easy to do multiple track selection.

When you press the Send button, the Command 8's rotary encoders display the last send selected on each track. Using the Console View buttons A to E enables you to select and view the respective send assignments; to access sends F to J, you

Console View displays a single parameter, such as Send A, for each of the eight channels currently in focus.

need to hold down the Shift button. Then by pressing button A you access send F, and so on. The top row of the display shows 'Send x — All channels' and the bottom row displays the send label. In this mode, the channel's Select button toggles the track's

If you have the sends in Pro Tools' Mix window set up to display the level and pan sliders, pressing a Command 8 Send button in Console View will switch to the appropriate send in the Mix window.

send between pre-fade and post-fade. Note also that if you have your Send View in the Mix window set to show the fader and routing for a particular send, as you press the Console View buttons, the Mix window will display the corresponding send view. However, this doesn't work if your Mix window is showing Assignments.

Pressing the Insert button sets the Console View A to E buttons to enable you to view the appropriate insert on the display. If a track has a plug-in on that insert, the name is shown on the bottom row of the display and the track's Select button lights. When you press the Select button, the display shows the controls for that plug-in; if there are more than eight controls, the Command 8 displays them in pages. I dislike rotary controls on plug-in windows, as they're hard to use with mice, so it's great to have physical rotary controls to adjust the virtual ones. Also, as with the faders, you can control more than one thing at the same time.

When you press the Display Mode button, all the text names on the Command 8 screen are temporarily replaced by the settings values. This works for faders in Home View, send levels in Console View Sends mode, and control settings in Insert mode. There's also a Mon/˜ button, which enables you to toggle globally between Auto Input and Input Only on record-enabled tracks.

Channel View

The third Command 8 View mode focuses all the unit's display and controls on one single track, showing all the plug-in, pan, send and insert parameters horizontally across the display. To access the various sections of Channel View you use the eight buttons immediately under the display.

When you press the EQ button, the Command 8 will identify any track with an EQ plug-in (in other words, any plug-in that ends up in the EQ section of the plug-in menu) by flashing the track title in the display and lighting up the respective track's Select button. If a track doesn't have an EQ plug-in assigned to it, the display is left blank and the Select button stays unlit. When you press a lit Select button, the controls for the first EQ plug-in on that track will be assigned to the rotary encoders and shown in the display. The relevant Channel Select buttons control any switched functions, such as Master Bypass. Likewise, the Dynamics button prompts the Command 8 to show any tracks that have compressors, limiters, gates and so on inserted.

Pressing the Inserts button will identify any track that has a plug-in or hardware insert assigned to it. Pressing a lit track Select button will show the names of all the inserts on that track. Now for each of the insert points containing plug-ins the appropriate Select button will light. Pressing one of those Select button will display the first page of that plug-in's settings and assign the rotary encoders accordingly. Remember that any hardware insert won't display settings, because it hasn't got any!

The Pan/Send/Pre button identifies any tracks that have sends assigned to them. When you press the desired Select button the first rotary encoder displays the pan control for that track. Encoder 2 is unused, for some reason. I assumed that on a stereo track it would be the right channel's pan control, but annoyingly, this is not the case — you need to press the Pan/Meter button ▶

Troubleshooting & Testing

I include this advice to help anyone with troublesome moving faders. Most of it will apply to any moving-fader control surface, not just the Command 8. If you are experiencing 'sticky' faders that fight you or don't react correctly to your touch, but respond to automation within Pro Tools, try working through the following.

Touch-sensitive faders rely on the capacitance of the human body relative to ground. Human capacitance is different between individuals, and can be changed by our immediate environment. For example, a touch-sensitive fader may react differently to a person with dry hands than one with moist hands, since the moisture will conduct better during the process of attaching you to the circuit. Heavy carpets, socks and soft-soled shoes can further isolate the operator from earth, affecting the ability of the circuit to effectively sense your capacitance to ground. Try using the controller while standing, as a comfortable chair may also be isolating you from ground.

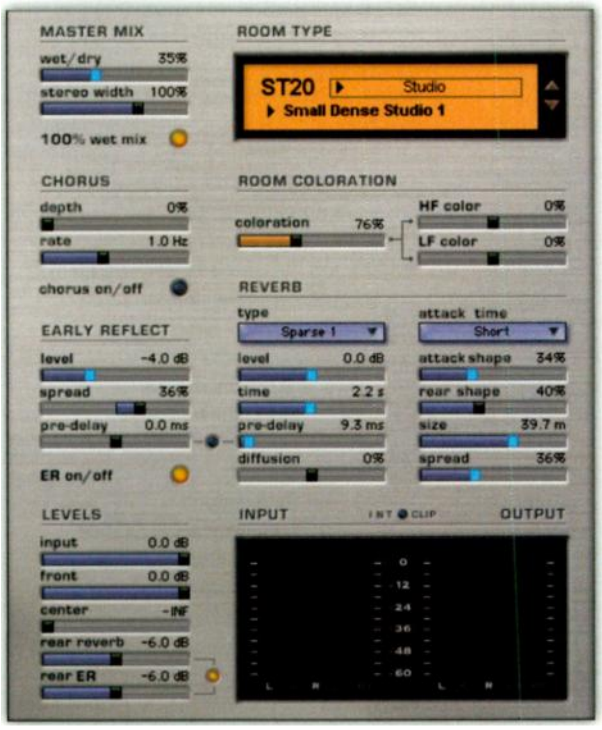
Sometimes the metal paint coating under the fader cap which makes contact with the glide on the fader can peel, causing intermittent contact. Consider checking the underside of the fader cap for missing paint. You can get replacement caps from Digidesign dealers or via the DigiStore on-line.

Dust can also affect the responsiveness of the faders; try blowing out the fader track with an aerosol canister of compressed air.

The utility functions available by hitting F1 include various fader tests which may help to narrow down problems. Triangle, Sine, Step and Cycle tests will move the faders in the selected pattern up and down the fader throw, while the Group test allows you to move all faders together by only touching one. In these tests, the display will report which step-bit the fader is currently on. There are a total of 1024 steps (0-1023), as the faders have a 10-bit resolution. If you have faders that don't seem to travel the entire throw, or seem as though their range is limited, you should recalibrate the faders by running the Fader Cal test.

Getting back to fader caps, the Touch test will show you the current capacitance of the fader itself, when idle and when touched. Because the grounding in the environment and the capacitance of the operator vary, the absolute value displayed is not crucial: the thing to look for is whether or not the 'touch' is registered.

Most Pro Tools plug-ins are set up to assign their eight most commonly used parameters to the first page of Command 8 controls.



The eight parameters assigned automatically to the Command 8's first page won't necessarily be the ones that appear first in a plug-in window, as you can see here. The light-blue squares denote assigned parameters.

to set encoder 1 to right instead of left. Why leave encoder 2 idle when they could have made it possible to adjust both pan controls on a stereo track simultaneously? Encoders 3 to 7, meanwhile, are assigned to Sends A to E; switching to the second page changes their focus to Sends F to J. In this mode the Select buttons again serve as pre/post switches.

The Page left and right arrow buttons enable you to scroll through the plug-in parameters, using as many pages as required to cover all the parameters for that plug-in. As you scroll through the pages, the plug-in window shows which controls are assigned to the encoders; you will see that they don't always follow in their on-screen order. The first page is usually carefully selected to offer the most commonly used parameters.

The Access Virus Indigo plug-in provides an excellent example of how well plug-in adjustment works from the Command 8. When you load it up, you're presented with 40 pages of parameters to go at, but the first page gives you the eight most commonly used from the plug-in's Easy page. In the screenshot opposite you can see that the controllers assigned to the Command 8 encoders are identified by the light-blue background in the text field. Now the great thing is that if you have the plug-in window open as you navigate around the pages on the Command 8, the different pages of the plug-in window open and show which parameters are assigned; for example, page 12 of 40 on the Command 8 opens the Filter/Env page on the plug-in, and you can see the parameters that are currently assigned. If you then use the Command 8's Flip function, you bring the plug-in parameters down onto the touch-sensitive faders, and once you have enabled automation on the plug-in (don't forget the neat shortcut of holding down all three modifier keys and clicking on a parameter to automate that parameter, without having to open the plug-in's automation window), you can now easily record the automation changes of any of your chosen parameters.

The Master Bypass button will put the selected plug-in into bypass if you are viewing an individual plug-in. However, if you are displaying multiple plug-ins, as you are when you have pressed Insert in Channel View, then using the Master Bypass button will put all that track's plug-ins into Bypass. That is a neat trick you can't do from the QWERTY keyboard. Likewise, because you can open any plug-in from the Command 8, you can control a plug-in that isn't targeted on screen — you don't even need to have its window open. Again, this is something you can't do from the mouse and keyboard; you have to target the plug-in you want to adjust and then make the adjustments.

What You See Is What You Get

One of the neat features of the Command 8 is the way you can control what appears on the Pro Tools screen. For example, the three buttons grouped together above the Transport section enable you to switch between the Edit, Mix and Plug-in windows. The Plug-in button is especially useful: for example, once you have selected a plug-in via Console or Channel View, pressing the Plug-in Window button will open the window for that plug-in on the screen, which is helpful as you navigate through the different parameters. The four multi-function arrow buttons in the navigation section also help you get around the Pro Tools interface. When the Zoom button is enabled, the left and right arrows zoom the Edit window timeline in and out, and the up and down buttons increase and

Command 8 Display Conventions

The Command 8 display is limited to six characters per channel, so Pro Tools makes an 'intelligent' truncation of the track, send and insert names, but these can be confusing. For example, as I record a lot using an M+S pair, I often have a Waves S1 Imager to decode the M+S pair, However, the Command 8 displays that as 's1im=2' which isn't the most obvious abbreviation! Others are clearer — Waves Renaissance Compressor comes out as 'rcmprs' — but the problem isn't limited to Waves plug-ins. Digidesign's Revibe comes out as 'revb=2' and their Reverb One as 'rvro=2'.

If you're puzzled by the '=2', it indicates that a plug-in is stereo — a 5.1 Revibe is displayed as 'revb=5'. Personally, I would rather Digi used those two characters to make the abbreviated name clearer. There also seems to be no logic to the use of upper and lower-case characters for the control and plug-in names: some plug-ins use only lower case, some only capitals and others some of both.

Access's Virus Indigo plug-in: although there are 40 pages of controls, it's still straightforward to navigate from the Command 8.

decrease the track waveform height. In Bank and Nudge modes, by contrast, the up and down buttons mirror those on the keyboard, so they can be used to mark in and out points on the fly.

Functional Stuff

The five Function keys at the right-hand side of the Navigation and Zoom section are oft-neglected, but can be useful. The first of them puts the Command 8 into Utility mode, where you have access to a range of information and test procedures. The first press displays the Command 8's firmware version, which should read v02.01.12. If you suffer from over-sensitive faders, or faders that sit there and move very slightly on their own, make sure you have this version of firmware installed.

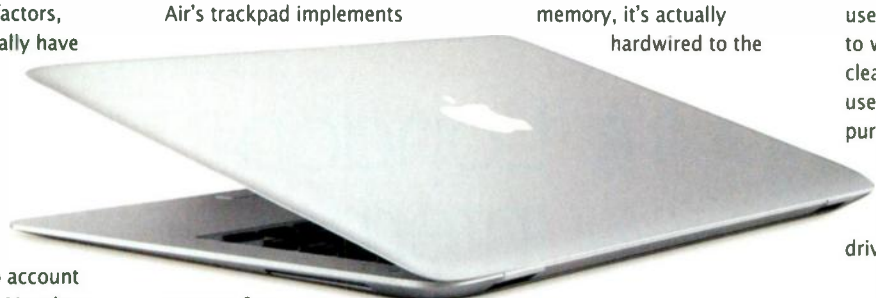
F2 lets you edit and name custom MIDI maps when using your Command 8 in 'stand-alone' mode, F3 provides control over an attached Digidesign Pre mic preamp, and F4 temporarily stops the moving faders from moving. If you are working in an environment where the mechanical noise of the faders moving is a problem, this can be a life-saver. Finally, F5 is called the Focus button, and toggles the LCD display between the current plug-in view and the previous Command 8 Console or Channel View, which is very useful when moving around the 'mixer' on the Command 8. SOS

As weeks go, the first couple in January were pretty good for new Mac hardware, with Apple introducing updated Mac Pros and Xserves, along with a new stunningly thin MacBook. We dissect the potential of Apple's new offerings with a musician-shaped scalpel.

Mark Wherry

January is often a good month for Mac users, with the annual MacWorld show taking place in San Francisco, and this January was no exception. As anticipated, the highlight of Steve Jobs' opening keynote was the introduction of the MacBook Air, which, with a height of between 0.16 and 0.76 inches (0.4 and 1.94cm), is being touted by Apple as "the world's thinnest notebook."

In addition to being incredibly thin, the MacBook Air is also the first Mac portable to feature innovations such as enhanced, iPhone-like multi-touch gestures on the built-in trackpad, and the option of using a solid-state drive instead of a hard disk. However, in order to create such ultra-mobile form-factors, manufacturers usually have to compromise on performance. And since performance is usually one of the most critical factors musicians and audio engineers take into account when purchasing a Mac, how suitable will this new MacBook be for demanding applications?



All I Need Is The Air...

Compared to the processors other laptop manufacturers have chosen for similar products, the MacBook Air's processing abilities are quite impressive, featuring a choice of either a 1.6 or 1.8 GHz Core 2 Duo processor from Intel, which offers a 4MB Level-2 cache and runs on an 800MHz system bus, just like the regular MacBook or MacBook Pro. This chip was apparently custom-built for the MacBook Air, and Intel CEO Paul Otellini made another MacWorld appearance to present one of these processors, which, judging by the size of Jobs' thumb and forefinger, is a remarkable exercise in miniaturisation.

The MacBook Air also features a 13.3-inch display with 1280 x 800 resolution, just like the MacBook, except that the screen on MacBook Air uses LED backlighting, and Jobs was keen to point out the environmental progress Apple has made in manufacturing the laptop. Unlike many competing ultra-mobile laptops, the MacBook Air offers a full-size keyboard that looks like the MacBook's keyboard, but also features backlighting, similar to the MacBook Pro.

As mentioned before, the Air's trackpad implements support for additional multi-touch gestures, allowing you to perform actions with multiple fingers to carry out certain operations. For example, where previous MacBooks (and PowerBooks) allowed two-finger scrolling, the Air's trackpad also allows you to use gestures such as pinching and expanding two fingers to zoom in and out, and swiping three fingers across the trackpad to either move backwards or forwards.

Although applications like iPhoto already support these gestures, because I haven't been able to get my hands on a MacBook Air at the time of writing I wasn't able to see if an application needs to implement support directly for these new gestures, or if such operations are provided 'for free' by the operating system.

In addition to the thinness, the Air name also refers to the networking abilities of the notebook, which features the same Airport Extreme 802.11n wireless networking as the MacBook and MacBook Pro, plus upgraded Bluetooth 2.1+EDR (Enhanced Data Rate) support.

Take A Deep Breath

While the MacBook Air undoubtedly has some great features, making a laptop as thin and as light as this — it weighs just three pounds (1.36kg) — inevitably involves some compromises. To begin with, although the MacBook Air ships with 2GB of 667MHz DDR2 memory, it's actually hardwired to the motherboard and there are no slots for adding additional memory. This won't be a big deal if you intend to use your MacBook Air for general computing tasks, but it's certainly less than ideal if you like to run a large number of software instruments.

It's also worth bearing in mind that although the 1.6GHz or 1.8GHz processor is impressive, the regular MacBook offers either a 2.0GHz or 2.2GHz Core 2 Duo processor, while the MacBook Pro has either a 2.2, 2.4 or 2.6 GHz Core 2 Duo chip. And unlike the MacBook Pro (but like the MacBook), the MacBook Air uses Intel's GMA X3100 graphics processor, that shares 144MB of main memory.

Moving on to storage, the MacBook Air is the first Mac portable in a number of years (since the Duo, if I remember correctly) that doesn't feature some kind of removable media, such as an optical drive. For me, this is perhaps the least significant limitation of the Air, since I rarely use the optical drive on my MacBook Pro, as it's usually connected to some kind of network. This is clearly the type of working practice Apple hope will appeal to MacBook Air users, although when it comes to watching DVDs, Apple also clearly hope that MacBook Air users will want to instead purchase or rent movies from the iTunes store.

To compensate for the lack of a built-in optical drive, Apple do offer an optional external Superdrive for

Photos courtesy of Apple

The MacBook Air exudes Apple's design elegance and has some incredible features for a computer that's almost impossibly thin. Unfortunately, with its lack of connectivity and limited storage option, along with the fact that you're reading *Sound On Sound*, this may not be the laptop for you.

New Mac Pros: More For Musicians

Although not featured in Jobs' MacWorld keynote presentation, Apple unveiled an update to their line-up of Mac Pros, which now feature Intel's latest 5400-series 'Harpertown' Xeon processors, and officially support SAS (Serial Attached SCSI) drives, which can be purchased as build-to-order options along with the required RAID card. These features were discussed in last month's Apple Notes column, along with performance benchmarks for running audio tracks on SAS drives via an Xserve (www.soundonsound.com/sos/feb08/articles/applenotes_0208.htm).

With the revision, the base price for a Mac Pro has now increased slightly, to £1749, and this 2.8GHz model now features eight cores (using dual quad-core Xeon processors) as standard, although a single-processor 2.8GHz four-core model is also available for £320 less. If you want the ultimate performance, you can opt for either 3.0 or 3.2GHz Xeon processors as well, although this adds £500 or £1000, respectively, to the price.

Each processor offers 12MB of Level-2 cache, and the Mac Pro now has a 1600MHz system buss, along with the ability to accommodate up to 32GB of 800MHz DDR2 ECC fully buffered memory. For graphics, the base Mac Pro features an ATI Radeon HD 2600 XT graphics card with 256MB memory and support for two dual-link DVI connections, and you can also order your Mac Pro with multiple ATI cards, or either an Nvidia GeForce 8800 GT with 512MB memory or an Nvidia Quadro FX 5600 with a staggering 1.5GB of memory.

While the base model comes with a single 320GB 7200rpm SATA drive, you can also order 500GB, 750GB or 1TB SATA drives as well; and for the ultimate in storage performance you can now order 300GB 15,000rpm SAS drives, which also require the optional RAID card. The Mac Pro accommodates up to four drives, as before, although they have to be either all SATA drives or all SAS drives, so you can't mix and match types.

We'll look more closely at the new Mac Pro in a future Apple Notes, but this looks like a highly desirable Mac for those who demand the ultimate in performance. The only thing to bear in mind right now is that the new Mac Pros are currently unsupported by Digidesign, partly because they ship with Leopard, with which Pro Tools is currently incompatible, pre-installed.

Apple's new Mac Pro incorporates Intel's latest 5400-series quad-core Xeons and now offers internal SAS drives. If you want the ultimate in performance, this could well be the Mac you've been waiting for.

£65, and the MacBook Air also ships with Apple's new Remote Disc software for either Mac OS X or Windows, which allows you to easily use the optical drive of another computer as if it was attached to your MacBook Air. While you can use Remote Disc to easily install software, there's no mention of whether disc burning is also supported. However, for backing up your files, Jobs also announced a new product called Time Capsule, which is basically an Airport Extreme base station with either a built-in 500GB or 1TB hard drive for £199 or £329 respectively. It works with Leopard's Time Machine feature, allowing you to wirelessly back up your files.

Perhaps the biggest potential performance bottleneck, though, is the 1.8-inch 4200rpm 80GB parallel ATA drive supplied with the basic MacBook Air model, which includes a 1.6GHz Core 2 Duo processor and retails for £1199. Having owned a laptop with such a drive in the past, I know that it really can slow a computer down, especially where virtual memory is concerned, and while it's not really fair to judge the MacBook Air on this, since I haven't seen an Air yet, I would find it hard to imagine this drive being suitable for any kind of audio work.

On the plus side, as mentioned earlier, one of the new features of the MacBook Air is the optional 64GB solid-state drive, which comes as standard with the higher-end model that has the 1.8GHz processor and costs £2028, or adds an additional £639 to the cost of the basic model. While this is expensive, the performance increase should be significant, and such a drive should also have a positive impact on the Air's quoted five-hour battery life, as well as being more durable, with no moving parts.

Musicians and audio engineers who order a MacBook Air with the solid-state drive will probably be frustrated by the laptop's physical connectivity. Unlike Apple's other MacBooks, which offer an extensive range of inputs and outputs, the MacBook Air has just one USB 2.0 port, one headphone output, and a video output port that supports either DVI or VGA (using the included adaptors), or composite or S-Video (using separately available adaptors).

Thin Air?

In many ways, the MacBook Air is something of an unexpected product for Apple. Although there has certainly been a gap in the company's portable line-up since the demise of the 12-inch PowerBook, that gap was arguably for a laptop with the performance of a MacBook Pro but with a smaller form factor. The MacBook Air has a stunningly small form factor, but is less powerful than a MacBook, while costing almost as much as a MacBook Pro, and possibly more if you opt for the solid-state drive.

The MacBook Air clearly caters for an elite clientele but one which is not demanding in terms of performance, and with its technical excellence and aesthetic desirability it seems curiously reminiscent of the PowerMac G4 Cube — a brilliant and beautiful product that was also expensive and limited for the type of users who would normally afford such a Mac.

In terms of power, the MacBook Air's processor, combined with the solid-state drive, should provide reasonable performance, but, as with the MacBook, shared memory for the graphics processor is not ideal for demanding applications that make intensive use of OS X graphics frameworks. And, perhaps more importantly, the fact that there's only one USB port means that your choice of audio and MIDI hardware is limited, especially if you need to connect via a hub to accommodate a USB-based copy-protection device.

It's a real shame that Apple weren't able to fit a Firewire port, since this would obviously have made a big difference to those connecting audio or video hardware, as well as external hard drives. And if not for musicians and audio engineers, what about all those highly-paid executives who might want to connect their equally expensive video camera to edit their family movies in iMovie?

Hopefully I'll get the chance to test out a MacBook Air soon, but my initial impression is that, from the music and audio perspective, you'd be wiser to spend the money on a MacBook Pro, unless you really, really need something incredibly thin, and really, really, don't need the ultimate performance or connectivity. SOS

pc notes

Are you stuck on the PC upgrade bandwagon? This month, PC Notes discusses whether it might be possible to step off it, as computer power finally starts to catch up with the needs and aspirations of musicians.

Martin Walker

When I first started writing for *SOS*, way back in 1996, my PC featured an Intel 486DX33 (33MHz) processor, 4MB RAM, a 250MB hard drive and a Sound Galaxy Basic 16 (16-bit only) soundcard. It seems incredible just how far we've come with PCs and music over the last 12 years, and how increasing processing power has kept on moving the goalposts.

Upgrade Bandwagon

For me, the first revelation was the 'plug-in', which arrived in late 1996 with Steinberg's Wavelab 1.5 and Sonic Foundry's Sound Forge 4.0a (now part of the Sony Creative Software range). Being able to add further functions to an existing application was truly liberating, and for the first time musicians could treat an audio signal with effect algorithms in real time within their software.

However, even a single metallic-sounding reverb plug-in of the time could easily consume 100 percent of your available processing power, so I upgraded to a faster PC with a Cyrix P166+ processor (clock speed 133MHz), 16MB RAM and a 1GB hard drive. I had just climbed onto the upgrade bandwagon, and during subsequent years I formulated two good rules of thumb, although one eventually proved less durable than the other:

- If you want your computer to be able to run the latest audio software, you will, on average, need a major computer upgrade every two years.
- When upgrading, you should ensure that your processing power increases by at least 50 percent, so that the expense is justified by a significant improvement in performance.

Sure enough, two years later I upgraded to a Pentium II 300MHz processor, 64MB RAM and a 30GB hard drive, largely so that I could run some of the new real-time soft synths, such as Native Instruments' Generator (later to evolve into Reaktor). This machine lasted me another couple of years until I was desperate for more processing power and switched to a Pentium III 1GHz processor, 256MB of RAM and a couple of 30GB hard drives. Two years later, in late 2003, the next upgrade became essential to cope with all the latest software, and this time I bought a Pentium 4 2.8GHz processor, 1GB of RAM and a couple of 80GB hard drives.

Divergence Of Demand

However, at this point, while my second rule of thumb still held true, the first began to break down: I was happy with this machine for three years, and it would still be perfectly adequate had I not decided to abandon my MIDI hardware synths and move over to entirely 'in the box' song creation with soft synths and software samplers.

So in December 2006 I invested in an E6600 dual-core CPU with two cores running at 2.4GHz, offering a colossal 300 percent increase in processing power over my previous machine. Even a year later this might seem like small fry compared with today's quad-core processors, yet the majority of my songs have yet to use more than 50 percent of this dual-core model.

Over the last couple of years I've noted a growing number of musicians also getting off the PC upgrade bandwagon. While some are currently buying and building quad-core machines, others decided some time ago that they were perfectly happy with the hardware and software they were already running, especially if it was rock-solid and bug-free.

Available computing power finally seems to be starting to outstrip the needs of musicians, and many of us no longer have to compromise when choosing plug-ins and soft synths. Moreover, MIDI + Audio sequencers now contain so many functions some of us never use that an increasing number of musicians are not bothering to upgrade them any further. Will I find myself desperate to trade up my PC again in late 2008? Somehow I very much doubt it.

VST Library Storage

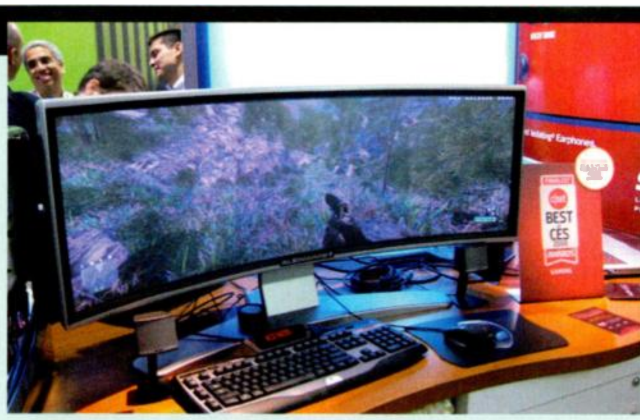
VST Instruments are now the foundation of many a musician's sonic arsenal, and are getting ever more sophisticated. To keep pushing at the boundaries, some rely on the latest physical or analogue modelling techniques, but with many designs it's the associated sample library that provides the sophistication, and nowadays this generally means many megabytes and sometimes even gigabytes of sample data.

All you need to do to install the majority of simpler VST plug-ins and Instruments so that they appear within your chosen VST host application is to place the DLL file (containing the synth code itself), plus any associated presets, in the default 'vstplugins' folder on the same C: partition as your Windows operating system. Most larger VST Instruments also store their DLL file in the VST

PC Snippets

- **Screen Star:** An interesting product launch for musicians from the CES (Consumer Electronics Show) in Las Vegas is Alienware's prototype curved screen, featuring a resolution of 2880 by 900 pixels (equivalent to two 24-inch displays side by side). It uses DLP (Digital Light Processing) technology with rear projection and offers a much faster response time than a traditional LCD display, at 0.02 milliseconds. It will be available in the second half of 2008, and while it's primarily intended for geeky gamers I can see it really appealing to composers too! No price has yet been mentioned.
W www.alienware.co.uk

- **Penryn Proliferates:** I first discussed the benefits of the larger 6MB cache and extra SSE4 instructions of Intel's new Penryn processor range back in PC Notes July 2007, but I'm pleased to report that they should finally be available by the time you read this. Intel have launched 16 new Penryn processors, covering desktop, laptop and server markets, in both dual-core and quad-core configurations.
W www.intel.com



If you fancy a single monitor-screen equivalent to two large monitors placed side by side, look no further than Alienware's new curved display, with a resolution of 2880 x 900 pixels.

Freeware Spotlight: Ferox Tape Simulator

If you want to try adding some analogue tape saturation effects to digital sounds, you could do worse than visiting Jeroen Breebaart's web site (www.jeroenbreebaart.com). Jeroen has plenty of expert knowledge, having worked for seven years as a Senior Scientist with the DSP group at Philips Research, and has developed a very comprehensive VST tape-simulator plug-in in his spare time. Ferox looks good and sounds good, with separate controls to tweak Rec Level, Low gain (bass hump), High gain (HF peak), Saturation (odd harmonic level) and Hysteresis (non-linearity). It even offers variable Noise (although I'd personally leave that set to minimum), plus Tape speed and Feedback controls to simulate vintage tape-echo effects. The presets cover high-quality tape decks, cassette machines, worn-out tape and analogue overload, and Ferox is capable of a versatile range of quality effects, from gentle thickening to total mashing. As far as I'm concerned, this is the king of freeware tape emulations, and gives plenty of commercial ones a run for their money.

With neat graphics and sophisticated sounds, Jeroen Breebaart's freeware Ferox plug-in offers a full range of tape-modelling effects, including bass and HF 'humps', odd-harmonic generation, saturation and hysteresis.

plug-ins folder, but provide the option to install the associated sample libraries on a different partition or drive. This is good news for those who want to keep their Windows partition as small as possible, so that backup files are quicker to create and smaller to store. However, just recently I've noticed musicians grumbling about various large VST Instruments that don't offer such a storage option, such as Toontrack's EZdrummer, with 1.5GB of compressed files; Manytone Music's ManyStation, again with a 1.5GB library; and the Yellow Tools Independence instruments. VST Instruments created using Jeff McClintock's SynthEdit engine also currently make you place associated files in the same folder as the main DLL file. You can end up with many gigabytes of data in your C: VST plug-ins folder that you'd prefer to house elsewhere, so here are a couple of ways to slim it down.

VST Slimming: Part One

The first method is to install the VST Instrument and library to your default VST plug-ins folder, but then manually drag all the larger sample-based files to a more convenient location (you could create a dedicated Samples partition on a different drive for all your libraries). Once they are in place in their new location, select all relevant files, then right-click on the lot and select the 'Create Shortcut' option. A batch of tiny shortcut files will appear with identical names, and you then drag these files over to the original VST plug-ins folder, so that the synth's DLL file will be able to link to the real files.

This method will apparently work with some applications, such as Propellerheads' Reason factory soundbanks, and is also used by some commercial VST Instruments, including Spectrasonics' Atmosphere and Trilogy, to link to their data.

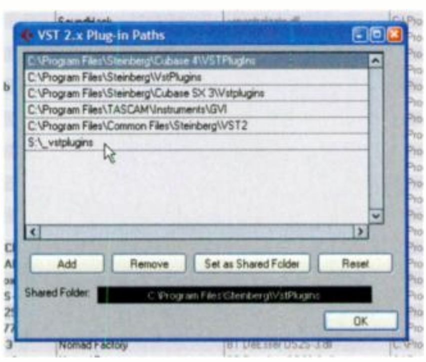
VST Slimming: Part Two

That first approach won't work with SynthEdit-based creations, or various other large VST Instruments, so here's an alternative that's often easier to implement and should work with quite a few VST host applications. It works because many of them let you define multiple 'vstplugins' folders; indeed, some VST Instruments, such as Tascam's GVI, rely on this fact; by default, GVI installs its own DLL file inside a separate Tascam Instruments folder and expects you to inform your sequencer of this additional VST plug-ins path.

First, choose a destination for any larger VSTi installs without their own library destination options, and create a new VST plug-ins folder there. I placed mine on my S:\Samples partition, where I install my other sample libraries, but gave it the name '_vstplugins' (starting with an underscore) so it appears at the top of the list of folders when alphabetically sorted.

Some sequencers, including Ableton Live and Cubase LE, can only recognise one VST plug-ins folder, so this trick won't work for them, but others are more helpful. Here's how to add extra VST plug-ins folders to the list of recognised paths for a selection of popular hosts:

- **Cakewalk Sonar:** Use the Add button of the VST Configuration Wizard in the Tools menu.
- **Cockos Reaper:** Launch the Preferences dialogue from the Options menu, navigate to the VST Plug-ins area, then use the Add button adjacent to the 'VST plug-in paths' read-out.
- **Steinberg Cubase SX and Nuendo:** Launch the Plug-in Information window from the Devices menu and use its Add button.
- **Steinberg Cubase 4:** Launch the Plug-in Information window, click on 'VST 2.x Plug-in Paths', then click on the Add button on the new window that appears.
- **Sony Acid Pro:** Offers two VSTi Search Folder paths, which you can define from the VST Instruments tab in the Preferences dialogue.



Many sequencers (including Cubase 4, shown here) allow multiple plug-in paths to be defined, so by adding another path, such as S:_vstplugins, to a convenient location, you can install large VSTi libraries anywhere you wish.

Once your VST host application has been informed of the extra VST plug-ins folder path, all you need to do is install the entire VST Instrument (DLL file, sample banks, preset files, and so on) in the new VST plug-ins folder and your sequencer will find it all automatically. If you already have bulky VST Instruments stored on your C: partition, you may be able to drag them across to the new destination, but some with serial number or challenge/response copy protection may need un-installing and re-installing in the new folder to register their new locations correctly.

Even so, the migrations aren't likely to require much effort. I've installed VST Instrument libraries in my S:\Samples partition wherever possible, but I still had nearly 3GB of VST Instruments plus sample data that I couldn't do that with, which needed moving. I managed to transfer it into my new VST plug-ins folder in under an hour, reducing the size of my Windows music partition from 8GB to 5GB. The music partition image backup file now compresses to under 3GB, takes just four minutes to create and fits on a single DVD-R disk.

After a major reshuffle and 'data shrink' like this, it's wise to defragment your partition to optimise performance, and you may also want to reduce the partition's size to allow future expansion for other partitions you might have on that drive. SOS

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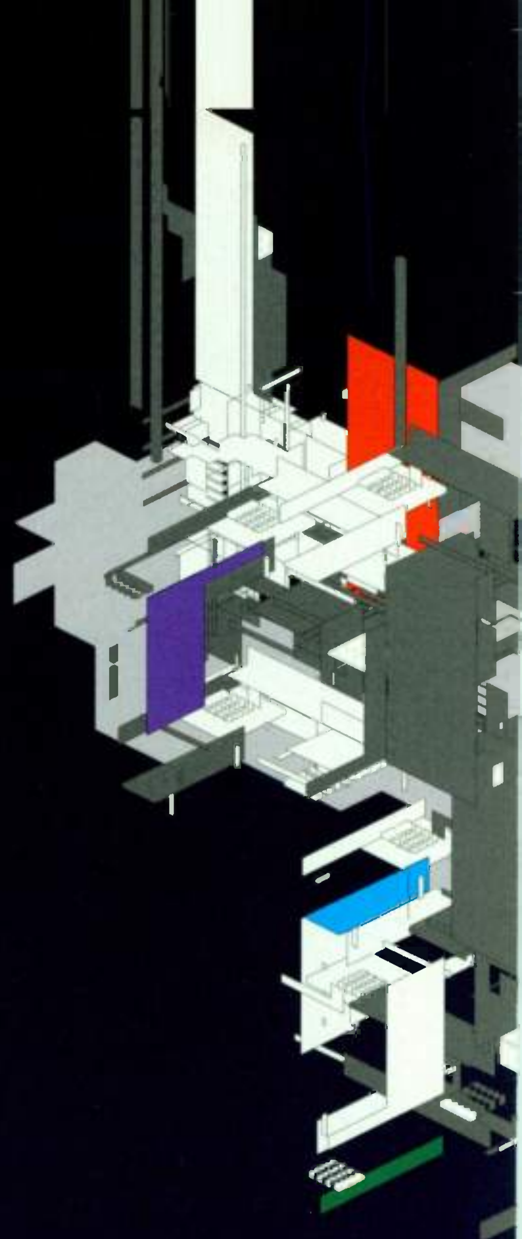

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Surround? Who's even hearing stereo?!

Mike Senior

While the jury's still out regarding the importance of surround to *SOS* readers, stereo is taken for granted. Certainly, most people I speak to mix entirely in stereo, with a handful switching to mono for a quick compatibility check on the off-chance that some weirdo is actually listening that way. Is mono at all relevant these days anyway?

The traditional reason for checking mixes in mono revolves around the fact that some broadcasters still transmit in mono. However, this argument holds less and less water as more advanced transmission formats come on-line. I'm happy to let that debate rage on between the beards in the AES wine bar, because, to be honest, I think it's pretty irrelevant. There are much more important reasons why the stereo information in our recordings rarely survives its passage to the listener.

Even in the murky underworld of the hi-fi enthusiast, stereo imaging is a delicate little flower. You might get clear positioning while you're perched precariously in the sweet spot, but shuffle across the room to peer more closely at the yellowing flyers for Elvaston steam rally and the stereo field scrunches up into whichever speaker you've moved closer to.

And for those hi-fi buffs who aren't bachelors, exactly how often do you reckon they actually listen in the sweet spot anyway? It's not the most communal experience — unless you take the concept of 'close friends' rather literally. And what of the much-quoted WAF (Woman Acceptance Factor)? Hands up how many of you have convinced her indoors of the need for acoustic foam treatment in the living room. Exactly. And how many have had their speakers banished to that corner under the side table, behind the plant? In which case you'll be lucky to hear the lyrics, let alone any stereo.

Normal domestic speaker systems fare even less well. The left speaker of your teenager's hi-fi mini-system is far too busy holding the wardrobe door open to worry about stereo imaging, even if it weren't wired up out of phase. And it stands to reason that your web PC's speakers have to huddle together at one side of the monitor to leave space for your latté. Furthermore, watching any TV from the sweet spot between its two built-in speakers guarantees eye-strain. Maybe you can save the proper stereo experience for the kitchen boom box, where its bespoke shoulder harness will keep you smack between the speakers whether you're doing the dishes or whipping up a stir fry. Or not. (Are you even sure both speakers are working? Mine recently presented me with a rendition of 'Yesterday' without the strings...)

Move outside the home and the situation is no less dire. Consider the inventive speaker placements that proprietors of shops and eateries use to avoid any risk of a stereo image intruding on your listening experience. Car stereo? If my car's anything to go by, you'd need ears on your ankles! But at least in these cases you have different speakers playing different things. What about systems that don't even make a token stab at stereo? Things like telephone hold music and the shopping centre and supermarket announcement systems that claim so many of what sinister market-research companies probably refer to as our 'total listening hours quota'.

Is there anywhere you can hear stereo? On headphones, assuming you're not sharing your ear buds with the person next to you. Or in the cinema, if you're lucky enough to get the centre seats, and assuming you're not one of the estimated 4.4 million UK residents who, like Brian Wilson, suffer from unilateral hearing impairment.

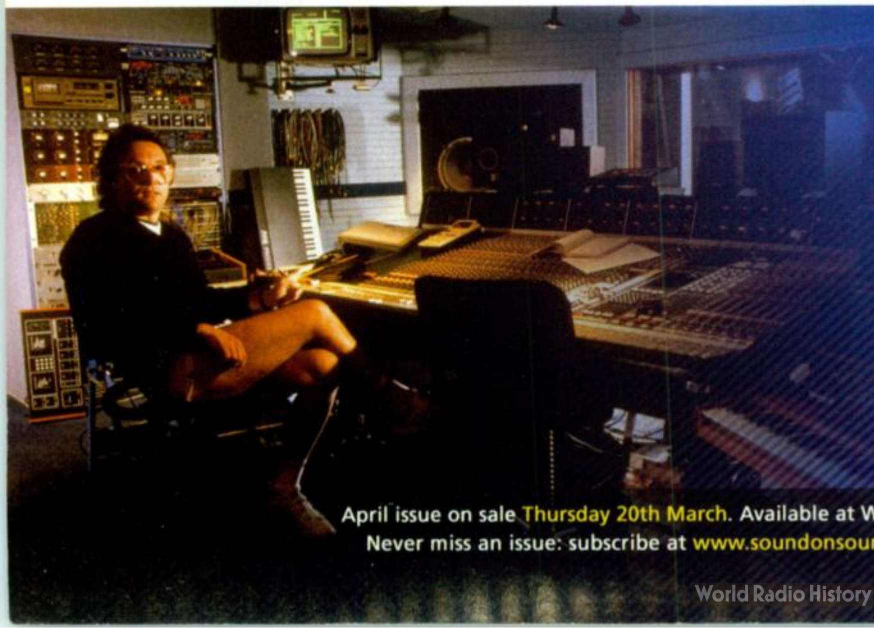
With all this in mind, ask yourself how many of your audience are going to benefit from improved intelligibility when you pan your electric guitars out of the way of your lead vocals. Or how many people will be treated to a 'creatively phase-cancelled remix' of your latest track while they browse B&Q's bathroom fittings.

So maybe next time you mix, you should try mono instead; just remember to switch to stereo for a quick compatibility check on the off-chance that some oddball is actually listening that way... SOS

About The Author

Mike Senior's recent *SOS* features include 'Recording Guitars', 'Perfect Piano' and our 'Strings' cover feature this issue. He also runs Cambridge Music Technology (www.cambridge-mt.com), offering intensive courses focusing on the techniques of the world's most famous producers. He has two ears, but likes to alternate them to reduce wear and tear.

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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With its controversial lyrics and pioneering production, Frankie Goes To Hollywood's 'Relax' became one of the biggest-selling singles of all time, despite being banned from the airwaves. Engineer Steve Lipson and producer Trevor Horn (left) tell the story in April's Classic Tracks.

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