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Introducing the 828mkII

24-bit 96kHz resolution. DSP-driven mixing and monitoring.
Front-panel programming. Stand-alone operation.

828mkII feature highlights

- **CueMix DSP™** — the 828mkII delivers DSP-driven digital mixing and monitoring for all 20 inputs. Connect mics, guitars, synths and effects processors, and monitor everything from the 828mkII's main outs, headphone out or any other outputs with no separate mixer needed.
- **Front-panel control** — access your entire mix, or any 828mkII setting, directly from the front panel.
- **Stand-alone operation** — program your mixes at the studio and then bring the 828mkII to your gig — no computer needed. Need to tweak the mix? Do it on site using the back-lit LCD and front-panel controls.
- **Multiple CueMix DSP mixes** — create different monitor mixes for the main outs and headphones. Add send/return loops for outboard gear — with no latency.
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- **Support for 96kHz ADAT optical digital I/O** (S/MUX) — provides 4 channels at 88.2 or 96 kHz.
- **Sample-accurate MIDI** — connect a MIDI controller and/or sound module with no separate interface needed. MIDI I/O is sample-accurate with supporting software.

Basic features

- **Expandable 24-bit 96kHz audio interface** for Macintosh and Windows with 20 channels of input and 22 channels of output (simultaneously).
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- **8 TRS analog inputs** with switchable input levels.
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- **8 channels of 24-bit ADAT optical** input/output with sample-accurate ADAT SYNC.
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- **Compatible** with virtually all audio software on Mac OS 9, Mac OS X and Windows Me/2K/XP.
- **Includes AudioDesk®** sample-accurate workstation software for Mac OS with 24-bit recording/editing and 32-bit automated mixing/processing/mastering.

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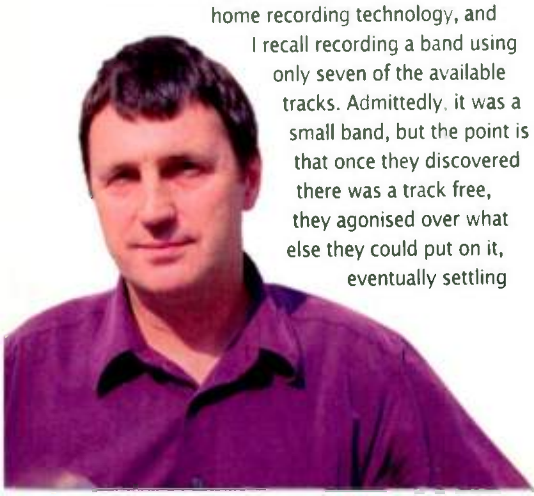
leader editor's comment

Too Many Hooks Spoil The Broth

Members of the computer music community argue continually about whether it's better to opt for the value-for-money of a powerful PC system or for a more user-friendly but costlier Macintosh. I've no doubt that these discussions will continue regardless, but the reality is that either system provides more than enough horsepower for recording music. There are very few hit records or classic albums that couldn't have been made using 24 tracks or less, and where more tracks were used, it's often the case that the drum kit and unused vocal takes accounted for over half of them. These days, with virtual tracks at our disposal, alternative takes cost us nothing. What's more, while keyboard parts were once played and recorded to tape like any other instrument, in today's computer-assisted studio they tend to be handled via MIDI, which has a very low processor overhead.

Of course, you do need computing power to run virtual instruments and plug-ins, but again, if you analyse most classic records, you'll find that there isn't as much going on as you might expect. Even dance mixes can be broken down into a modest number of simultaneous tracks, many of which are sampler-based, so it can be argued that processer speed isn't a serious limitation for most users, provided they are putting their songs together in a sensible way.

I operated an eight-track tape-based studio back when eight-track was the pinnacle of home recording technology, and I recall recording a band using only seven of the available tracks. Admittedly, it was a small band, but the point is that once they discovered there was a track free, they agonised over what else they could put on it, eventually settling for tambourine. The song wasn't written with a tambourine part, and it didn't really need one, but because there was an unused track, they felt they had to put something on it.



The urge to use every available track is still prevalent, and current recording systems, both software and hardware, provide enough power in this respect for full-scale musical megalomania. Sure enough, a common fault with many of the home recordings I hear is that there's no space left in the mix. At one time this might have been down to the use of too much reverb, but nowadays it's more likely to be due to too many unnecessary layers. Part of the problem is that many of us use a lot of off-the-shelf samples and synth patches, and if we can't find something exactly right, we try to create it by layering multiple elements. While this can be effective for string pads and the like, it's more likely to have a muddying effect, leaving you with a mix like an overworked watercolour painting.

Perhaps a worse mistake still is to use excessive processing to try to knock an audio recording into shape. If the unprocessed recording doesn't sound pretty good in the first place, then it's probably not good enough to use. Examine both the performance and your recording technique and see how you can get closer to the sound you want. Often this is as simple as moving the microphone or hanging up another duvet. Similarly, choose your synth sounds and samples with care and edit or EQ them where necessary to make them sit properly in the song. Not only will this reduce the load on your computer, it will also almost certainly result in a better-sounding recording with more space and definition. This might sound like hard work, but it's not nearly as difficult as thinking up things to record on all those spare tracks — believe me!

Paul White *Editor In Chief*

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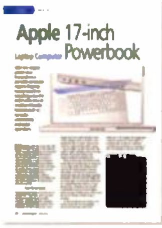

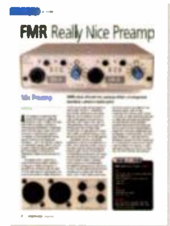

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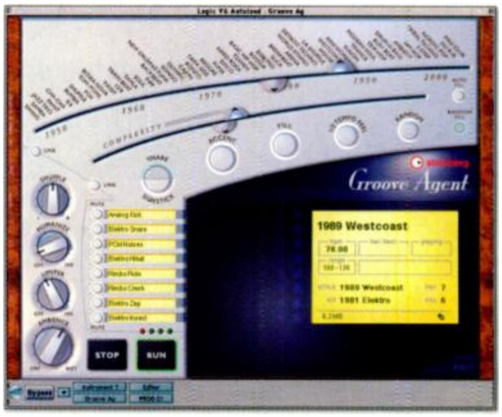

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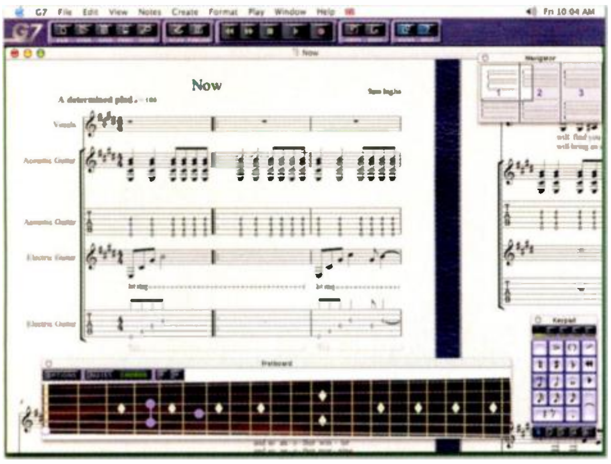

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Yamaha revive classic effects processor

Yamaha's SPX series of multi-effects processors, from the SPX90 to the SPX1000, have found their way into many a studio rack over the last 15 years. In fact, the SPX90 was reviewed in *SOS* March 1986! Now Yamaha are injecting some new life into the line with the SPX2000. Don't be fooled by its decidedly retro facia though — there are plenty of new features inside. The SPX2000 is built around a 96kHz audio DSP chip, featuring 24-bit/96kHz audio throughout and 32-bit internal processing. The unit has been designed to meet the standards of Yamaha's DM-series digital mixers, with 24-bit A-D/D-A converters and a dynamic range quoted as 106dB. There's a new reverb algorithm on board, titled RevX, which features in 17 of the 97 redesigned presets, and there are some new parameters like Room Size and Decay Envelope. Yamaha haven't completely turned their backs on the past, however. In addition to the Preset bank and User bank (which can store 99 user programs), there is a bank of 25 presets called Classic, which should bring back a few memories of the original SPX90 presets. Menu navigation should be familiar to SPX users too, with two sets of cursor keys and a number of dedicated function buttons. The SPX2000's LCD can light up in five different colours which can be assigned to user programs. Although this might seem frivolous at first, it has its uses. For factory presets, the colours signify effect type (blue for reverbs, white for delays, and so on), for instant recognition, even from a distance or in the dark. Red is reserved for warning messages. The SPX2000 is equipped with XLR and quarter-inch inputs and outputs, each switchable between -10 and +4dBu. Digital connection is catered for by AES-EBU in and out and BNC word-clock input, and there are MIDI In and Out/Thru connectors. A USB port connects to a computer for editing and data management using editor software which is scheduled for release at the end of the year. The SPX2000 itself will be launched in September, priced somewhere under £900.

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Yamaha plan update for Motif Rack

Yamaha have announced plans for a software OS update to the Motif Rack, the new rackmount synth reviewed in last month's *SOS* by Nick Magnus (see www.sound-on-sound.com/sos/jun03/articles/yamahamotifrack.asp). As regular readers will know, Nick noticed timing problems when using the Motif Rack multitimbrally during the course of his review, which we reported to Yamaha.

The new version will be released for free download from the Yamaha web site (www.yamahasynth.com) during June, at the same time that the Motif Rack begins to ship in the UK. Also being made available from the site at the same time is a new Multi-part editor application for PC. At the time of going to press, tests with a beta version of the new update show a decent improvement in the multitimbral MIDI timing, although the update is still being worked on. There will be a full report in next month's Crosstalk pages after the final version of the update ships in June, and after we've had the opportunity to do some more tests!

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US winner gets the bundle

Our October 2002 competition prize of the Waves *Gold Native Bundle* was won by Dr Ernst Karel of Chicago, Illinois, one of the steadily growing number of *SOS* readers in the USA. The software package, which is worth an impressive £1169, was kindly donated by Sonic Distribution (+44 (0)1582 843900/www.sonic-distribution.com) and contains 19 plug-ins, including *L1 Ultramaximizer*, *Q10 Parametric Equalizer*, *AudioTrack* virtual channel strip and the *Maxxbass* processor, in VST, MAS, Direct X, RTAS and Audiosuite formats. Ernst (pictured below left) described the Waves plug-ins as "a more than welcome addition" to his home studio, which contains some pretty serious analogue gear, including the Doepfer modular synth behind him in the photo.

Back on this side of the pond, Jon Magill of Birkenhead, Merseyside, was first out of the hat to win the TC-Helicon Voice One studio vocal processor that we featured as a prize in *SOS* December 2002. The unit, which costs £849, was provided by the nice people at TC Electronic UK (+44 (0)800 917 8926/www.tc-helicon.tc) and combines voice-modelling and pitch-correction facilities with a wide range of vocal effects and real-time MIDI control. Jon says he expects the Voice One will "come in very handy", and since he's recently started studying at the SAE Institute, it should be especially useful for making his fellow students turn green with envy.

Carillon develop purpose-built Pro Tools PC

Carillon Audio Systems have produced the world's first Windows-based music PC specifically designed for use with Digidesign's Pro Tools HD system. The Carillon AC1/HD, which was designed in conjunction with Digidesign (and is officially endorsed by them), features an Intel motherboard and processor and is capable of 128 audio tracks with full *Pro Tools* functionality. The AC1/HD is available on its own for those who already have a Pro Tools HD rig, or it can be supplied and configured in conjunction with the full system. CPU speed and RAM and drive sizes can all be specified when ordering, with prices starting at £1249 for the basic Pentium 4 machine.

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Gateway School to offer honours degree

The Gateway School of Recording and Music Technology is launching a new BA (Hons) degree in Audio Technology and Music Business Studies. The degree will complement the School's existing Higher Education Diploma in Audio Technology and Music Industry Studies, allowing past and present students to undertake a third year of study to complete the honours degree. The degree is validated by Kingston University and accredited by the Association of Professional Recording Services (APRS) and the Music Producer's Guild (MPG). The Gateway School has also been named as a Digidesign Certified Training Location, one of the first in the country, and will initially be offering two Pro Tools courses.

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Spectrasonics release virtual instruments for OS X

Spectrasonics have released their range of virtual instruments for OS X. Available as a free download to registered users, the update means that Spectrasonics' *Stylus*, *Atmosphere* and *Trilogy* instruments are now fully compatible with Audio Units, VST and RTAS host applications. Support for the MAS platform under OS X is to follow shortly.

W www.spectrasonics.net

New compressor is safe and sound

Safe Sound Audio are a new company located in West Yorkshire's Peak District. Their first product, designed and built in the UK, is the P1 Audio Processor, a compressor/expander which packs an impressive range of features into a small and unassuming box. There are balanced and unbalanced mic, line and high-impedance instrument inputs, an insert send/return jack (TRS), plus a pair of inputs for monitoring an external source. The level of the source is controlled by a knob on the front panel, as is headphone volume. Output is via a single balanced TRS jack. Elsewhere on the front panel, in addition to input gain, expander and compressor controls, there are switches for +48V phantom power and the high-pass filter (80Hz, -18dB per octave), plus two eight-LED meters, one measuring gain reduction, the other showing output level prior to the output gain control. The P1's compressor section features something called 'peak ride', which utilises multiple side-chains to allow the compressor to ride the peaks in the audio input. This means that the compressor can respond quickly to sources which contain fast or sudden peaks, such as vocals, yet retain release characteristics which complement the natural decay of the audio waveform. The limiter provides protection against overload and has three zero-delay side-chains so that the limiter threshold can quickly adjust to the dynamics of the incoming audio. Finally, the P1 is equipped with an internal power adaptor, taking power directly from the mains without the need for a wall-wart. This well-specified processor is expected to retail for around £250, underlining Safe Sound's resolve to compete with products manufactured in the Far East. Look out for a review in *SOS* soon.

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Four-way headphone distribution from Presonus

Following on from the launch of the diminuitive Tube Pre preamp (see *SOS* News April 2003), Presonus have released the HP4 headphone distribution amplifier, adding to their range of small, affordable studio tools. The HP4, which is a third of a standard rack space in size, has four discrete headphone outputs on its front panel, each with its own volume control. The unit can cope with balanced or unbalanced and stereo or mono input signals and has a mono selector button to sum a stereo signal to mono. There's an extra pair of TRS outputs which can be used to hook up a second HP4 or to feed control room monitors. This monitor output also has its own volume control on the front panel, along with a button to mute it. The HP4 is powered by a wall-wart adaptor to ensure high output, and an excellent noise floor figure of -98dB is claimed. The HP4 should be available in the early summer and will cost around £145.

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W www.presonus.com

Digi 002 is racked without pain

Less than a year after the launch of the Digi 002 combined control surface and Firewire audio/MIDI interface, Digidesign have announced the Digi 002 Rack, a 2U rackmount version of their Digi 002/*Pro Tools LE* system, without the hardware controller but also without the original 002's £2000 price tag. The Digi 002 Rack, which will cost £1058, features 18 simultaneous channels of audio I/O (eight analogue, eight ADAT digital and two S/PDIF digital) in 16- or 24-bit and at sample rates up to 96kHz. There are four mic/instrument preamps with XLR and quarter-inch jack inputs, each with individual gain controls and phantom power (selectable in pairs). The remaining analogue inputs and outputs are on quarter-inch, balanced/unbalanced TRS jacks, and inputs are switchable between -10 and +4dBu via mini-switches at the rear of the unit. The headphone and monitor outputs each have dedicated volume knobs and there's an additional main stereo output on unbalanced RCA connectors and an RCA stereo input for routing audio equipment directly to the monitor out. Add to this a one-in, two-out MIDI interface, S/PDIF and optical digital I/O, a footswitch input and a second Firewire port to allow daisy-chaining of a second Firewire device and the package is complete. The Digi 002 Rack ships with Digidesign's *Pro Tools LE* software which supports up to 32 tracks of 16- or 24-bit audio, with integrated MIDI sequencing and real-time mixing and processing. The system is currently only compatible with Mac OS 10.2, but Windows support will arrive with the forthcoming release of *Pro Tools* v6.1.

Digidesign have also added a new educational section to their web site. Christened DiSK (Digidesign Sound Knowledge), the mini-site features short instructional videos on various features of *Pro Tools* software, downloadable templates, sounds and samples, plus hints and tips on Pro Tools and its third-party plug-ins. You can find it at www.digidesign.com/disk.

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New HHB catalogue now available

HHB Communications have published their 2003 catalogue, containing 1500 products from over 150 manufacturers. The 140-page colour catalogue contains comprehensive product information on a wide range of microphones, mixers, monitors, outboard hardware, recorders and recording media, and the section covering computer hardware and software has been expanded. The catalogue has already been mailed to all current HHB customers and is freely available to qualifying audio professionals in the UK. Alternatively, you can browse the catalogue on-line at the web address below.

E sales@hhb.co.uk
W www.hhb.co.uk

More on the Mains

Ben Duncan, long-standing contributor to the British technical press on the subject of mains supply problems and author of this month's detailed article about mains in the home studio, has announced a compendium of his writings on the subject of enhanced mains supplies for audio. *Audio Quality Mains Supplies* contains readable DIY articles on the principles of audio-friendly power conditioning, as well as covering many other issues mentioned in the *SOS* article in greater depth. It's over 50 A4 pages in size, and is available priced £22 from the address below.

T Pro Audio Accessories +44 (0)1234 741152.
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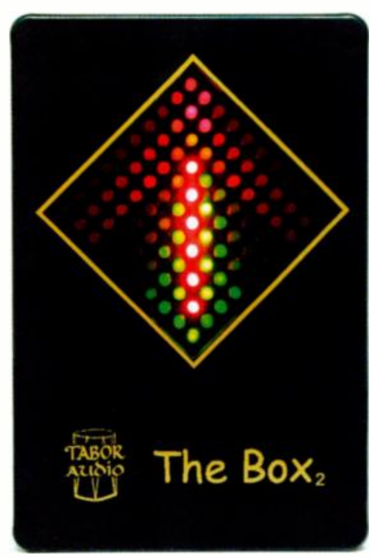
Boxing clever

The Box 2 is a thoroughly ingenious product manufactured by Tabor Audio, designed to give the recording or mixing engineer a visual impression of the stereo image. A hundred LEDs, arranged in a diamond shape, are driven by the incoming stereo signal. The central column of red LEDs represents the centre of stereo, and a mono test tone panned from left to right will be displayed as a vertical line moving from left to right. The height of the column indicates volume. Thus, in practical monitoring situations, the Box 2 offers a continuous visual analysis of balance, stereo width, stereo position, and even phase problems. The Box 2 connects to a line-level signal via a stereo, quarter-inch TRS jack and is powered by an external adaptor. A calibration button produces a vertical line in the centre of the display, which can be compared to a test tone to match up the centre of stereo for the source to the Box's centre. The Box was first conceived of and designed by the late Frank Fox (the 'Fox Box') and was further developed by Mike Skeet, who added the calibration button and the coloured LEDs. The Box 2, which is a little smaller than the original (it measures 150 x 100 x 60mm), costs £450 and is available now.



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HRS120 — a fine choice to be the ".1" in surround mixing applications.
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6.7" Active Studio Monitor
The 6.7-inch, 2-way nearfield monitor that's more than just a smaller version of the HR824. The perfect monitor for those who need incredible midrange accuracy for lead vocal placement within the mix and for dialog replacement in post.
HR824
8.75" Active Studio Monitor
As the industry-standard studio monitor, the 8.75-inch 2-way nearfield HR824 is ideal for those who need lots of bottom end, as well as comfortable high end for long sessions of general mixing.
HR626
Dual 6.7" Active Studio Monitor
The new HR626 has Phase-accurate D'Appolito driver alignment design with bi-amplification. Superb bass and midrange accuracy for mixing and vocal placement. Also an ideal choice for horizontal placement in center channel applications.
2002
NEW
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NEW
With the new HRS150 and HR626, there are now 5 ways to have Mackie studio accuracy.
LUCASFILM THX pm3
No matter what your application — from 7.1 surround to stereo monitoring, there's now a Mackie active monitor and subwoofer that will give you exactly what you want: unflinchingly accuracy.
Our new HR626 has ten more square inches of active cone area than the HR824 for even mo' bass extension, yet is as tight and fast as the compact HR624. It makes a great main stereo monitor or center channel for surround mixing.
And speaking of mo' bass, the new HRS150 uses a 15-inch active driver, two 12-inch passive radiators and 950 watts of FR Series power to achieve frequency response that's only 0.5dB down at 10Hz. Yes, you read that right. Ten Hertz!
Hear the whole precise Mackie family, including the new HR626 and HRS150, at your nearest Mackie dealer today.
TOOLS FOR ARTISTS
tools for artists
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"MACKIE." AND THE "RUNNING MAN FIGURE" ARE REGISTERED TRADE-MARKS OF MACKIE DESIGNS INC. IF YOU CAN FIGURE OUT WHAT'S LOGICALLY INCORRECT WITH THE HR626 PICTURE AND THE ARROW - YEAH FINE, WHOOPY DOO, WE KNOW - PLEASE STOP CALLING US ABOUT IT.
MACKIE.

SOS editors hit the road

Intrepid *SOS* Editor In Chief Paul White and Technical Editor Hugh Robjohns have been out and about, hosting question-and-answer sessions at music colleges around the country. So far, the pair have visited the Birmingham Conservatoire of Music, City College Manchester and Worcester College of Technology. Hugh was also present at the Radio 1 On The Road three-day event at Leeds Metropolitan University, 6th-8th of June, giving one-to-one advice on home and studio recording. Questions from students at the lively sessions have ranged from specific technical queries relating to equipment and techniques to requests for advice on how best to find work in the ever-competitive music industry. Any universities and colleges who are interested in arranging similar events are encouraged to contact us.

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Latest compressor from Ted Fletcher has multiple personalities

The latest bit of studio hardware from TFPro is the P8, or, to call it by its other name, Edward the Compressor. It's clear that the intention is to create a compressor with character and, as it happens, the P8 has four of them. Its four different compression modes are designed to mimic the characteristics of famous compressors. The VCA setting imitates modern VCA-controlled compressors, the 1176 setting is based on the Urei 1176 compressor, the LA2A setting mimics the Teletronix unit of the same name, and the Green Box setting revives Ted Fletcher's own Joemeek SC2. Besides the conventional input/output gain, ratio, attack and release controls, this stereo optical compressor is equipped with a stereo-width control, allowing the user to reduce stereo spread or to artificially widen it to 150 percent. Another unusual control, labelled 'transient release', only functions when in SC2 mode. The knob sets the recovery time of the compressor side-chain after a brief transient overload, changing the degree to which these transient peaks colour the compressed sound. The P8 costs £899 and is available now.

T **Jonny Monument UK +44 (0)1392 274477.**
F **+44 (0)1392 491447.**
E **sales@tfpro.com**
W **www.tfpro.com**

Missing credit

Last month's interview with Joe Zawinul was missing a photo credit. Our pictures of Joe, Ivan and his studio were taken by Mr. Bonzai.

Onboard mixing and stand-alone operation for MOTU Firewire interface

Having produced the first Firewire audio interface equipped for serious applications back in 2001 with the 828, Mark of the Unicorn continue to set the pace with the launch of the 828 MkII, adding a further two channels of audio input, MIDI I/O, 24-bit/96kHz operation and an eight-buss mixer, fully accessible from the front panel. In terms of its appearance, the 828 MkII is closer to MOTU's 2408 PCI-based interface than the original black and blue 828, and the new model features the 2408's five-segment LED level meters for the eight analogue inputs, two mic inputs, stereo S/PDIF I/O and main stereo buss. Perhaps the most striking new feature is the blue LCD display in the centre of the front panel, which shows the settings of the internal mixer. As before, these parameters can be edited from the computer desktop using the included *Cuemix* software, but now input gain, panning, stereo-pair grouping and so on can also be accessed using a set of knobs on the front panel. Separate monitor mixes can be set up for the main outs and headphone out, while two additional send/return stereo busses allow the user to patch in outboard gear. The unit also functions as a stand-alone digital mixer when not connected to a computer. The 828 MkII is equipped with eight TRS jack, balanced/unbalanced analogue inputs and outputs, individually switchable for +4 or -10 dBu input. These are located at the rear of the unit along with the main stereo out, the two sends, co-axial S/PDIF in and out and the eight-channel ADAT optical I/O (four channels at 96kHz). There are two mic inputs on Neutrik combi jacks on the front panel with dedicated +48V phantom power switches and trim knobs, and the grand total of 20 inputs and 22 outputs can operate simultaneously. The one-in, one-out MIDI interface is a welcome addition and the word clock I/O, ADAT sync input and SMPTE timecode I/O should be more than adequate for resolving any MIDI and audio timing issues. The 828 MkII is also equipped with a second Firewire port so a second Firewire device can be daisy-chained. The interface is compatible with Mac OS 9 and OS X and Windows ME/2000/XP and comes packaged with MOTU's Mac-only *Audiodesk* audio workstation software, based on the audio element of *Digital Performer*. Best of all, the 828 MkII will cost the same as the original 828 — £795.

T **Musictrack +44 (0)1767 313447.**
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E **info@musictrack.co.uk**
W **www.musictrack.co.uk**
W **www.motu.com**

MOTIF-RACK

Creativity - *concentrated*

Rarely does a new synthesizer generate such universal acclaim as the Yamaha MOTIF. Now, we've taken the best of its killer sounds - and many more besides - to produce this beautifully compact, 1U rack-mounted tone generator: the MOTIF-RACK. Yet despite its shrinking dimensions, over 1,000 voices, 128 note polyphony and two PLG expansion slots, mean MOTIF's creative possibilities continue to grow.

http://www.yamahasynth.com/pro/motifrack/

YAMAHA

www.yamaha-music.co.uk

London-based synthesis event Modular 2003 returns

Following on from the success of last year's event, the London College of Music and Media at Thames Valley University will once again be hosting its annual forum on creative modular synthesis. Modular 2003 will run from Wednesday September 10th to Friday 12th, and the schedule this year has expanded considerably. The three days will see lectures, presentations, exhibitions and workshops, with subjects ranging from advanced modular synthesis techniques to artificial intelligence and cutting-edge technologies, and representatives from Korg, Clavia, Native Instruments and IRCAM will be present to demonstrate new and existing products. Systems and programs featured at the event will include the Clavia Nord Modular and Nord G2, Native Instruments *Reaktor*, Software Technology *VAZ Modular*, Symbolic Sound Kyma, Cycling '74 Max/MSP, SynC Modular, IRCAM Jmax, SoftSynth Jsyn, Csound, and Pure Data. A key part of Modular 2003 is the Modular Room, a two-day installation featuring a wide range of analogue modular synths from the past and present. Analogue synthesis experts will be on hand to discuss and demonstrate the various systems. Tickets cost £50, which includes entry into all three days of the event plus the showcase concert on the evening of the 11th. You can find more information and purchase tickets at the Modular 2003 web site (address below).

W www.modular2003.com

New interfaces from Terratec

Terratec have produced a handy converter box for interfacing co-axial and optical S/PDIF formats. The Vice Versa essentially has two modes of operation. In bi-directional mode, the optical input is routed to the co-axial output and the co-axial input is routed to the optical output. In its other mode, the Vice Versa acts as a signal repeater. The input signal, which is set to either optical or co-axial by a switch, is routed to both outputs so that, for example, a single co-axial signal can be routed to both the optical input of a soundcard and the co-axial input of a DAT machine. The unit is powered by an AC adaptor and its integrated signal booster allows for longer cable runs. The Vice Versa can also be powered by a computer's USB buss. 24-bit/192kHz audio is supported, along with AC3 and DTS DVD surround formats. This handy little black box is reasonably priced at £49.99 including VAT and is available now.

Terratec have also produced a USB preamp for recording old vinyl LPs into a computer. Roughly the size of a pack of cards, the Phono Preamp Studio USB is switchable between phono and line inputs and ships with CD-burning software and *Algorithmix Sound Rescue*, an audio-restoration package. There are signal and peak indicator LEDs and the device is powered by USB. The Phono Preamp Studio USB is available now, costing £89.99.

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E mail@scvlondon.co.uk
W www.scvlondon.co.uk
W www.terratec.net

New distribution for Phoenix Audio

Audio Agency have taken on UK and European distribution for Phoenix Audio. The company was originally set up to service vintage Neve studio equipment before developing their own range of Class-A discrete mic preamps and DIs, drawing inspiration from the classic Neve consoles in both design and appearance. The Phoenix Audio DRS2 dual-channel mic preamp/DI, reviewed in February's *SOS* (www.sound-on-sound.com/sos/feb03/articles/phoenixdrs2.asp), now has a mono sibling, the DRS1. The half-rack unit features the same circuitry and specifications as a single DRS2 channel, offering +70dB gain and an impressive +24dB headroom on the input stage. There are balanced XLR and high-impedance DI inputs and outputs, and -20dB-pad, high-pass filter, earth-lift, phantom-power and phase-reverse switches. The DRS1 is priced at £799 including VAT.

Audio Agency are also distributing a new product from the US. Direct Sound's Extreme Isolation headphones are designed to isolate the listener from external noise, and claim up to 29dB of noise reduction. The list of those who might benefit includes drummers who need headphone monitoring, live sound engineers and one-room studio owners. An added bonus is that when external noise is reduced, signal level can also be turned down, preventing unnecessary wear and tear on the ears. The Direct Sound EX29 Extreme Isolation headphones cost £125. Finally, Audio Agency are distributing the Little Labs range of high-end studio tools for guitarists. Products include the Redeye passive DI/signal splitter, the PCP signal distributor and the IBP analogue phase-alignment tool.

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W www.audioagencyeurope.com

Emagic *Logic* v6.1 supports Digidesign TDM systems

The latest *Logic* update, version 6.1 for OS 9 and OS X, is now available from www.emagic.de as a free download to registered users of *Logic* 6. The update allows *Logic* to make use of Digidesign TDM systems under Mac OS X and means users can run Emagic software instruments, like the *ES2* synth, *EVP88* electric piano and *EXS24* sampler, via Digidesign hardware. The update also allows users to import Akai S-series sample CDs into *EXS24*. *Logic* v5 users who download the update can enjoy a free 28-day trial of version 6.1.

W www.emagic.de

"There wasn't any one thing, it was cumulative. I was impressed with every aspect of it.

NUENDO

With Nuendo I'm able to work in the way I like to work – controlled chaos!

Nuendo has allowed me to be a lot more creative with sound design within the system than I ever was with other systems. The VST plug-in architecture allows me to be as adventurous as I want with the ability to undo anything I've done."

Peter Baldock

In-demand Supervising Sound Editor/Sound Designer Peter Baldock recently put the flexibility of Nuendo to work on the forthcoming S-Club movie, edited and dubbed at Lipsync Post, Soho.

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For further information please contact
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AMG launch new web site

Sample specialists AMG have launched a new web site, www.samples4.com. The site allows visitors to audition demo files, purchase samples securely over the Web, and search the entirety of AMG's large catalogue (all releases in all formats are listed). There are many products which are only available via the site, and many more which are being made available for the first time. AMG have also introduced the option of paying by PayPal. For those of you who just can't get enough samples, AMG are releasing three new GigaDVD sample collections. *GigaRefill Plus Vol 2* features 4.2GB of data with 12 complete Refills, including Keith LeBlanc's *Essential Trilogy* and Neil Conti's *Dark Side of the Groove*, for £450, a saving of £180 over the total price of the Refills sold separately. *GigaREX Vol 2* and *GigaWAV Vol 3* both contain the same eight sample CDs: *Ultramagnetic Beats, Prototype Drum & Bass, Space Invader, Dirtbag, Cuckooland*, and *King Tone Grooves*, plus the Keith LeBlanc and Neil Conti CDs already mentioned. Totalling 4500 files in all, the *GigaREX Vol 2* and *GigaWAV Vol 3* DVD collections cost £375 each, representing a saving of £165.

T AMG +44 (0)1252 717333.
W www.samples4.com

New web site for UK *Nuendo* users

Steinberg's *Nuendo* media production software now has its own dedicated UK web site, set up by UK distributors Arbiter. The web site, which can be found at www.nuendouk.com, features full product information on all *Nuendo* software and hardware components, up-to-date price lists, plus news, features and reviews. The Downloads section covers all *Nuendo*-related software and hardware driver updates, while chargeable upgrades can be purchased from the on-line shop. You'll also find contact details for UK dealers and *Nuendo* technical support. *Nuendo 2*, Steinberg's most advanced pro audio application, started shipping at the end of May.

W www.nuendouk.com

Access offer live on-line customer support

Access Music have extended their customer support services by launching a live on-line chat room to help users with their queries. At present, visitors to www.access-music.de/chat can get immediate answers from Access staff during office hours, and the synth manufacturers are hoping to eventually provide 24-hour customer support.

New software for Emu Command Stations

Emu Systems have released version 2.0 software for their XL7 and MP7 Command Stations and Proteus 2500 Command Module, available as a free download from www.emu.com. The new OS features a number of new performance and sequencing features. A new function called XMix lets you swap tracks from one pattern to another on the fly, and patterns can now be assigned to and triggered by the 16 buttons. The Key Up Layer function means that keys can be programmed to trigger sounds when they are released rather than pressed. A full list of the new features contained in the update is contained in the 22-page PDF document accompanying the software download.

W www.emu.com

New WDM drivers for legacy MIDI interfaces

Owners of the MusicQuest 8 Port SE and Opcode MIDI Translator PC MIDI interfaces might be interested to know that WDM drivers are now available, meaning that these interfaces, which were previously only supported under Windows 9x, can be used with Windows 2000 and XP. The drivers were written by Evert van der Poll of Earth Vega Connection and are available to download at the addresses listed below. Both cost 20 Euros and demo versions are also available.

W http://surf.to/8portse
W http://surf.to/miditranslator

Remote editing software brings KARMA to the Korg Triton

Karma Labs have released new software that lets Korg KARMA users edit their keyboard workstation's settings from their PC or Mac desktop, and gives Korg Triton users access to the KARMA's sophisticated Generated Effects (GEs) for the first time. The software has been designed by Stephen Kay, who originally developed the KARMA's technology and licensed it to Korg (KARMA, if you didn't know, actually stands for 'Kay Algorithmic Real-time Music Architecture'). *KARMA MW* is a dedicated version of the KARMA software, exclusively for use with the KARMA keyboard. It offers complete access to the 400-plus parameters which make up a GE via a graphic editor. Generated Effects are sequences of arpeggios, evolving textures and rhythmic effects which respond to what's being played. Since only 16 GE parameters are accessible from the KARMA keyboard, *KARMA MW* greatly extends the workstation's possibilities. In addition, the software allows for graphic or grid-based editing of drum patterns, velocity patterns, envelopes and more. Phrases and drum grooves from the KARMA's internal sequencer, or from any Standard MIDI File, can be edited and converted into GEs. Data is exchanged between the computer and the keyboard via MIDI SysEx messages, or via floppy disk.

KARMA Triton offers the same functionality as *KARMA MW* but is designed for use with any of the Korg Tritons (Classic, Studio, Rack or LE), which is not as surprising as it might sound considering that the KARMA's sound engine is largely based on that of the Triton. *KARMA Triton* comes with the complete set of 1190 GEs, 384 programs and 384 combinations from the original KARMA factory set, and data can be freely exchanged with the Triton via SysEx. *KARMA Triton* can convert Triton arpeggios into KARMA GEs, automatically assigning parameters to knobs and switches to give instant hands-on access. *KARMA MW* is available through Korg UK, priced £119, or directly from the Karma Labs web site (address below), priced $149. *KARMA Triton* is only available direct from Karma Labs and costs $199. Both programs are compatible with Windows 98, ME, NT and XP and Mac OS 8.6 or later. Mac OS X will not initially be supported.

As mentioned briefly in May's News pages, Korg have recently announced OS v2.0 for the Korg Triton — and it's now available to download. The OS update adds a number of new commands and functions, including one-touch recording and better loading and handling of samples. A Swing parameter has been added to the quantise section and there's a new tap-tempo feature which allows users to set the speed of an arpeggiator or tempo-based effect by tapping a foot pedal or button on the front panel. Perhaps the biggest single advance is that the new OS allows two tracks of audio to be recorded directly to the internal hard disk. There are two dedicated audio tracks on the sequencer and audio can be normalised, stretched, bounced down or mixed, amongst other functions. It's also possible to place WAV files directly into these sequencer tracks. Files can also be shared between the Tritons and the KARMA, even without the *KARMA Triton* software. Korg Triton OS 2.0 is avalable to download now from the Korg web site.

T Korg UK Brochure Line +44 (0)1908 857130.
F +44 (0)1908 857199.
E info@korg.co.uk
W www.korg.co.uk
W www.karma-labs.com

TL Audio chop 25 percent off price of Valve Classics

From the 1st of June, TL Audio are cutting the price of four outboard processors from their Valve Classics range by up to 25 percent. There's a hefty £500 off the list price of the VP1 Mono Valve Processor (pictured), which drops from £1999 to £1499. The EQ2 Dual Parametric Valve Equaliser, previously £1399, now costs £999, while the EQ1 Limited Edition Dual Valve Equaliser, previously £999, now costs £699, with limited quantities remaining. Finally, the price of the PA1 Dual Pentode Valve Preamp has been reduced from £899 to £699. These price cuts may encourage those interested in TL Audio's Ivory Series to spend a few hundred pounds more and move up to the Classic range.

T TL Audio +44 (0)1462 492090.
F +44 (0)1462 492097.
E info@tlaudio.co.uk
W www.tlaudio.co.uk

Zero Seal Systems promise zero noise

If you're into building your own studio or converting an existing room for recording purposes, you might be interested in the new catalogue of acoustic-insulation products from Zero Seal Systems. The 20-page brochure covers their range of hinges and head, jamb and door bottom seals. A brief introduction to the basic acoustic principles at work explains how even the smallest air gap around the door frame will allow external noise into the room. Zero's Sound Trap door kits claim Sound Transmission Loss (TL) figures of over 52dB reduction in volume, though individual components can also be purchased separately. You can order a catalogue directly from Zero, or browse the Web-based version at the Zero home page (see below).

T Zero Seal Systems +44 (0)1785 282910.
W www.zeroplus.co.uk

Audiotrak offer 192kHz audio for £100

Audiotrak have produced a two-in, eight-out 24-bit PC soundcard capable of 192kHz output in stereo mode, all for a penny under £100. The Prodigy 192 features a stereo analogue line input, six analogue outputs and two digital outputs (optical or co-axial S/PDIF). Analogue input is passed through 24-bit/96kHz A-D converters, while output is handled by 24-bit/192kHz D-A converters with a quoted dynamic range of 104dB. The optional MI/ODI/O daughtercard, sold separately for a reasonable £49.99, adds a 24-bit/96kHz S/PDIF optical input, co-axial input and output and a one-in, one-out MIDI interface. For those not already in the know, Audiotrak is the consumer products division of ESI (formerly known as Egosys), and the Prodigy 192 is equipped with the same EWDM (Enhanced Windows Driver Model) driver technology as the ESI Wami Rack and Waveterminal soundcards, compatible with Windows 98SE, ME, 2000 and XP. The 192's driver supports ASIO 2, GSIF, MME and Multi-MME formats, with full duplex (simultaneous recording and playback) capability. The card also supports ESI's Directwire system, allowing the outputs of numerous virtual instruments and audio applications to be re-routed internally.

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news
www.sound-on-sound.com

From Pocket PC to DAT

Those of you who were intrigued by the review of the *Griff* Pocket PC Sequencer in *SOS* May 2003 might also be interested in a product being developed by an American company called Core Sound. PDAudio is a relatively inexpensive system of hardware and software designed to convert your PDA into a mobile location recorder. At the core of the system is a Compact Flash card S/PDIF interface with optical and co-axial inputs, which can be mounted in any host with suitable hardware slots running Windows CE/Pocket PC 2002 or Linux. It can also be used with laptop and desktop PCs running Windows 2000, XP or Linux. Using the *PDAudio Recorder* software for Windows and Linux, the system can record to the Pocket PC's memory or to removable PC or Compact Flash cards, via a dual-card expansion adaptor. *PDAudio Recorder* allows monitoring of signal level, time elapsed and storage media capacity, with conventional transport controls and level meters. It allows data to be recovered in case of power failure and features a 'lock' mode to prevent accidental button pressing. The system can record in single- and dual-channel modes, in 16- or 24-bit, and supports sample rates up to 192kHz, though it's worth noting that some PDAs will not support sample rates above 96kHz. Contact Core Sound for more information.

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W www.core-sound.com

PSP release plug-ins in RTAS format

PSP Audioware have produced an RTAS version of their *Vintage Warmer* plug-in for Mac OS 8, 9 and X. *Vintage Warmer* emulates analogue-style single or multi-band compression and limiting, and features carefully modelled overload and saturation characteristics, a wide range of presets and support for sample rates up to 192kHz. The plug-in's retro GUI, with its accurately calibrated VU/PPM meters, completes the package. *Vintage Warmer* is available to download from the PSP web site now and costs $149. PSP's free metering plug-in, *Vintage Meter*, has also been ported to RTAS format for Mac OS 8, 9 and X. The plug-in features the same VU/PPM meters as *Vintage Warmer* and allows users to indulge in a little nostalgia when monitoring the level of mono or stereo tracks on screen.

W www.pspaudioware.com

EUROPE'S LOWEST PRICES GUARANTEE*

The first 3 pages of this advertisement feature a small selection of the deals from our

32 PAGE CATALOGUE

IN THIS ISSUE OF SOS

the buyer's

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The definitive monthly reference directory of best selling pro audio and music technology products, free with this magazine.

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£399.99
ALES-QS6.2

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EMU

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Emu's feature-packed E6400 Ultra sampler now offers even better value, with a massive 128MB of sample RAM factory fitted, and an internal 20 GB hard drive onboard, stuffed with sounds from the Emu library. The E6400 Ultra is 64 note poly with 8 balanced analogue outputs, S/PDIF, floppy drive, 48 track sequencer and Emu's newest EOS Version 4.01 software with 'Beat-Munging' for re-writing the time signature of rhythmic loops, FIR phase linear filters and Transform convolution.
Strictly limited stocks - first come, first served!

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£699.99
EMU-E6400ULT

MODULAR RACK SYNTHESIZER

nord

60% OFF!

Rack synth version of the Nord Modular keyboard, for those with MIDI controllers more playable than the Nord's little 2 octave keyboard! The analogue modelling is based around core blocks such as oscillators, ADSR envelope generations, filters, modulators etc which are all modelled by the software. All patch editing is done via the computer and the result is loaded into the Nord. The architecture gives you virtually unlimited routing of modular synth blocks for analogue modelling, and it's all user definable and with more modular software blocks being created by Clavia, the upgrade potential is huge.

KEYBOARD VERSION ALSO AVAILABLE only £649.99!

RRP £1499
£599.99
CLAV-NORDMODR

DARKSTAR XP2 SYNTHESIZER

RED SOUND

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With the new XP2 model, the Darkstar analogue modelling desktop synth module gains 4 assignable outputs, headphone output, a new jog edit wheel, additional sine waveform and colourful purple highlighted finish! Retro analogue modelling comes from a 5 part 8 note poly 2 oscillator synth engine with hands-on joystick parameter control. A great tool for dance sounds.

RRP £279
£149.99
RED-DARK/XP2

ABSOLUTE ZERO PASSIVE NEARFIELDS

SPIRIT

OVER 35% OFF!

There are several well thought out points of good design in Spirit's dinky little Absolute Zero nearfields, and to be honest (as always) they are arguably the best performing monitor speakers on the market for under £200, with a very acceptable sound - it's not too bright and raspy, and neither is it too muffled and boxy. The overall sonic balance is good, and benefits from rear porting, rounded edges and a recessed, time-aligned tweeter. Woofer is 6.5 inch with a 1.5 inch voice coil. Impedance is 8Ω nominal, for a loud 95 watts RMS at 89dBSPL per watt.

RRP £205
£129.99
SCRA-ABO

CUBASE SL MEGA BUNDLE

steinberg

OVER 60% OFF!

Steinberg annouce the Mega-Bundle package deal putting together Cubase SL recording / sequencing software with Halion software sampler offering 32 bit 96kHz and surround support, and the popular VST plug-in effects packages, GRM Tools 1 and 2. Furthermore you get three volumes of drumloops on CD from the Pocket Fuel series by Rhythmic Architectural Design Systems (RADS). These are Vol 1 - Acoustic Rock drumloops, Vol 2 - Acoustic Hip Hop drumloops and Vol 3 - Techno Rhythm and Loops. The Mega-Bundle elements complement each other perfectly and make this a great value offer!

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STEI-CUBHALPC

YST-MS30 SUBWOOFER SPEAKER SET

YAMAHA

50% OFF!

Turnkey's great cost saving offer brings you these Yamaha active multimedia speakers with two satellites and a separate subwoofer, with Yamaha's YST bass enhancing technology. Satellites are 6 watt each with 2 inch full range drivers, while the subwoofer is a single 5 inch cone with 18 watt onboard amplification. The whole system is magnetically shielded.
Don't miss out on this deal!

RRP £99
£49.99
YAMA-YSTMS30B

10 DAY MONEY BACK GUARANTEE

(see below for conditions)

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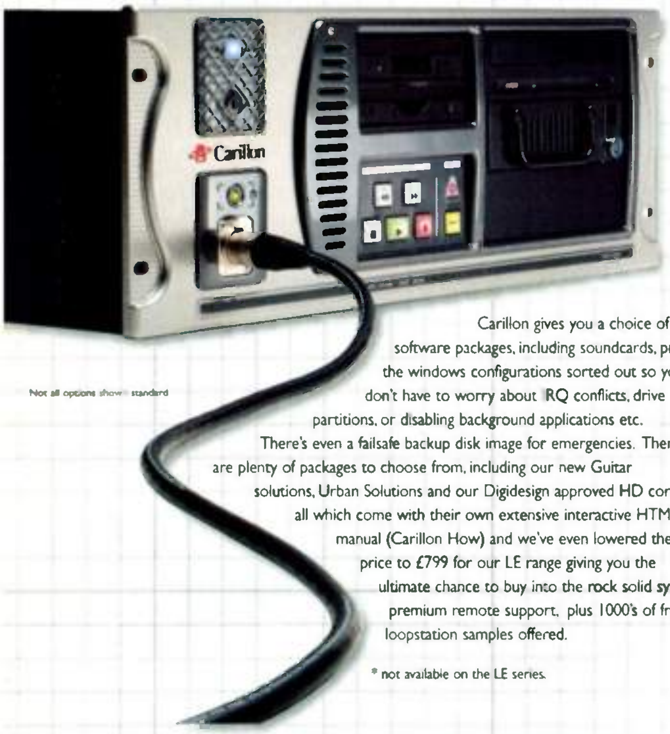
WRH

Carillon

Carillon Audio Systems proudly present the first ever "Audio Computer", designed and built specifically for music recording studio use, where noisy fans, drives, RFI interference and unreliable flimsy hardware are absolutely unacceptable!



Not all options shown standard

Carillon's heavyweight steel and aluminum rackcase is built to last, won't rattle or vibrate, and provides considerable sonic and electrical isolation, and inside are Carillon's own UltraMute PSU, plus super low noise fans, a hard drive sleeve, and many other noise killing innovations, including sorbathane anti-vibration feet and rackear dampers.*

Carillon gives you a choice of pre configured systems for popular music software packages, including soundcards, professionally installed and set up, with all the windows configurations sorted out so you don't have to worry about IRQ conflicts, drive partitions, or disabling background applications etc. There's even a failsafe backup disk image for emergencies. There are plenty of packages to choose from, including our new Guitar solutions, Urban Solutions and our Digidesign approved HD cores, all which come with their own extensive interactive HTML manual (Carillon How) and we've even lowered the price to £799 for our LE range giving you the ultimate chance to buy into the rock solid systems, premium remote support, plus 1000's of free loopstation samples offered.

* not available on the LE series

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FROM £799.99

CARI-AC1

LOGIC 5 BIG BOX
EMAGIC PACKAGE DEAL

emagic

This incredible package deal brings you five great Emagic products for under £200, for which Emagic normally charge £645 if you bought them all separately!!! Firstly it is built around the brand new Logic Audio 5 professional 48 track audio and MIDI sequencer platform with 24 bit 96kHz support and 28 high quality plug ins. This includes reference quality POWr dithering worth the price alone! Further to Logic Audio 5 itself comes the Emagic EXSP24 - the latest playback only version of the popular 24 bit software sampler, plus the complimentary Xtreme Analog EXS sample CD library, containing over 200 fat sounds. Finally, to finish this collection are two great soft synths - Emagic EVP73 vintage electric piano plug-in, based on the classic Rhodes - and last but not least, the Emagic ES1 virtual analogue modelling synthesizer, providing retro style single oscillator synth voices with resonant LPF and great preset library. All round, the substantial cost saving makes this a worthwhile package for those tempted to go the Emagic route, and provides a wide scope of musical possibilities for dance creation in particular, at a very affordable price. It would be E-tragic to miss out!

RRP £200 — £174.99 — EMAG-B GBOXDU

REASON 2.5
VITUAL STUDIO

propellerhead

REASON STRING REFILL ONLY £59.99!

NEW — OVER 40% OFF!

Reason 2.5 is the latest version of this software synth, effects and audio sequencer environment, which brings several brand new processing tools. These include the new RV7000 advanced reverb, a 12 band vocoder, the Scream 4 Sound Destruction unit, the UN16 Unison chorus, Spider audio merger and splitter and a version for CV or gate switching signals. Reason ships with a large (500MB+) sound bank with hundreds of loops, drum kits, multisampled instruments and synth patches. Get total reveal and MIDI control of all the gear you can from a single software platform.

RRP £300 — £179.99 — STEI-REASON

WAMI RACK 192
AUDIO INTERFACE

EGO SYS — OVER 30% OFF!

Outboard multichannel A/D and D/A converter in a 1U rack, with four XLR mic preamp inputs and switchable phantom powering. There are 8 analogue outputs including two master balanced XLR outputs. The A/D converters are 24 bit 96kHz max while the D/A converters support 24 bit 192kHz rates. The device offers high performance with over 100dB analogue dynamic range and includes a MIDI interface. The Wami Rack 192X version does provide A/D conversion at 192kHz. 192L...£279.99 192X...£399.99

FACTORY DIRECT — £279.99

WAVETERMINAL 192
PCI AUDIO INTERFACE

Eight channel audio interface on a PCI card with 6 analogue output channels and two digital output channels on S/PDIF optical. There are 2 analogue input channels for either stereo line in or a mic preamp input via TRS balanced jack. Both the 192L and the 192X feature 24 bit 192kHz D/A converters ready for DVD-Audio playback. The A/D converters on the 192X version are 24 bit 192kHz, while the 192L version supports 96kHz A/D. The S/PDIF optical supports 24 bit 96kHz max. The PCI card is full duplex with ultra low latency and is great for 5.1 surround playback. The 192M version includes an breakout box with 8 outs, 4 line inputs, 2 mic inputs and 2 headphone sockets. It has 96kHz A/D and 192kHz D/A out.
192L...£129.99 192M...£169.99 192X...£199.99

FACTORY DIRECT — £129.99

top deals!

computer | mixers | outboard | monitoring | integrated studios

DIGIDESIGN M-BOX
- USB audio interface for ProTools 5.2 LE
- 24 bit converters, mic preamps by Focusrite
- Stereo analogue in and out
- Stereo S/PDIF digital I/O
- Headphone socket with volume control
- Insert points via TRS jacks

RRP £405 — £374.99 — DIGS-MBOX

CM LABS DASHBOARD
- Hardware control surface for software editor workstations
- Provides transport controls, jog / shuttle and scrub wheel, timecode display, 8 assignable rotary pots for control of pan, fade etc. and an 80 character LCD
- Wide compatibility and templates, plus exp options!

FACTORY DIRECT — £849.99 — CMAU-DASH

EGOSYSTEMS NEAR 05
NEW
- 2 way bi-amplified active nearfield monitor
- 5 inch rear ported woofer, 1 inch silk dome tweeter
- 42 watt LF & 32 watt HF amps, crossover 2.7 kHz
- Balanced XLR & TRS jack inputs, 20kΩ load
- Thermal overload, RF interference protection

FACTORY DIRECT — £169.99 — EGOS-NEAR05

ADK SC1
NEW
- Small diaphragm condenser mic
- Fixed cardioid pattern
- 10mV/Pa sensitivity
- 18dBA inherent noise
- 134dBSPL peak handling
- Nominally flat response

FACTORY DIRECT — £54.99 — ADK-SC1

PRODIGY 192 SOUNDCARD
- 2 in 8 out full duplex PCI format soundcard
- 24 bit, 96kHz AD, 192kHz DA
- 6 analogue outs, 2 digital outs
- S/PDIF optical and coaxial
- Mic preamp, headphone minijack
- Daughterboard for MIDI and S/PDIF input

FACTORY DIRECT — £109.99 — ATRA-PRODIGY

YAMAHA 01X
- Firewire hardware control surface
- 28 channel digital mixer with 8 mic inputs
- 32 bit FX, 4-band EQ and dynamics processing
- 24 bit / 96 kHz
- Motorized faders

RRP — £1199.99 — YAMA-01X

JBL LSR25P
OVER 55% OFF!
- 2 way bi-amplified active nearfield monitor
- 5 inch front ported woofer with 80 watt amp
- 1 inch titanium tweeter with eliptical waveguide and 40 watt amp
- -3dB at 55Hz, switchable 80Hz HPF and volume
- Nice aluminium cabinet

RRP £799 — £349.99 — JBL-LSR25P

ADK A51/SM
NEW
- Large diaphragm cardioid condenser mic with shockmount
- Switchable -10dB pad built in, and 100Hz HPF
- 15mV/Pa sensitivity
- 17dBA inherent noise
- 140dBSPL peak handling

FACTORY DIRECT — £179.99 — ADK-A51S/SM

MIDIMAN AUDIOPHILE
- A basic stereo in /out soundcard designed to give high quality audio performance
- Analogue with 24 bit 96kHz converters and S/PDIF coaxial I/O
- Breakout for MIDI ports
- ASIO2, GSIF, DirectX

RRP £180 — £139.99 — MIAN-AUDIOFIL

TASCAM TMD4000
ALMOST 55% OFF!
- 32 channel 8 bus digital mixer with motorized faders
- Meterbridge optional extra
- 4 band parametric EQ and dynamics on each channel
- Stereo multieffects bus
- Full machine remote control

RRP £2799 — £1299.99 — TASC-TMD4000

SPIRIT ABSOLUTE ZERO
OVER 35% OFF!
- Small two way nearfields
- Rear ported 6 inch woofer, soft dome tweeter,
- 8Ω nominal load
- 95 Watts RMS, 89dBSPL per Watt
- Last few units at this amazing price

RRP £206 — £129.99 — SCRA-AB0

ADK A51SL
NEW
- Large diaphragm condenser mic
- Cardioid pattern, FET electronics
- 1 inch diaphragm
- 10mV/Pa sensitivity
- 16dBA inherent noise
- 135dBSPL peak handling
- Price includes shockmount and case

FACTORY DIRECT — £189.99 — ADK-SL

SOUNDQUEST INFINITY V2
OVER 65% OFF!
- Software synth editor / librarian for detailed patch editing via full computer interface with graphic user interface
- Supports over 400 popular MIDI instruments
- Also works within Cubase

FACTORY DIRECT — £99.99

TC M2000
50% OFF!
- Price crash on the 24 bit M2000!!!
- 24 bit multieffects unit with AES/EBU + coaxial digital I/O, MIDI control
- 2 simultaneous effects
- 500 patches, Wizard help

RRP £1099 — £549.99 — TC-M2000-24

KORG D1600 & CDRW2
- Also available with a built in CDRW drive for audio mixdown mastering, or burning backups of your multitrack data.
- Same features and spec with 20GB drive, 16 track recording and 24:8 mixer

RRP £1199 — £989.99 — KORG-D1600CD

AKG C900
ALMOST 40% OFF!
- BARGAIN!!!! Handheld 48V cardioid condenser mic
- Includes case, stand clip & PB1000 presence boost adapter (brightens treble response on axis only)
- 17.5dBA self noise at 6mV/Pa sensitivity
- 139dBSPL peak handling
- Gentle 80Hz bass rolloff

RRP £130 — £79.99 — AKG-C900

SAC-2.2 CONTROLLER
- Hardware controller interface and USB hub
- 9 motor faders, 12 assignable pots
- 67 illuminated buttons, LED scribble pads
- User templates for Pro Tools, Logic, Cubase, Sonar 2.0 etc.
- Optional 8 ch expander

FACTORY DIRECT — £1099.99 — RADI-SAC2K

AKG K240DF
OVER 55% OFF!
- Semi-open back padded circumaural dynamics
- 200mW at 88dBSPL/mW
- Professional 600Ω load
- Response -10dB, 15-20,000 Hz

RRP £110 — £49.99 — AKG-K240DF

TASCAM 788
OVER 50% OFF!
- 8 track digital portastudio
- 7.5GB hard disk drive
- 24 bit or 16 bit recording
- 250 virtual tracks, 999 undos
- 3 band EQ, digital effects and timestretch up to 50%
- SCSI & S/PDIF digital out
CDRW Drive only £289.99!

RRP £800 — £395.99 — TASC-788

BEYER MCE-530
- Bargain small diaphragm back electret condenser mic with slim body
- Cardioid polar pattern
- 30Hz - 20kHz ±6dB
- 136dBSPL peak handling
- Great low cost, ideal for buying in pairs or groups for miking drumkits

£79.99

WAVECENTRE
8 CHANNEL SOUNDCARD

FRONTIER — OVER 50% OFF!

The Frontier Design WaveCentre/PCI is an eight channel digital only PCI soundcard with 24 bit I/O support. It offers S/PDIF coaxial in and out for a selectable pair of channels in stereo and ADAT optical in and out for 8 channels, up to 24 bit 48kHz. The ADAT input may be switched to S/PDIF optical operation if necessary. All ins and outs are active simultaneously for full duplex operation across the formats, although the sample rate must match. Also offers 2x2 MIDI interface via mini DIN plug with breakout cable for four MIDI plugs.

BUY WAVECENTRE + TANGO 24 FOR ONLY £555.99!

FACTORY DIRECT — £169.99

DAKOTA & DAKOTA PLUS
SOUNDCARDS

ALMOST 55% OFF!

Dakota is a PCI soundcard for 16 channel digital I/O and MIDI. It includes 2 in 2 out MIDI interface via mini DIN socket with breakout cable and four ADAT optical sockets for 8 channels in & outs up to 24 bit 48kHz rate, with additional 15 pin D-sub for dedicated ADAT Sync. Dakota also generates or slaves to SMPTE timecode. The Montana expansion card works alongside, adding 4 optical sockets for another 16 ch ADAT I/O, plus ADAT Sync and video sync out. Dakota Plus package combines Dakota card with Montana expansion card.

BUY DAKOTA PLUS FOR ONLY £349.99!

FACTORY DIRECT — £229.99

M8 and M12
ANALOGUE MIXERS

Soundcraft — OVER 45% OFF!

Incredible savings on these M series mixers! The M8 is a 16:2 format with 8 mono and 4 stereo channels, while the M12 is 20:2 with 12 mono mic channels. Both offer 4 aux sends, 3 band EQ with sweep mid, channel direct outs, 100mm long throw faders, signal and peak LEDs and even a build in stereo A/D converter with S/PDIF digital output.
M8...£269.99 M12...£319.99

RRP £499 — £269.99

MA5A/D
DESKTOP MICRO MONITORS

EDIROL — OVER 50% OFF!

Compact active desktop speakers available in two versions - MA-5A with analogue stereo inputs, and MA-5D with additional front-panel optical S/PDIF digital input and internal D/A converter. Each front ported speaker cabinet has a full range 2.5 inch drive unit with 5 watts amplifier, and protective grille. Includes bass / treble and volume controls. Speakers are each 19 x 14 x 12 cm in black finish. Analogue version £49.99 Digital Version £59.99

RRP £100 — £49.99 — EDIR-MA5A

266XL
DUAL COMPRESSOR GATE

dbx — ALMOST 35% OFF!

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

RRP £150 — £99.99 — DBX-266XL

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*Our guarantee is that we will beat any genuine quote, on identical goods, from a UK or European dealer at time of order.

EUROPE'S LOWEST PRICES GUARANTEE

Emu's legendary rack samplers keep their place with re-vamped specs and better value each year, because the famous design and interface needs no improvement! Other companies may dis-continue and replace their products each year, but an Emu sampler is going to last!

60% OFF!

INCLUDES 128MB RAM, 128 VOICE EXPANSION & 20GB DRIVE STUFFED WITH 128 VOICE LIBRARY!

E-6400 ULTRA SAMPLER

Emu have just increased the factory spec of the E6400 Ultra sampler so it now comes as standard with a fully expanded 128MB of sample RAM and 128 note polyphony (it used to be 16MB and 64 note poly so this is a big upgrade!) PLUS they also include an internal 20GB IDE hard drive with hundreds and hundreds of professional samples to get you off to a flying start!!! The 3U rack E6400 Ultra sampler has a large LCD screen, built-in floppy disk drive and 50 pin SCSI to easily load samples from a computer or CD-ROM drive. It has a generous 8 balanced analogue outputs (expandable to 16 analogue or AES/EBU or ADATdigital outputs) and 3 external and 1 internal expansion slot for sound ROM boards. Emu's most sophisticated EOS Version 4 firmware includes FIR phase linear filters which allow you to blend filtered and unfiltered versions of the same sounds together without the nasty phase shift cancellations, along with Emu's old-school 6th order (36dB/oct) Z-plane morphing filters (21 filter types altogether) and the 48 track sequencer which makes a neat scratchpad for song ideas. Also V4 has "Transform" command, which performs a convolution process on two samples to create a new one which is the fusion of both - ideal for out-of-this-world modulation effects or adding the acoustic signature of something (like a guitar amp or valve EQ) to another sound. The "Beat-Munging" facility lets you take any rhythmic sample loop and adust its musical time signature - using advanced time-stretching and chopping up beats.
The addition of the optional RFX-32 board allows you to transform your patches at individual sample level, running up to 16 simultaneous 32-bit effects, which gives you even more creative scope than the already powerful twin 24 bit global effects processors. You can also run external audio signals into the Emu's patented morphing filters and synthesis modulators, and then control and mix them in real-time using MIDI controller data. This opens up the possibility of looping the Emu's own output back through itself again!
Strictly limited stocks - first come, first served!

RRP £1799 **£699.99** EMU-E6400ULT

EMU RFX-32 16 CHANNEL EFFECTS EXPANSION BOARD only £299.99 WHEN BOUGHT WITH E6400

YAMAHA YST-MS30 SPEAKER SET

Turnkey's great cost saving offer brings you these Yamaha active multimedia speakers with two satellites and a separate subwoofer, with Yamaha's YST bass enhancing technology. Satellites are 6 watt each with 2 inch full range drivers, while the subwoofer is a single 5 inch cone with 18 watt onboard amplification. The whole system is magnetically shielded. *Don't miss out on this deal!*

OVER 50% OFF! RRP £99 **£49.99** YAMA-YSTMS30B

YAMAHA SU200 PHRASE SAMPLER

Setting in between the SU10 and the SU700 this new sampler provides up to 83 seconds of phrases with loop remixing and realtime effects control. The 8 rubber pads (which light up pink) are chosen from 3 banks of different samples and are 6 part multitimbral and polyphonic. There is a touch sensitive ribbon controller to use for scratching samples or realtime effects / filter sweeps. The SU700 can resample the effects internally, loop (with tap tempo) and remix, and offers distortion, lo-fi, delay, tech mod LFO, slice gating FX and 17 filter types. It has mic / line in, SmartMedia card slot, battery / PSU operation and includes a sample CD.

ALMOST 35% OFF! RRP £305 **£199.99** YAMA-SU200

VOYAGER SYNTHESIZER

This monophonic analogue synth is based on the original Model D from the 70's and is a truly authentic modern equivalent of it's classic ancestor. Featuring a velocity sensitive 44 note keyboard with aftertouch, an X-Y touch pad pressure controller, 3 oscillators, a noise generator, two LPF resonant filter stages and an LFO with selectable waveforms including two sample and hold types. The Voyager B model (shown) features illuminated blue pitch/mod wheels, while the top of the range Limited Edition Signature model features the blue wheels, gold name plate, a certificate of authenticity and the signature of Bob Moog himself. Voyager B Model £2299.99 Limited Edition Signature Model £2999.99

FACTORY DIRECT **£2199.99** BIGB-VOYAGER

CLAVIA NORD MODULAR VIRTUAL ANALOGUE KEYBOARD

Available in both 2 octave keyboard and rackmount versions, the Clavia Nord Modular is a versatile analogue modelling synth offering 5 oscillator voice architecture with 5 LFOs, two resonant 4 pole low pass filters and powerful ADSR envelopes. It also offers classic 6 operator FM synthesis models and 16 band vocoder. Clavia have designed the synth for integration to a computer software editor where you can construct the sounds by simply arranging virtual patch chords on the screen and upload the result into your synth as presets.

OVER 40% OFF! RRP £1095 **£649.99** CLAV-NORDMODK

RACK VERSION ALSO AVAILABLE only £599.99!

mics | sampling | keyboards | modules | recording

BEYER MCE-90
OVER 35% OFF!
- Large diaphragm, back electret mic
- Up to 130 dBSPL peak
- 75Hz HPF switch
- 40Hz to 20KHz frequency response
- 10 dB pad

RRP £230 **£149.99** BEYE-MCE90

BEYER MC-834
OVER 30% OFF!
- Large diaphragm cardioid condensor
- Up to 150 dBSPL peak
- Switchable HPF at 80Hz or 160Hz
- 10 dB pad
- 18dBSPL-A self noise, 20mV/Pa sensitivity

RRP £499 **£349.99** BEYE-MC834

SHURE SM58 LCE
ALMOST 45% OFF!
- THE absolute industry standard mic
- Used by countless top performers
- Dynamic cardioid, with upper mid presence
- Includes stand clip and carrying pouch

RRP £125 **£70.99** SHUR-SM58LCE

ACCESS INDIGO REDBACK
- Limited Edition signature model
- 3 octave analogue modelling synth keyboard
- 32 note poly, 1024 presets, 4 oscillators
- 98 simultaneous FX, 6 outputs
- 6 pole filters, aftertouch, pitch / mod wheels

FACTORY DIRECT **£1329.99** ACCE-REDBACK

ACCESS VIRUS C KEYBOARD
- 6 outputs, 2 inputs
- 32 note polyphonic 16 part multitimbral
- 61 note velocity and aftertouch sensitive keyboard
- 98 Simultaneous DSP FX Surround sound
- 1024 programs

FACTORY DIRECT **£1429.99** ACCE-VIRUSKC

ALESIS QS6.2
ALMOST 45% OFF!
- 5 octave synth keyboard featuring QS composite synthesis
- 16MB sound library and Ocarl expansion slot
- 64 note poly, 64 part multitimbral, 3 LFOs, 3 envelopes, 4 layer voices
- 4 bus 24 bit effects

RRP £699 **£399.99** ALES-QS6.2

OBERHEIM MC1000
- 76 note weighted MIDI controller keyboard
- 2 zones, 2 MIDI outs, 1 auxiliary MIDI input
- Aftertouch, pitch, mod
- 2 assignable sliders
- 88 note version also available only £399.99

FACTORY DIRECT **£399.99** OBER-MC1000

OBERHEIM MC2000
- 88 key weighted action MIDI master keyboard
- Aftertouch, pitch wheel and two mod wheels
- 8 zones, 2 MIDI ins, 4 MIDI outs, 4 data sliders
- MC2000 EX stage piano also available only £729.99

FACTORY DIRECT **£599.99** OBER-MC2000

OBERHEIM MC3000
- 88 key hammer-weighted MIDI master keyboard
- Aftertouch, pitch wheel and two mod wheels
- 8 zones, 2 MIDI ins, 2 MIDI thru, 8 MIDI outs, 8 data sliders, MIDI patchbay
- Oberheim OP1 Sound Expansion only £29.99!

FACTORY DIRECT **£849.99** OBER-MC3000

WALDORF MICRO QKB
- New 37 note keyboard version of Micro Q Rack
- 37 note keyboard
- 300 programs 100 multis
- 20 drummaps, 8 edit dials
- 16 buttons, 2x20 char LCD
- 3 stereo outs, 1 stereo in

FACTORY DIRECT **£699.99** WALD-MICROQK

WALDORF Q
OVER 40% OFF!
- Digital analogue modelling synth with sequencer + arpeggiator
- Digital output, sound morphing, lots of knobs!
- Built in digital effects
- Q Voice expansion also available only £499.99

FACTORY DIRECT **£1299.99** WALD-Q

ACCESS VIRUS RACK
- Analogue modelling rack synthesizer
- 16 note poly, 4 outputs
- New Version 4 firmware
- 12 buttons and 5 assignable rotary encoders
- 2 audio inputs
- 6 pole sweeping res filters

FACTORY DIRECT **£599.99**

ACCESS VIRUS RACK XL
- 1U rack virtual analogue modelling sound module
- 32 note polyphonic
- 3 oscillators, 6 outputs
- 98 simultaneous effects
- Nice red LCD screen!
- Includes Emagic Sound Diver software for PC/Mac

FACTORY DIRECT **£849.99** ACCE-VIRUSRXL

CLAVIA NORD MICRO MODULAR
- A complex synth engine with PC editor software
- 100 patches, compatible with Nord Modular
- 4 note poly, 2 audio ins & outs, phones, PC, MIDI
- Patch libraries available from www.clavia.se
- Unbelievable deal at this price. Don't miss out!

RRP £375 **£275.99** CLAV-NORDMIC

CLAVIA NORD MODULAR RACK
ALMOST 40% OFF!
- Fully modular analogue modelling synth
- Editing software for PC or Mac included
- Great for polyphonic Moog style lead lines
- First come first served on this amazing low price!
- Keyboard version also available only £649.99!

RRP £995 **£599.99** CLAV-NORDMODR

NOVATION NOVA
OVER 30% OFF!
- Virtual analogue modelling digital rack synth
- 3 oscillators per voice, 12 note poly, 6 part multitimbral
- 6 outputs, 24 bit converters
- 40 band vocoder
- 42 simultaneous effects!!
- 256 sounds
- Last chance to grab one at this amazing low price!

RRP £649 **£449.99** NOV-NOVA

REDSOUND DARKSTAR XP2
NEW
OVER 45% OFF!
- Hot new version of the acclaimed Darkstar synth
- 8-voice polyphony 5-part multi-timbral
- 2 external audio inputs
- New features: 4 outputs, sine waveform, jog dial, headphone output

RRP £279 **£149.99** RED-DARKXP2

SOUNDART CHAMELEON
NEW
- 1U rack 24 bit mainframe DSP processor for MIDI and audio
- Open ended software allows many uses depending on downloaded firmware
- Users can write their own software too

FACTORY DIRECT **£599.99** SART-CHAMELEO

WALDORF ATTACK RACK
- VST soft synth now as hardware version
- 24 sounds over 2 octaves per percussion set
- 6 outs/2 ins
- MIDI in/out/thru
- 2 oscillators, waves, FM, 4 effects, 2 envelopes

FACTORY DIRECT **£449.99** WALD-ATTACKR

WALDORF MICRO Q RACK
- 2U rack virtual analogue synth
- 25 voice poly, 6 outputs
- 5 osc, 3 LFOs, 4 pole filters
- Vocoder, prog arpeggiator, ring mod, 16 mod routings
- 5 simultaneous effects
- Omega version also available only £599.99

FACTORY DIRECT **£499.99** WALD-MICROQR

KORG KARMA SYNTHESISER

BUY WITH 2 PCM BOARDS FOR ONLY £100 EXTRA!

Using Korg's Hyper Integrated synthesis, the Karma provides 425 multisamples and 413 drum sounds from 32MB of ROM, and delivers 640 user presets including a full GM2 bank, plus 64 user drumkits and 768 combis. There are 5 stereo insert effects, 2 global effects and a global equaliser, with 102 effect algorithms. The 16 track sequencer holds 200,000 notes, with 200 songs, and reads and writes SMF format 0 and 1 to its floppy drive. Patterns can be assigned to single keys. 8 knobs, 4 outputs and 61 keys.

ALMOST 35% OFF! **£999.99**

OBERHEIM OB12 SYNTHESISER

NEW LOW

12 note poly analogue modelling synth with 4 analogue outputs and S/PDIF digital. Oscillators can mix 3 waveforms including pulse width modulation and noise, and even have a ring modulator, dual multimode filters can be routed in a variety of different ways for maximum flexibility, multi waveform LFO's can be synced to MIDI. Features 49 keys, aftertouch, ribbon controller, knobs which transmit MIDI plus a 32,000 note sequencer and arpeggiator. Effects include sound morphing, EQ, reverb and that legendary fat Oberheim sound. *An incredible synth even at the old price, now it's unbelievable, with a saving of £200 !!*

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For more hints, tips and problem-solving visit the SOS Discussion Forum
www.sound-on-sound.com/sosforum.htm

Q&A

Your technical questions and queries answered

Q How do I connect a SCSI drive to my VS multitracker?

I have a Roland VS880 V-Xpanded Version 2.06 and I'm trying to hook up a Syjet 1.5GB drive to backup data. Despite a number of tries I haven't been able to get it to work. Can you tell me what I might be doing wrong?
Dan Farrow

Reviews Editor Mike Senior replies: *It's impossible to diagnose the exact nature of your problem without further information, but here's a list of possible causes to investigate. Firstly, are you sure that the Syjet actually works properly? Syquest drives have a bit of reputation for being unreliable, so it might just be that the drive is faulty. Try hooking it up to a computer to check that it's still working. Secondly, how have you connected up the drive? You should check that you're actually using a proper SCSI cable, as there are other cable types with*

The venerable Roland VS880 multitracker can connect to external drives via SCSI.

25-pin 'D'-Sub connectors which aren't suitable for interfacing SCSI devices. Next, what are the SCSI IDs of the VS and the Syjet? You may have a conflict. You'll probably be able to set the Syjet's SCSI ID via a switch on the drive casing. The VS880's SCSI ID can be set from the Disk PRM submenu under the System menu; the option you need is called SCSI Self. The important thing is to make sure the Syjet and the VS are set to different SCSI ID numbers. If there any other devices on the SCSI chain, try removing them while you sort out the Syjet. If there are two SCSI sockets on the Syjet, then make sure termination is switched on. If there is no termination switch, then buy a SCSI termination (from a computer supplies shop) suitable for the SCSI plug type the Syjet uses, plugging it into the spare port. You may need to experiment with which way round you plug up the two Syjet ports as well. Finally, you may need to format the drive specifically for use by the VS880. I'm not sure that drives formatted in normal computer formats are acceptable.

Q What's the difference between Humanise and Groove Quantise?

My sequencer offers both Humanise and Groove Quantise functions. Are they different things? Do I need to use both or just one?
Eddie Barton

***SOS* contributor Len Sasso replies:** *Generally you would use one or the other as they are somewhat at cross purposes. Humanising usually refers to making slight, random adjustments to the timing of notes. Often there are options to also randomise Velocity and note length. The idea is that you want the adjustments to be barely noticeable — just enough to remove the rigid quantised feel of step-entered sequences.*

If you have sequencing software that offers both step-entry of notes and humanising, try step entering simple drum and bass parts. Listen to the results then try humanising one of the parts, and finally, try humanising both. See which you prefer — rigidly quantised parts fit some musical forms better while humanised parts fit others.

Groove quantising, on the other hand, is the process of matching the timing of one sequence to that of another. The idea here is that you pick a part (typically bass or basic drums) that has the feel or 'groove' you want, then align other parts to the same groove. The groove might be played in or might come from commercial groove templates, such as the DNA groove templates available from Numerical Sound (www.numericalsound.com).

Groove quantising is just another form of quantising, in which the quantise 'grid' is supplied by a human player rather than being composed of rigid note divisions. As such, it is already humanised and randomising it would tend to destroy the effect. But most sequencing software will allow you to control the degree of quantisation (rather than snapping everything exactly to the quantise grid) as well as to quantise Velocity and note length. That provides a further degree of humanisation in that all parts are not exactly the same. If your software doesn't offer those features, humanising after groove quantising might be useful.

If you have sequencing software that offers groove quantising and supports REX files or some other beat-slicing format, you have a ready source of grooves to try. Simply take the MIDI files that reflect the timing of the slices and use those for your groove-quantise grids.

Q How can I get rid of a standing wave?

I want to sort out the standing wave in my control room, but I'm not quite sure what to do. The room is four metres square, with walls sloping away from each other on two opposite sides. My setup is along one of these walls, just off centre, with the monitors sitting about five feet apart. I've isolated my speakers as much as possible so they are away from my PC monitors and standing on little metal feet. I do need to tip them forward slightly so I am in the best monitoring position but when I do move into that position there's little difference. The wave's fundamental is at 44Hz, but where I sit to mix I hear it most at 88Hz. Frequencies above this aren't really affected. Besides double tacking the room, which isn't really feasible at the moment, what do you suggest I do to solve this problem?
SOS Forum Post

Editor In Chief Paul White replies: *As I understand your description, your room is roughly square-shaped, which is clearly bad news for standing waves, and I'd guess that the height of your ceiling isn't far off half the wall length. Sloping walls have very little effect on low-frequency problems but dense foam corner bass traps should help even out the bass response while three-inch foam panels either side of where you sit will cut down flutter echoes. The other strategy for small rooms is to choose monitors that don't have too great a bass extension — monitors that roll off at around 60 to 70Hz will generally give better results than those that go right down to 40Hz. You may also find that moving the monitors by as little as six* ▶

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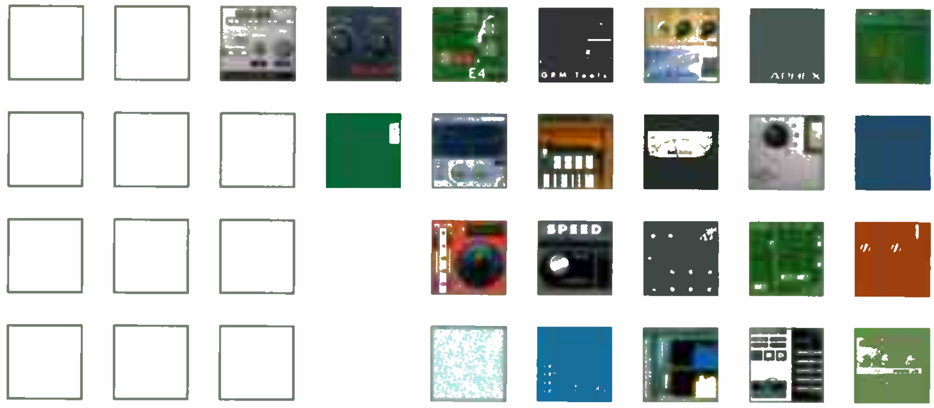

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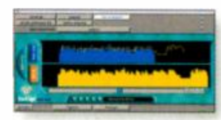

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Q&A

▶ *inches will affect the way the bass end behaves, so it's worth experimenting with this while playing back a sequence of equal-intensity bass lines. If you can, it would also help to put some foam on the front and back walls, but only use one to two square metres in total on each wall as too much foam will soak up all the high end and allow the low- and mid-frequencies to dominate.*

Q What's the difference between pan and balance?

Most of the audio recording and playback software I use has a pan control for stereo audio channels. Shouldn't that be called 'balance' and isn't it different from panning?
***SOS* Forum Post**

***SOS* contributor Len Sasso replies:** *Yes, technically mono channels should have pan controls and stereo channels should have balance controls. But most audio software will play both mono and stereo audio files on any track and even allow you to mix the two on the same track. The software is smart enough to perform the proper function based on the data being played. Panning distributes a mono signal between the left and right output channels. Balance simultaneously alters the levels (in opposite directions) of the two channels of a stereo signal, but the left and right channels go exclusively to the left and right outputs, respectively.*

To the extent that the two channels of a stereo signal share the same information, balance has a similar effect to panning — the shared information appears to shift from one channel to the other. To the extent that they have different information, balance acts to suppress the information on one side and enhance it on the other. As an unlikely example, if you have a guitar panned hard left, a singer in the middle, and a bass panned hard right in a stereo file, the balance control will pan the singer while controlling the levels of the guitar and bass.

Q Can I record control voltages on my sampler?

If analogue modular synths utilise control voltages, and a digital sampler converts incoming voltage levels into digital data and then back again, could you record a control voltage signal into a sampler, then play it back from the sampler into a modular's CV input, and get the same result as the pure CV signal?
James Kennedy

Editor In Chief Paul White replies: *In theory, yes, but the capacitor coupling used in audio equipment, including samplers, precludes the passing of DC voltages, so while modulation waveforms may work, slow envelopes and pitch information would almost certainly not. Most audio gear rolls off any frequencies below 20 or 30Hz, so even slow LFOs would suffer. Also, you'd need to calibrate the output gain control on your sampler very carefully so that you get back exactly the same level of control waveform as the one you put in.*

Q Which PC and monitors should I buy?

I am interested in buying a music PC and have read the *SOS* reviews of both the Carillon and Digital Village PCs. Although impressed by both, I'm still rather undecided, so could you please tell me which one would be more suitable for my needs. I have been writing my own rock, pop, and dance music using the Yamaha AW4416, but would like the flexibility a music PC would bring to my home studio. As I live in Northern Ireland and only have limited computer experience, I'm also concerned about technical support if any problems should arise. I also want to buy a good monitoring system on a budget of eight hundred pounds such as the KRK V4s or the Mackie HR624s. Would these be easy to set up with my new computer?
Ciaran Doherty

Music PCs from Carillon and Digital Village — just two of the growing number of purpose-built PC DAWs on the market.

Editor In Chief Paul White replies: *Both PCs should do the job so I'd suggest buying whichever you feel is quietest, especially if you plan to record in the same room as your PC, as I expect you would with the AW4416. Read over the reviews again and see which one seems to best suit your needs. Providing you buy the computer with the software and soundcard of your choice already installed and configured, and you don't go adding any software other than music plug-ins, you shouldn't run into any technical difficulties that can't be fixed with a phone call to tech support. Having said that, it's always helpful to get to know a local PC user just in case. As to monitors, the Mackie HR624s would work well with either system, as would the KRKs, but if you are planning to use any active monitors without a hardware mixer, a monitor control box such as the inexpensive Samson C Control would be useful to control the monitoring level, and would also provide you with a headphone feed for overdubbing. Personally, I'm a fan of the Mackie monitors, but don't let that put you off trying out the KRKs too, as these are also very good. Also, check out any Dynaudio models in your price range as well as some of the less costly active monitors from other manufacturers, such as the Alesis M1 MkII Actives.*

Q What's wrong with my AKG valve mic?

My AKG SolidTube has been playing up for a couple of years and shows the following symptoms. When I power up, I get no signal, or occasionally a really quiet, really distorted one. When I shout into it (loudly), it suddenly 'unblocks' itself and I get the full signal. From that point onward, the mic seems to work fine, although during one project it didn't seem to have completely unblocked itself and I had to go back and shout down it ▶

Q&A

▶ again. I've just taken it to my local shop and their engineer reckoned there was a dry joint on the valve base which he's re-soldered and on testing he said the problem had gone. I've just tried it back at home and sadly the problem is still there. I realise it's difficult to comment without seeing the mic, but do you have any idea what the problem is? Once it gets going, it works fine, and it seems odd that it should behave this way.

Simon Lees

Editor In Chief Paul White replies: *Unless this is a hardware fault (in which case the mic needs to go back to Arbiter, the distributor, to be fixed), all I can think of is that you have condensation on the diaphragm. Capacitor mics can be very susceptible to picking up condensation, especially when they are in a cold or humid environment, so storing them in a warm dry place helps. Using a pop shield should also help keep the moisture in the singer's breath from reaching the capsule. To test the hypothesis that condensation is to blame, next time the fault occurs, place the mic somewhere warm for half an hour and see if the problem goes away.*

The AKG SolidTube, a popular valve condenser mic.

Q Can I use multiple ASIO drivers in *Nuendo*?

Is it possible to use multiple ASIO drivers at the same time in *Nuendo*?

SOS Forum Post

Reviews Editor Mark Wherry replies: *In a word, no — not even the new architecture in* Nuendo 2 *supports this. However, some manufacturers' ASIO drivers do allow for multiple hardware devices, so a single ASIO driver can service two RME cards (of the same type), a variety of Creamware boards, several MOTU devices (of the same type), and so on. But you can't mix and match different devices from different manufacturers in a single ASIO driver.*

Q What compressor settings should I use for the spoken word?

I need to set up a compressor for a spoken word recording. Can you offer any advice?

SOS Forum Post

Technical Editor Hugh Robjohns replies: *You'll need to select a compression ratio that isn't too severe and set the threshold so that the compressor applies the appropriate amount of dynamic reduction to control the signal without sounding obvious. Squashing*

When recording speech, the right compressor settings are just as important as good diction, posture and a neat and tidy haircut.

voices too heavily is extremely unpleasant and immediately noticeable.

Attack time is never too critical on voices, but release time is. If the release time is set too slow, a loud word will punch a hole in the following speech. Set it too fast and you'll hear pumping.

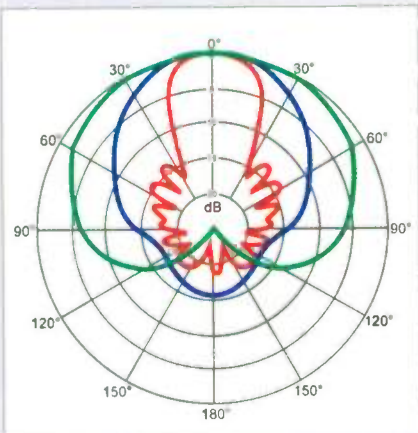
I'm not going to give any figures for these settings here. Instead, experiment and use your ears. You will quickly understand what all the controls do and how best to use them. If it sounds right, it is right! You might also want to look at an article I wrote a few years ago on recording speech, which is available at www.sound-on-sound.com/sos/1997_articles/jan97/spokenword.html.

Q What's the best way to mike up a choir?

I need to reinforce the sound of a performance by a 40-voice childrens' choir, some of which will be accompanied by electric guitar, drums and a funk/blues brass section (played by kids, hence incapable of playing softly). I've done similar things before where I put a couple of basic dynamic mics in front of the choir, but I found I couldn't really get the choir loud enough without getting feedback. Please could you recommend some fairly cheap mics that would do the job and tell me the best way to set them up. The choir stand in four rows of 10 across, with the brass usually in a line behind them, though I suppose it would make more sense to put the brass to one side to avoid picking them up with the mics.
SOS Forum Post

Technical Editor Hugh Robjohns replies: *Rather than blowing your budget on mics that will only be used once or twice a year, why not hire some more appropriate kit for the job. That also means the money comes from a different budget than capital purchases! You've already mentioned the most important thing — you need to move the brass away from behind the kids. Send them over to the other side of the stage! Apart from anything else, there could be a noise safety issue with brass players blasting away in the delicate ears of the youth choir. By moving the brass to the side, their contributions will be coming in to the side of the mics directed at the choir, where you'll encounter some rejection, rather than straight down their main axis. Putting the kids on risers is also a good idea, as is making them look upwards, and the brass* ▶

Q&A



The 'squashed spider' polar response pattern of the Sennheiser MKH416 rifle mic for high frequencies (above 8kHz) is shown in red. The mid-frequency (1-8kHz) polar pattern of the same mic is shown in blue, as compared to the typical cardioid response, shown in green.

section downwards! You could, for example, set the choir in the middle of the stage, with the brass to one side and the drums and bass to the other. You can then use a bunch of cardioid mics close to and in the choir, let's say two or three mics across the front, and maybe another two in front of the third and fourth rows. Alternatively, you could achieve a similar coverage with just a couple of short rifle (or shotgun) mics, which have a far narrower pickup pattern, so could be mounted further back.

It might help if you try to think of mic polar patterns in the light of everyday three-dimensional objects. A cardioid mic's pickup pattern is a little like an apple, with the stalk pointing to the back of the mic. The more gain you wind into the mic amplifier the bigger the apple gets. So imagine that you've put a single mic up in front of your choir, and that you've given it enough gain to make the apple grow until it encompasses all the singers. By the time the frontal area of the mic has covered the choir, the sides are extending a long way out too, hence possibly picking up a lot of spill from other sound sources in those directions. If instead you use two or three mics, and/or move the mic or mics closer to the choir, you can see (in your mind's eye, that is) that you won't need as much gain in each mic before their apples have collectively encompassed the choir. There will therefore be less of a problem with spill, although you will now have the problem of the same sounds arriving at several spaced microphones, potentially causing phasing problems. The way to avoid that is to make sure that the distance between mics is at least three times greater than the distance between each mic and its source.

Rifle mics have a polar pattern that looks a mess (see diagram, left), with lots of narrow side lobes that look a bit like a squashed spider! However, for practical purposes you can generally think of its pattern as being something more like a lightbulb. Although the frontal pickup is relatively narrow compared with a cardioid mic's more rounded pattern, it has much greater side rejection, so you can use it from further away without capturing so much spill. Because it's further back, the mic can also see more of the choir. Two or three mics should be enough for this kind of job. At low frequencies, below about 500Hz, the rifle mic's response pattern becomes very similar to a hypercardioid, which is why I recommended filtering off the LF when using this kind of mic in a PA application.

Q Why are the waveforms of the miked and direct signals different?

I've recorded some acoustic guitar using both a mic and the guitar's internal pickup simultaneously, but the waveforms of the mic and pickup tracks do not look similar at all (see screenshot, below). Can you offer any explanation for this? There's obviously some kind of delay between what seem to be the same peaks in the two tracks (around 220 samples). Is this just because current through cables travels faster than sound through air? Do I have to align the tracks and match up the peaks?
Ralf Lehmann

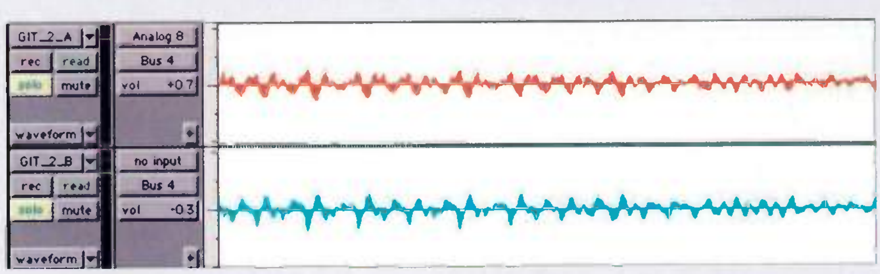
Reviews Editor Mike Senior replies: *Firstly, yes, sound does travel slower through air than along cables. If the mic is a fair way away from the DI'd sound source, discrepancies in timing can be introduced. When mixing DI and miked signals, some engineers would argue that you should match the timing of the two signals to get the best results, but I'd say that this is simply a question of taste. Looking at your screengrab, it looks like you also have a polarity reversal between the mic and DI signals. Again, some would argue that you should therefore flip one of them to match the other, but I'd say that you should just trust your ears — if you match up the timing first, then matching the polarity will matter more, I'd have thought.*

The basic principle is that a phase difference (time delay) or polarity inversion (vertically flipped waveform) between two similar signals has the potential to cause peaks and troughs in the combined frequency spectrum. Whether you happen to like a specific set of frequency fluctuations is up to you. The ability to play with phase for effect is one of the most creative and under-rated aspects of 'real' recording, so do make some time to experiment here. If you're trying to minimise frequency-cancellation effects, then it would be best to match the phase and polarity as closely as you can. Remember, however, that your guitarist may be moving around in relation to the mic, so it probably won't be possible to match the phase of the two signals exactly, as the degree of phase difference won't be exactly the same throughout the track. That said, it should be good enough to avoid any undesirable bass ripples which are the most noticeable side-effects of phase mismatch.

Q When should I use an effect as an insert, and when as a send?

Are there general rules about when it's better to use an effect as an insert and when it's better to use it as a send effect?
SOS Forum Post

***SOS* contributor Len Sasso replies:** *It's probably fair to say that any type of effect you can think of can and has been used both*



Ralf Lehmann's waveforms exhibit discrepancies in phase and polarity.

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The VX351 Voyager CV Expander connects to the Voyager's accessory port, accessing all 21 Control Voltage (CV) and Gate outputs of the Voyager. It also includes two 4-way mults and two attenuators. Adding this to your Voyager adds the capabilities of a modular synthesizer. £199 inc VAT and delivery.

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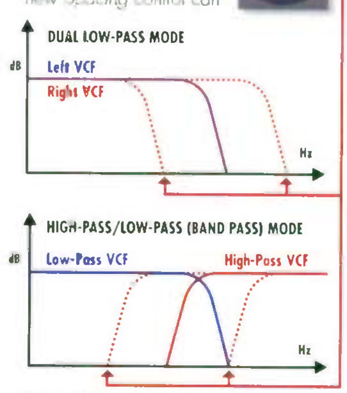
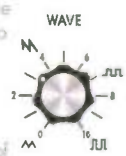


Bob Moog has always favoured the warmth and rawness of voltage control to the predictable uniformity of digital but his VCOs now boasts the best of both worlds with ultra-stable tuning thanks to a new temperature regulation circuit. Sonically, the Voyager's oscillator waveforms are continuously variable so you can morph between waveforms for a huge range of timbres, and in practice, the addition of a dedicated LFO frees an extra VCO for even fatter tones. New Oscillator Sync and Linear FM deliver characteristic metallic & acoustic tones with intuitive tactile control.



The Voyager's two Modulation Busses are many times more flexible than those on the original. Each bus (1 shown) patches from six modulation sources, through four shapers to six destinations via the modulation wheel or pedal controller. The LFO can also be controlled from a variety of external sources including MIDI clock, and Voyager also includes dedicated Sample & Hold.

All sections of the Voyager are equally comprehensive and well thought out. Both filter and amplifier envelopes are four stage ADSR and a five channel mixer includes level control for an external input.

Obvious advances include memories & performance control. The MIDI keyboard is both velocity and pressure sensitive and an all new 3D Touch Surface provides 3 continuous control

signals (x and y coordinates and touch area amount). Control routing and patching possibilities are endless - try using the pad to simultaneously control the level & rate of modulation morphing between waveforms!

This short description can't do justice to a machine that Bob Moog has had in the gestation phase for over a quarter of a century. Call for a brochure or better still check Voyager out in store or on a free trial in your own studio.

VOYAGER™ by Bob Moog

Available in the UK from:

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

media tools
020 7692 6611
12 Flitcroft Street, London WC2H 8DL
email: info@mediatools.co.uk www.mediatools.co.uk

Q&A

ways. But, there are good reasons to choose one method or the other in specific contexts. Effects such as reverb, when used to create an overall ambience — say that of a hall or shower stall — are typically placed as send effects to affect the mix of all audio channels. Effects such as EQ and compression, which are often used to enhance a specific track, are generally used as insert effects. On the other hand, it's not uncommon to insert a reverb effect to create an ambience for a specific instrument, or to place EQ or compression before or after a send effect.

One thing to keep in mind is that send effects don't necessarily need be applied to a mix of channels. You might dedicate a pre-fader send to a single channel for an effect whose level you want to automate separately from the unprocessed signal. Although you could accomplish something similar with the send-level or wet/dry-mix controls, because you're controlling the level of the effect at different points in the signal path, one method might be preferable to the other. For example, varying the input to a delay line produces a very different effect to varying its output.

Another case where you might want to use a send effect is to control the wet/dry mix for effects without mix controls. Typically EQ and dynamic effects don't have a mix control because you only want the processed signal. Should you want the special effect of a wet/dry mix, you can achieve it by placing them as send effects.

Finally, keep in mind that you're not limited to inserting effects into individual channels. For example, in finalising applications, compression, EQ, and limiting are used as insert effects on the master output. As with all signal processing, use what works.

Can I get 3D spatial effects from a plug-in?

Is there a plug-in that reproduces the psychoacoustic spatial effect from the Roland RSS boxes, that gives the impression of surround sound on two speakers and headphones ?
SOS Forum Post

Reviews Editor Mike Senior replies: *The first thing to get clear is that you can't have 3D spatial effects from a stereo system which work on both headphones and speakers. Without going into too much technical detail, getting 3D effects from speakers requires out-of-phase signals to be added to opposite speakers to cancel crosstalk, whereas headphones don't need that because they encounter negligible crosstalk. So you can either have your 3D effects accurately presented on speakers or on headphones, but not on both. Roland's earliest RSS units were designed for speakers only, but the output of later models such as the RSS10 (pictured) can be switched between speakers and headphones.*

The Roland RSS10 Sound Space Processor.

There are numerous software plug-ins available which produce 3D effects, such as Prosoniq's Ambisone *(www.prosoniq.com), Steinberg's* Free-D *(www.steinberg.com), and Wave Arts'* Wave Surround *(www.wavearts.com) to name but three.*

As an aside, I'm not convinced that the RSS techniques are awfully effective unless you're listening in an unfeasibly ideal nearfield monitoring environment, which almost everyone isn't. On the other hand, the RSS Chorus effect in the Roland machines is still pretty cool — it's just as nice on headphones as speakers, although different in each case.

What's the difference between velocity and volume?

Could you explain the difference between MIDI Velocity and MIDI Volume. I'd like to have a Velocity fader to do Velocity automation, but none of my synths or software have that.
Chandra Murphy

***SOS* contributor Len Sasso replies:** *The reason you rarely see Velocity controls is that, unlike volume and other MIDI controller information, Velocity is attached to individual notes. Each MIDI note has a Velocity, which is meant to reflect how hard it was played. It's called Velocity because synthesizer manufacturers discovered early on that it was much easier to detect the velocity with which a key was descending than the pressure with which it was hit. Pressure sensitivity came along much later in the form of Aftertouch and Poly Pressure.*

Inexpensive MIDI keyboards that are not Velocity sensitive do sometimes have a knob or slider for setting Velocity. In addition, most programmable hardware or software synths and samplers have Velocity scaling and offset settings for some or all the parameters to which Velocity can be applied. If you have software that offers some degree of MIDI processing, you can probably set up your own faders to scale and offset incoming note Velocity before it is routed routed to be recorded or to play your synth. Scaling refers to multiplying each note's Velocity by the same amount — halving each note's Velocity, for example. Offsetting refers to adding or subtracting a fixed amount from each note's Velocity. Since scaling changes both the average Velocity and the Velocity range, it's common to use an offset after scaling to preserve the original average Velocity, or, in other words, the original level.

MIDI Continuous Controller (or control change) messages, on the other hand, are intended to communicate the changing value of a fader (volume), knob (pan), or wheel (modulation). That usually means a separate stream of MIDI data for each control, although it's common to have single values inserted in sequences to represent the initial state of those controls. Most sequencing software programs have some form of automation editor to display, create and edit MIDI Controller data graphically. Those editors usually allow you to view and edit note Velocity as well as other MIDI messages, like Aftertouch, Poly Pressure, and Pitchbend.

What's wrong with my power amp?

I've just bought a second-hand power amp and it has a few 'issues'. There's a slight buzz present, even when the outputs are turned right down. Turning the amp up does not make the buzz any louder. It's a 1200W amp so the level of music coming through it will easily drown the buzz out, but I suspect an earthing problem, and if the amp isn't safe to use then it definitely is a problem! There is a switch labelled 'earth link' on the

back of the unit but it doesn't seem to have any effect. My other concern is that the amp sends a large amount of current to the speakers when I turn it on, even with the outputs turned right down — the speaker cones really jump. I find this rather worrying. Could you advise?
SOS Forum Post

Technical Editor Hugh Robjohns replies: *The buzz you're hearing is not uncommon in this kind of amp. It could be caused by an earth loop somewhere in the rest of your system, or in the connection to the amp. However, I notice that you say the earth-lift switch on the amp apparently makes no difference, which suggests that the loop isn't created by the amp connection. Two other possibilities are that the power supplies in the amp are less than perfectly regulated, or that magnetic fields from the transformer(s) are affecting the input stages. A good musical/electrical repair shop should be able to help, but you're obviously going to have to get your wallet out!*

You can easily check the basic safety of the amp by unplugging it, getting a test meter and checking the continuity of the mains earth from the earth pin on the mains plug (the big one in the middle, for those of us in the UK) through to the amplifier chassis metalwork. While you are at it, you can also check the earth-lift switch. When it's in its closed position, the screen connection of the input sockets should be linked through to the mains earth. With the switch in the open position there should be at least 100Ω between them. The fact that your speakers jump when you turn the amp on is quite normal for many older amp designs, and not a cause for concern.

Q How can I restore my old half-inch tapes?

I have some Ampex 456 half-inch tape reels that are around 15 years old. I was hoping to record all of them onto my PC. If only life was that easy! The tapes grind to a halt after around a minute or so when playing on my old Tascam 38. I'm pretty sure that this is because of oxide shedding — when the glue that binds the tape together loosens with age. I've heard that you can bake the tapes in a fan-assisted oven to temporarily fix the problem. Do you know of any professional audio companies that offer this service at a reasonable price? I'm not confident about doing it myself, not least because my oven isn't fan assisted!
Jayne Drake

Editor In Chief Paul White replies: *There are plenty of companies offering audio format-transfer and restoration services, including some who advertise in the back of* SOS. *But if you want to do it yourself, don't put your tapes in the oven as this will be too hot. Ideally, you want to bake the tapes for two days at approximately 50 degrees centigrade. They should then play well enough to transfer, although they will eventually revert back to being sticky, so transfer them as soon as possible after baking. Don't try to play them again until then as the oxide shedding may cause permanent audio dropout. I borrowed a chicken-egg incubator to bake some old tapes of mine (see below) and it worked fine.*

Q How do I set up a condenser mic for vocal recording, and which mic should I buy?

I'm thinking about buying a couple of large-diaphragm condenser mics to record some vocals into *Sonar 2 XL*. I'm going to be re-recording some songs originally recorded live and straight to Minidisc using SM58s. I'd like to improve on the vocal sound while keeping the warm, intimate feel of the original. How far from the mic should the vocalist be, and do I need to use compression? Also, could you help me decide between the Oktava MK219 and the Rode NT1A?
David Johnson

Editor In Chief Paul White replies: *You can record as close as six inches to a capacitor mic providing you use a pop shield and take care over your gain settings. At this distance you should need little or no EQ but compression will certainly help give you that intimate up-front sound as well as keeping the level even. The amount of compression required depends on the voice, but I often choose a hard ratio as opposed to soft-knee setting with a ratio value of between 3:1 and 8:1. Use a fast attack, a release of around 100ms, and then adjust the threshold so the compressor just kicks in on the signal peaks, and reduces the gain by no more than 6 to 8dB. This is just a starting point though, and, in the end, you must judge everything by ear as every compressor and compressor plug-in works slightly differently. Also, take note of the environment you are recording in and make sure there's something acoustically absorbent behind the singer.*

Either of the mics you suggest will produce fine results and the choice of which to go for really depends on which suits the specific voices best. I use a Rode NT1 in my own studio and really like it. The NT1A should have a more extended high end and be even quieter, but if you like that warm, dynamic mic sound without too much high end, an original NT1 might suit you better. SOS

Paul White's Ampex reel-to-reel tapes, ready for baking in a chicken-egg incubator. Delicious!

This month we investigate two new technologies from Microsoft that could change the way we use audio and video for entertainment; plus, as video becomes more pervasive in the music technology market, we look at some of the latest announcements from NAB.

An early prototype of a Media2Go device.

Dave Shapton

I recently went to a Microsoft press event where they were showing two new technologies: Windows Media Centre Edition, which has recently been launched in the US and will shortly be available on these shores, and Media2Go, a concept still at the prototype stage for a personal audio and video player. Microsoft have always been keen to get a version of the Windows operating system on to as many diverse platforms as possible, and these latest technologies aim to establish versions of the Windows XP and CE operating systems in the home entertainment market. In fact, should toasters ever require advanced computing components, you can be sure that Windows For Toasters wouldn't be far behind.

Media To Go

At the press event, Microsoft showed what they readily admitted was an early prototype of a Media2Go device, and although it looked rather like a bloated iPaq, the prototype clearly showed the direction the company is heading with portable entertainment. What distinguishes a Media2Go device from a standard MP3 player is that it can play video directly from an internal hard disk on a relatively small screen. However, the limited size of the screen is actually quite an important advantage in many ways since there's no need to store video at high resolutions, and even the proposed hard disk size of 20GB should therefore hold several hours' worth of video. Among the supported formats will be Microsoft's own Windows Media 9, and video in this format offers better quality than VHS, while taking up less space per minute than a Wave (*.WAV) file.

As with Pocket PC devices, Microsoft don't intend to manufacture Media2Go devices themselves, but rather to provide a reference platform that other manufacturers can build their products around. Among the many partnerships already established for Media2Go, Creative Labs have already announced that they will support the platform with a 'Nomad' branded device — at least in the US (the Nomad brand isn't used for Creative's Jukebox MP3 players in the UK).

However, the companies who do jump on to the Media2Go bandwagon will find themselves facing two fundamental problems. First, typical users are only just getting used to the idea of entertainment media as files rather than disks or tapes: MP3 is enough of an open standard that you don't get bogged down in issues around format or digital rights management to use them, but video is a different matter. The only tool used on a widespread basis to exchange video files is DivX, a video codec based on MPEG-4; and although the group behind the format is taking steps to legitimise itself, I suspect that the vast majority of users are currently using DivX without much regard to matters concerning intellectual property.

Secondly, I put it to Microsoft that aside from the internal hard disk, there really isn't much difference between a Media2Go device and an iPaq, for example, which is also capable of playing media files via its own version of Windows Media Player. Their response was, with all due honesty, to agree, but to add that their research had indicated that people tended to want specialist devices with a specific function, rather than one multi-purpose device to do everything. However, whether this proves to be correct or not, I think Microsoft hopes that the distinguishing factor of a Media2Go device is that it will be running Microsoft software. From my perspective, though, the reality is that in a very short time from now, you'll be able to buy a device that works as a phone, games console, PDA and media player; and the chances are that it will be running Symbian or Linux rather than Windows.

The '10-foot' user interface makes it easy to operate Windows Media Centre Edition from a distance.

Media Centre Edition

Windows Media Centre Edition (MCE) is a version of Windows XP that allows you to control your media player from a remote control while you're across the room from your computer monitor. While this might sound a little tame, I was really taken with MCE when I saw it in action because I've wanted to find a practical way to use a PC as a home media server for a long time now. It's all very well navigating through the files on your computer, clicking on the file you want to play and listening to it on 'multimedia' speakers. But wouldn't it be better if you could sit back on your sofa, in front of a decent hi-fi, and just click on a remote control? That's exactly what MCE lets you do, and I must say that it does it rather impressively, using what Microsoft call a '10-foot user

▶ interface'. This means big writing on the screen, of course, but also a rather clever change in the way that the menus work, which seems completely natural to use and is clearly related to the interface used for Media2Go.

However, MCE isn't limited to playing back audio, as it also features a personal video recorder facility, similar to the less-than-successful Tivo, letting you record programmes that are marked up in an electronic programme guide. While MCE has already been launched in the US, it doesn't work with the digital TV systems we've got in the UK at the moment, which is why Microsoft are delaying the UK release of MCE until they have a solution. Apparently MCE is selling well in the US, which is an important sign that computers are finally making it into the mainstream entertainment arena, and it will be fascinating to see how this affects the design of devices like Sony's Playstation 3 towards being home media servers in addition to games machines.

News From NAB

I've just got back from the National Association of Broadcasters show in Las Vegas, and while Musikmesse and NAMM are dedicated solely to music, you can't beat NAB for a wider perspective on where digital media is heading — even if it does mean visiting a place where all forms of vegetation have been replaced with slot machines. This year, the biggest announcement came from Avid, the default leaders in the video editing (or 'content creation', as it's now known) industry, despite the hypersonic rate at which their main competitors, Apple and Pinnacle, are catching up.

I won't go into too much detail here because we're in the audio and music business after all, but I just wanted to mention the shift in the computing paradigm that's pointed to by the Avid developments. What Avid have done is to externalise their processing, or provide 'acceleration' as they call it, so that if you buy a copy of Avid *Xpress Pro*, (formerly known as *Xpress DV*), you'll have the option of buying a hardware accelerator box called the Mojo. This is a useful box with composite and Y/C (that's S-Video) inputs and outputs, along with multi-channel audio punch-in with low-latency monitoring plus 20-bit stereo analogue I/O, and it's connected to the computer via Firewire.

While there might appear to be nothing remarkable about this on the surface, what isn't at all obvious, until you stop to consider exactly how Mojo does its acceleration, is that it isn't DV (ie. compressed) video that's going up and down the Firewire cable, but raw, uncompressed video. If you're not familiar with video data rates, this is possibly not the most impressive thing you've ever heard, but to provide some indication of its signifance: uncompressed 16-bit stereo audio sampled at 44.1kHz has a data rate of around 1.4Mbits/s. DV, the format used by most digital camcorders, on the other hand, has a data rate of 25Mbits/s. This sounds like a large amount of data until you realise that the capacity of Firewire is 400Mbits/s, and although uncompressed standard-definition digital video has a bandwidth requirement of 270Mbits/s, this is still within the capabilities of Firewire.

Don't go losing your... Avid's Mojo accelerator being used with Avid *Xpress Pro*.

So where's all this leading? Well, what this suggests to me is that the notion that within a very short time all DSP would be done by the host computer's processor has taken a bit of a side turning, and that dedicated hardware processing still has a role to play. But anyone who still makes dedicated PCI processing cards must be feeling very worried by these announcements. Consider the advantages of external processing in areas such as configuration: if your computer supports Firewire and you have the right software then you can just plug it in and know that everything will work. No more IRQ or DMA worries, no resource battles, and it'll work with notebooks and small-form computers.

What's even better, though, is that music technology has led the way. I'm thinking of the forthcoming products like TC's Firewire Powercore and Yamaha's 01X, a digital mixer so closely integrated with its host PC (with a great deal of I/O bandwidth courtesy of mLAN) that the two devices virtually merge together into one distributed computing platform. Suddenly, who cares about the PCI buss? SOS

Sony Acquire Assets From Sonic Foundry

The rate of progress is speeding up, and if you have the slightest doubt about this look at Raymond Kurzweil's site at www.kurzweilai.net. In fact, as I was writing this month's column, a colleague emailed a press release with the shock news that another audio company has effectively been bought by a giant corporation. Sony Pictures Digital have paid $18 million to acquire the assets for Sonic Foundry's desktop software, which includes products such as *Vegas*, *Acid* and *Sound Forge*.

My reaction to the acquisition is sadness because a company that three or four years ago seemed to be doing all the right things should be swallowed up for what seems to me a pittance, when you consider what good products they have. Of course, it's a good move for Sony — *Vegas* alone should have been worth a multiple of what Sony paid for it, as it's a totally original video editing application in a market awash with Avid clones.

If you're already an Avid editor, you'll find *Vegas* tricky at first; but *Vegas* is particularly suitable if you're coming from an audio background, since *Vegas* even has a video output buss. This means that if you want to add an effect to your whole production, such as Bleach Bypass, the futuristic and slightly disturbing colour effect used throughout *Minority Report*, for example, you apply it to the video buss just like using a send and return on an audio mixer. And since *Vegas* grew up as a multitrack audio tool, the audio facilities are superb, supporting high sample rates and resolutions. The latest version even has ASIO support, and, as far as I know, it's the only application that will allow you to play Windows Media Video, MPEG-1, MPEG-4 and DivX, in addition to MP3 and mixed-sample-rate audio, all on the same timeline in real time.

I hope Sony continue to develop Sonic Foundry's software, and that they will sell the products at a realistic price, rather than be tempted to give them away with their Vaios or even port them to the Playstation.

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FMR Really Nice Preamp

Mic Preamp

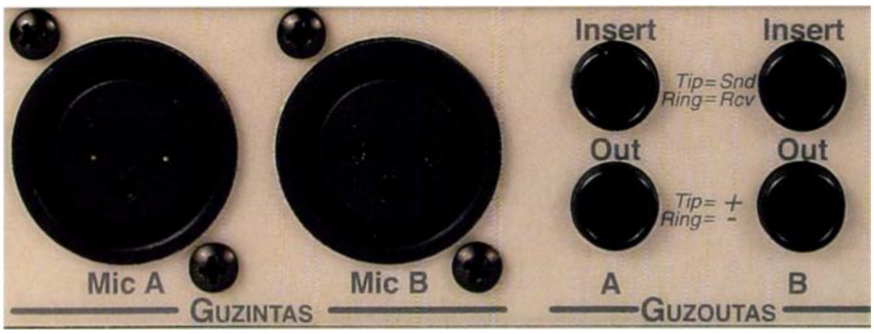
FMR's dual-channel mic preamp offers a transparent sound at a project-studio price.

Paul White

After seeing how impressively FMR's Really Nice Compressor performed during our review, I couldn't wait to get hold of their mic preamp, which has an interesting design philosophy. The designers reckon that most project studio owners will have access to high-sensitivity capacitor microphones, so they decided that having plenty of headroom was of paramount importance — their circuit can take a massive +27.5dB without clipping, which is over 24V. Moreover, while this preamp is electrically quiet, the designers made clarity and transparency a higher priority that ultra low noise.

The preamp circuit is a fully Class-A instrumentation-type amplifier capable of working up to 100MHz and beyond. It is designed to be self-biasing, with very good common mode rejection, and, because it can accept levels as high as +28dBu (balanced), it can also accommodate line-level signals via the XLR input. The EIN of -120dB doesn't seem quite as impressive as some designs when you see it written down, but there's more to noise figures than meets the eye, and what tends to matter more is how the noise performance holds up at typical gain settings rather than at full gain. Many coveted classic mic preamps are also noisier than the RNP, so perhaps focusing on clarity and lack of coloration was the right approach.

Like the RNC, the RNP is remarkably inexpensive in the UK for the performance it offers, and to make this possible it comes in a no-frills metal case and is powered by an external adaptor. You get two channels of mic/line/instrument preamp for your money, and while the features are limited to a gain control (actually a stepped 12-position switch), phase invert and independently switchable 48V phantom power (with a slow rise time to avoid pops), you do get both jack and balanced XLR outputs and TRS insert points. A pair of high-impedance jack inputs on the front panel accommodate line and instrument signals, while all other connections are on the rear panel. The RNP can provide a +22dBu unbalanced output and a +28dBu balanced output simultaneously, which should keep even the greediest soundcard happy. Both the mic inputs and the XLR outputs are electronically balanced, though they can be used unbalanced simply by plugging in unbalanced cables. The input stage incorporates a third-order EMI filter (designed to attenuate RF) which is optimised for a 150Ω balanced source such as a typical studio microphone.The metering is a touch rudimentary, relying as it does on just three LEDs, but all the switches have miniature



SOUND ON SOUND

FMR Really Nice Preamp **£499**

pros

- Very low distortion with wide audio bandwidth.
- Clear, natural sound.
- Plenty of headroom for driving level-hungry soundcards.
- Inexpensive.

cons

- External PSU.
- Fairly basic level metering.
- No low-cut switch.

summary

I think that the right compromises have been made to deliver truly professional audio quality in a no-frills, low-cost package.

status LEDs and the gain controls are calibrated in accurate 6dB steps. There are, however, no low-cut switches, which could be inconvenient given that many budget capacitor mics also dispense with a low-cut filter.

Another feature that shows consideration for the user is that when the phantom power is switched on or when the unit is powered up, the outputs mute briefly to prevent pops and bangs over the monitors. The mute is implemented before the insert send, so externally connected devices are also prevented from receiving switching noises. The stepped gain control was chosen to ensure accurate gain-setting — important for stereo use or multi-channel recording — and the simple metering indicates signal present, +18dBu, and clipping (actually 1dB below clipping). To make clips easier to spot, the clip LED stays on for three seconds once activated. Surprisingly, the metering is controlled by a small microprocessor, which also looks after power supply management and the buttons.

Really Nice Or Really *Really* Nice?

Once plugged in, the phantom power LEDs flash for a few seconds to show the unit is muted, then you're ready for action. A similar flashing and muting routine takes place when the phantom power is switched on. Using an Audio Technica AT4033 mic as the test subject, and comparing the preamp with that found in a competent budget mixer, I could discern little subjective difference in the amount of background noise (which was more than adequately low in both cases), but tonally the RNP had a more open, better integrated sound — straightaway it sounded 'right'. By contrast the mixer's preamp made the mic sound slightly honky and less detailed. While the lack of flattering (or otherwise) coloration might not appeal to those users always on the lookout for a new magical ingredient, I liked it very much, because the better the sound you get at source, the less need there is to try to rescue it later with EQ or other processing.

The Really Nice Preamp's stepped gain pots allow you to match gain settings between the two channels precisely.

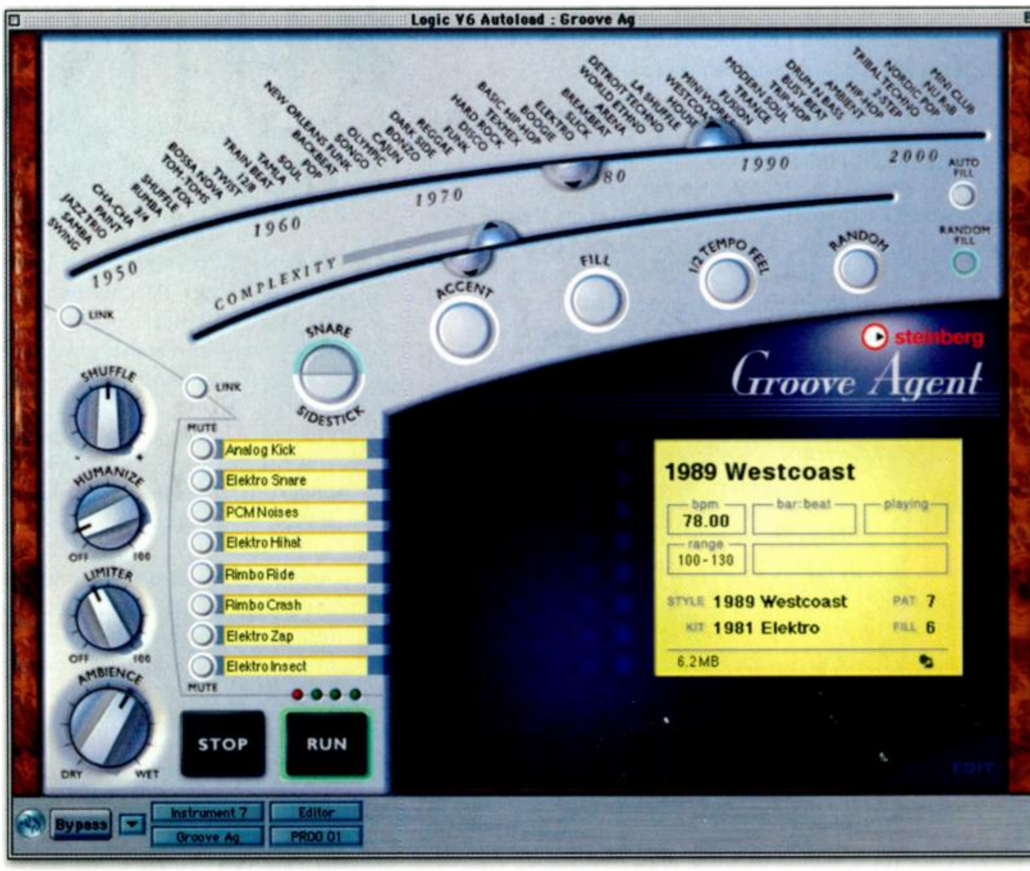
The RNP combines innovative circuit design with some well-reasoned corner cutting. The stated aim is to deliver really high-quality sound at a low cost, but without frills and without striving for a noise specification that would be unnecessary with most of today's high-sensitivity capacitor microphones. I think the RNP achieves that goal, and given its low UK price, it's something of a bargain. SOS

information

- £ £499 including VAT.
- T Audio Agency +44 (0)1908 510123.
- F +44 (0)1908 510123.
- E info@audioagency.co.uk
- W www.audioagency.co.uk
- W www.fmraudio.com

Steinberg Groove Agent

If you don't fancy programming drums from scratch, but you find pre-recorded drum loops too inflexible, perhaps what you need is Steinberg's virtual drummer...



Virtual Drummer VST Instrument For Mac & PC

Paul White

Groove Agent is a VST Instrument designed to do for drums what *Virtual Guitarist* did for guitar parts; and like *Virtual Guitarist*, it was produced by Sven Bornemark in collaboration with some great musicians and rather clever programmers. All the acoustic drum sounds were recorded in Sweden by top drummer Mats-Erik Bjorklund, whereas electronic sounds came from Primesounds in Stockholm. In some respects, *Groove Agent* is like a software drum machine, but rather than using REX file samples like *Virtual Guitarist*, it uses separate drum samples arranged into patterns by 13 different musicians specialising in different musical styles. Unlike a drum machine, *Groove Agent* can be played interactively, with a MIDI keyboard used to switch between related drum patterns of differing complexity, or to initiate fills and so on.

In order to provide plenty of sonic flexibility, the drums were recorded simultaneously with close mics, overheads, distant mics at 2 metres and distant mics at 7 metres. This is important, as *Groove Agent* includes a variable ambience control that uses this information to move from a dry drum sound to a very live room without using synthetic reverb.

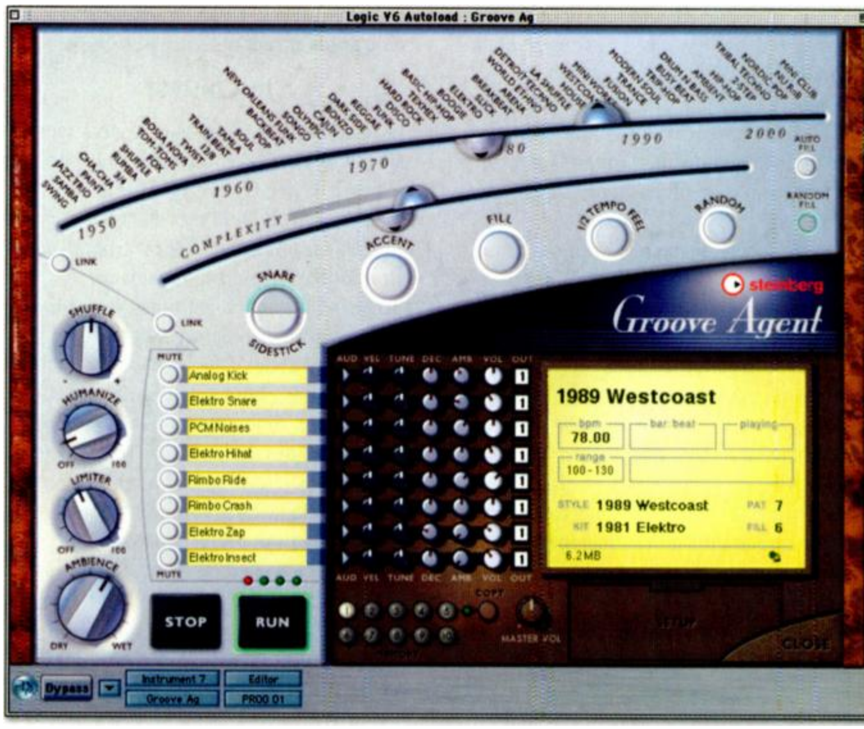
Getting Into The Groove

Groove Agent runs on any recent Mac (OS 9 to 10.2) or PC; there's no Audio Units version, so *Logic* OS X users will either have to wait or try a VST-to-AU wrapper. While *Groove Agent* doesn't demand a cutting-edge computer, it does take up lots of memory, so the minimum recommended is 512MB. Installation takes around 300MB of hard drive space and is from a copy-protected CD-ROM.

In *Logic*, *Groove Agent* is inserted in an Instrument track in the normal way and comes up with a main window that looks like a retro drum machine. This has a Styles slider at the top and a Complexity slider below it. Closer inspection reveals that each slider has two handles that are normally linked, but they can be unlinked, the upper one allowing drum patterns to be played using kits other than those for which they are designed, while unlinking the lower handle on the lower slider allows different fill to be used than the ones that normally go with the selected pattern. Each drumming style features 25 related patterns of various complexity, which may be selected on the fly using a couple of octaves of notes on a MIDI keyboard.

Below the Styles and Complexity sliders are four knobs to control Shuffle percentage, Humanization (a kind of virtual lager knob, though to be accurate, there should be a sampled sound of a drummer falling off his stool when you get up to 11!), Limiter and Ambience. Limiter works like a normal studio limiter to squash peaks and increase the overall level, while Ambience brings in the distant room mics for a big, live room sound. The type of ambience added is optimised for each kit.

Eight drum kit groups are also furnished, along with mute buttons, and of course there are Start and Stop buttons. When used with a sequencer, the Start button can be used to 'prime' *Groove Agent* so that it starts when the sequencer starts. When *Groove Agent* is running, four on-screen 'LEDs' show that the pattern is stepping through. There's also a virtual LCD screen showing the style type, tempo (as read from the host software), suggested tempo range, currently playing pattern variation and the amount of memory being used. Further buttons enable the snare sound to be switched to sidestick and allow accents to be added manually in real time. Accent triggers a kick and crash together, and if you hold this button down for around a quarter note after you hit it on an off beat, it produces a realistic syncopation. A dedicated button brings in fills, while another strips down the pattern to give a half-speed feel. The Fill button is interesting because if you hit it part of the way through a bar, you only get part of the fill, which provides a useful degree of extra variation. Patterns may also be started with a fill by 'arming' the Fill button before starting playback. Selecting Random wanders through pattern variations based on two patterns either side of the complexity level selected (randomly, as you might expect), while Random Fill does a similar thing for fills, selecting randomly from fills two patterns either side of the selected one whenever you trigger a fill. An Auto Fill button is also available and brings in a fill whenever you switch from one complexity level to another.



If you want further control over your drum sounds, you can individually adjust parameters such as tuning and decay time.

The panel surrounding the 'LCD' window can be opened by pressing the Edit tab to reveal extra functions including a mixer-like matrix that allows you to audition and edit individual drum group sounds. The velocity levels of the parts can be scaled, drums or drum groups can be tuned up or down by up to an octave and the decay and ambience of each drum can also be adjusted along with its volume.

Groove Agent also has a 10-slot memory that can be used to store snapshots of the panel settings for later instant recall. If a memory button shows dark it is empty, while one glowing red contains data and a brightly glowing one is currently active. Memories may also be copied from one location to another so that refinements can be made using the panel controls before resaving them.

Also lurking under the lid is a further hatch called Setup. Lift this and you see a few wires and switches as well as a Reset All button. The four switches activate MIDI Output, GM Output, Ambience to Output 4 and Vintage Mode. Used in *Cubase* or *Nuendo* (it may work in some other sequencers but the result is not guaranteed), *Groove Agent* can output MIDI data corresponding to the drum patterns enabling you to trigger your own samples, but sadly this feature isn't available in *Logic*. GM switches the drum output data to conform to a GM map while 'Ambience to Output 4' routes the dry drum parts to outputs 1 to 3 and the ambience to output 4 (assuming your VST Instrument host supports multiple outputs) where it can be processed separately. Finally, Vintage mode narrows the stereo width and uses EQ to warm up the sound. *Groove Agent* also functions as a drum sound playback module, and sounds within each kit are mapped to MIDI notes. The Reset All button lets you clear out the memory and set the controls to their default positions.

Using *Groove Agent*

Once I'd managed to get *Groove Agent* to work in *Logic*, getting impressive results out of it was no problem. The drum styles are arranged along a timeline starting in 1950 and they encompass Latin, Big Band, Jazz, Shuffle, Tamla, Soul, Pop, Funk and Disco as well as a number of rock genres and Reggae. Newer styles such as Hip-hop, Breakbeat, Ethnic, Techno, House, Trance, Modern Soul and Trip-hop are also represented, to name but about a quarter of them. All these beats have 25 further variations ranging from over-simple to

SOUND ON SOUND

Steinberg *Groove Agent* £180

pros

- Emulates the nuances of a real drummer without any complicated programming on behalf of the user.
- Generally well played and well recorded drum samples.
- Variable ambience is particularly impressive on the rock drum kits.

cons

- You have to work with the rhythms you get, which may involves some modification to your bass lines to make them fit perfectly (though you can always mute the kick track and add your own).
- Minor compatibility issues with *Logic Audio*.

summary

As a first step into software drummer emulation, *Groove Agent* is impressive. It is easy to use and sounds much more convincing than a typical drum machine.

seriously over-played, and the trick is to mix and match them throughout a song to get a natural feel. A different fill goes with each complexity variation, and as remarked upon earlier, you can randomise the fills that are applied and add humanity to the timing (another form of randomisation) to loosen up the feel.

On the whole, the acoustic kit sound is impressive, though I found the sameness of ride cymbals within any pattern that used them eight to the bar soon became wearing and made the patterns sound more drum machine-like. Randomly selecting from a handful of similar but not identical samples would have improved this. Each of the genres is represented with a viable playing style, but despite the huge range of options, I found it quite difficult to find 'normal' rhythms that a real drummer might use when playing blues, rock & roll or straight-ahead pop. In some cases I felt that adding more variations on a theme rather than more levels of complexity might have been more useful, especially in the area of kick drum patterns, which are very important when tying in with a bass player. To be fair, this is what happens on some patterns, such as the 3/4 time variant, but I felt that in other instances many of the variations were needlessly fussy, almost as though the drummer/programmer was trying really hard to come up with 25 variations on each pattern. The same is true of the more complex fills — most people would probably never use them, so why not have more varieties of usable fills? Of course products like this are upgradeable so, based on user feedback, these are issues that could easily be addressed.

Groove Agent And Logic

I used Emagic's *Logic* running under Mac OS 9.2 for my tests, and although the installation appeared to go flawlessly, when I tried to open *Groove Agent* I was greeted with a message showing me a file path and telling me that a key file could not be found. This is very reminiscent of what happened in *Logic* with early versions of *Virtual Guitarist*, so I emailed the manufacturers for advice. I got a reply almost immediately to the effect that this is a known *Logic/Groove Agent* problem for which they are developing a proper solution, but in the meantime, they suggested a workaround. This consisted of using *Simpletext* to write out the file path to my Vstplugins folder and naming the file '*Groove Agent* location'. This, I was told, was to be placed in the Mac's Preferences folder. I'd never heard of this kind of trick before, and to be honest I thought it smacked of trying to cure a clapped-out car by writing 'Please Start!' on a piece of paper and jamming it through the radiator grille. Anyhow, I went along with it and to my utter disbelief it worked first time!

In Control

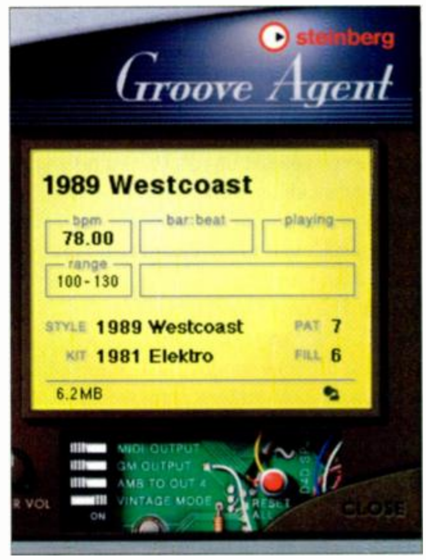
These admittedly subjective observations aside, it doesn't take too much skill with a keyboard to get *Groove Agent* sounding like a real drum track, and the effect of adjusting the ambience to bring up a real room sound just adds to the realism. In general I preferred the sound with the vintage mode switched off, but I can see that it would suit a number of styles. The trick is to switch between similar complexity variations throughout a song, adding fills at the point where a real drummer would. This way the track sounds effortlessly natural. You only have to decide where you want changes — *Groove Agent* can decide what sort of changes they should be if you don't want the responsibility!

Everything that can be accessed from the on-screen front panel can also be controlled over MIDI using Continuous Controllers to operate the various switches and buttons as well as to control levels, ambience depth and so on. The mod wheel can be used to trigger a fill, though Controller 66 does the same thing and MIDI note B3 stops *Groove Agent* playing. MIDI notes B0 to A3 play the internal sounds so you can add to the patterns if you wish, and if the MIDI channel is set to 10, the mapping changes to GM mapping. Where the MIDI channel is an odd number, C4 to C6 select the complexity level, and if you hit with a velocity of harder than 90, you can trigger a fill, which is pretty intuitive. Alternatively, if *Groove Agent* is set to run on an even-numbered MIDI channel, the white notes C4 to B4 may be used to mute and unmute individual sound groups, while the black keys above C4 select the 10 memory locations.

While all these options could be confusing, it's a simple matter to stick with the one that suits you and make that your way of working. I chose to leave *Groove Agent* running on MIDI channel 1 so that I could use two octaves of the keyboard to select complexity levels and to trigger fills, and in most cases, the random option produces good results by switching between subtly different complexity levels. No operational problems were discovered other than those peculiar to *Logic*.

Conclusions

Groove Agent is probably the first serious attempt to make a virtual drum machine instrument that sounds, or more importantly, plays like a real drummer without the user having to do an awful lot of programming, and in this respect it succeeds well on most counts. My own view



Beneath the Setup hatch lurk further controls which can be useful if, for instance, you want to treat *Groove Agent*'s ambience separately from the actual drum sounds.

is that there are too many 'frilly' pattern variations and not enough real-world 'bread and butter' rhythms, but the way the pattern variations are handled to create a realistic sound is most impressive, as are the various ways in which fills can be introduced. The installation problem with *Logic* is easily got around using the modified preference file as described in the box, left, though the sooner this is fixed properly the better.

By way of sound, there's plenty of variation amongst the acoustic kits, all with adjustable ambience, and the overall quality of drum recording is excellent. The choice of electronic sounds is also creditable, and because you can mix and match styles and drum kits, there are many more variations than you might at first imagine. If you like the idea of real drums but prefer the convenience of software, then *Groove Agent* is a must-have package, though expect to have to bend some of your song styles slightly to fit in with the drum rhythms on offer. Ultimately *Groove Agent* is exactly what many people have been waiting for — a drummer that does as it is told and plays in time, doesn't play between songs, doesn't try to get off with your girlfriend and doesn't throw the TV set out of the window. SOS

information

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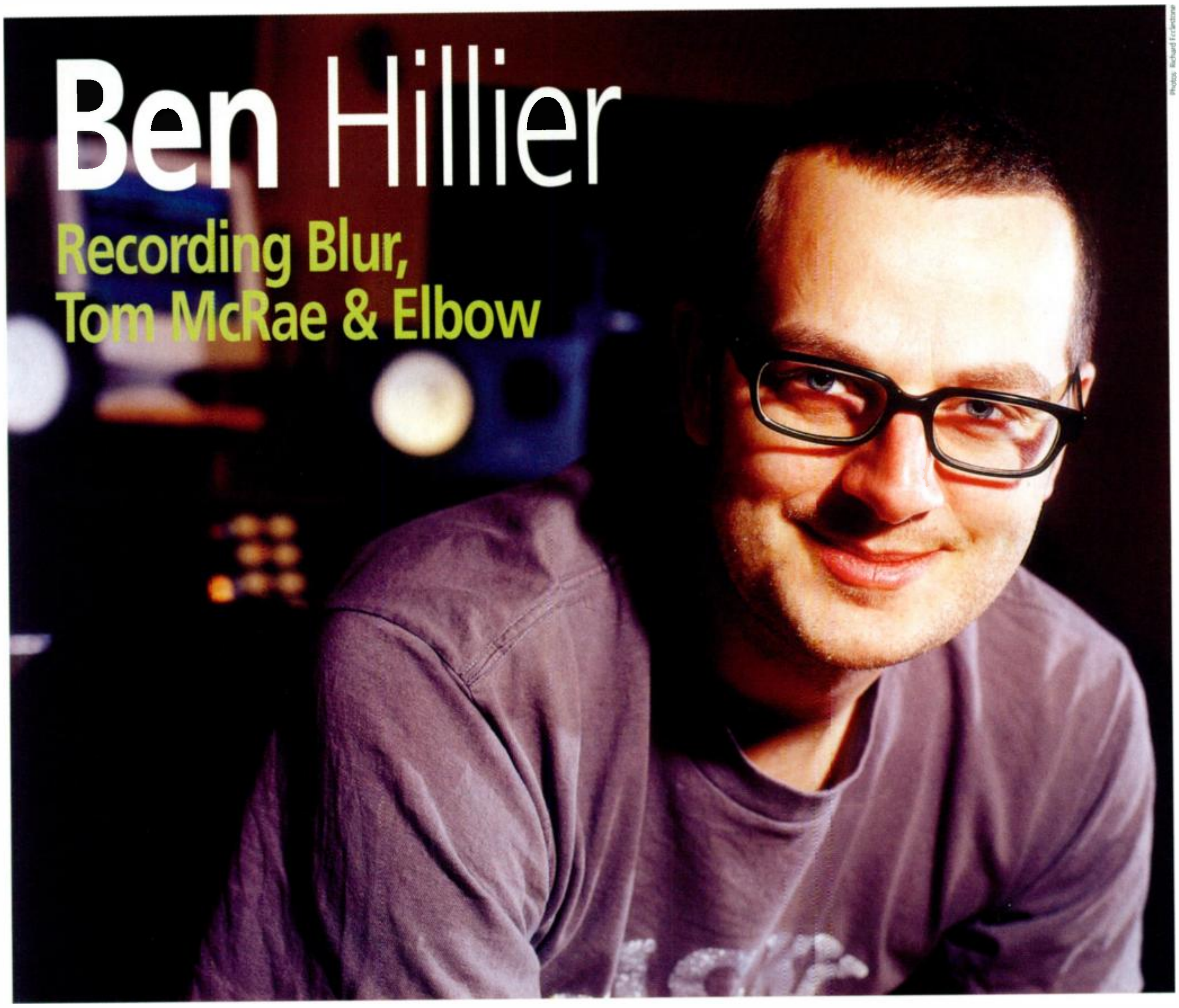
Ben Hillier

Recording Blur, Tom McRae & Elbow



David Greeves

Blur's latest album has taken the band 18 months to make, commuting between London, Devon and Morocco. Somehow, producer Ben Hillier also found the time to produce new records by Elbow and Tom McRae...

This summer saw the release of Blur's seventh studio album, *Think Tank*, co-produced by the band and British engineer/producer Ben Hillier. It was a project both Hillier and Blur had been working on since November 2001, when recording began in the band's West London studio, 13. Their first long-playing release in four years also swiftly became their first album as a three-piece, with the departure of guitarist Graham Coxon soon after sessions began. Rumours abounded that the band were splitting up, but according to Hillier, Coxon left on good terms. "Graham's an awesome guitarist and an amazing musician, but he wasn't really that interested in making a Blur album. His solo stuff is what he's into doing and Blur was becoming a bit of a chore for him, so it didn't seem right to have him involved — we're not at school!"

As a consequence of the reshuffle and the remaining band members' numerous commitments outside Blur, a real sense of urgency accompanied these early sessions. "When we started the record, the choice was to either sit around and wait for everyone to turn up, which wasn't really going to happen, or to get on with doing the record in the time we had. So we said 'Here's the two weeks we've set aside to work on it. We start at 10 and finish at six. If you're there you get to play on it, if you're not you miss out'. So everyone was really hungry to play on the record. If Damon [*Albarn*] wasn't there and we wanted to record a vocal then Alex [*James*] would sing, or if Alex wasn't there then Damon would be desperate to get on the bass. It was quite competitive but really exciting. Everyone had this real drive to play." Coxon does play on one of the tracks on the album (and two more which weren't used), while the rest of the guitar-playing duties were taken by Albarn himself.

The initial two weeks of recording in 13 were followed by another fortnight at the end of January 2002. "We were mostly working from demos Damon had done on his four-track. He'd done some work on them in *Logic* with [*13's in-house engineers*] Tom Girling and Jason Cox, so when it came to starting the recording, they were ready to work on and me, Jason and James Dring [*assistant at 13*] set about it with the band. I don't usually work with an engineer, I usually do it all myself, so it was nice — especially nice to have Jason. He's an awful lot more than an engineer, he's worked with the band for years, he knows 13 and the band's equipment inside out. By the end of the four weeks we had 17 songs — quite an astonishing work-rate considering that the band won't work past 6pm when they're in London."

In At The Deep End

Work on *Think Tank* began again in earnest in June 2002. "We went back to 13, tracking, doing overdubs and reworking what we'd already done, and all the time new songs would be popping up — I think we had 28 of them at one point." Hillier and the band also spent time working with other producers: "The Dust Brothers came in for a week, and Norman Cook came in for a couple of days. The Dust Brothers did good work but they probably came in a bit early in the process. I don't think we ended up using any of the things they did in the end, but that wasn't a reflection on them at all. It was more a question of them turning up at a point when we weren't quite sure what we were doing, and if anything they showed us that we needed to do a bit more work on the writing."

The band struck up a good working relationship with Norman Cook, who

From Ebay With Love

Ben Hillier is a self-confessed addict of the Ebay on-line auction site, and has recently been snapping up second-hand mics from the Russian manufacturer Lomo. Like their more famous cousins Oktava, Lomo were a large and diverse technology company during the Soviet era, manufacturing everything from missile guidance systems to microscopes; perhaps their best-known product worldwide is the small point-and-shoot Lomo camera (www.lomography.com). Though production of their high-quality valve microphones ceased in the early '90s, the mics still change hands over the Internet. Hillier has a pair of 19A18s, miniature bottle-style mics, and his 19A19, with its distinctive space age design, is used extensively for vocals. Hillier is also the proud owner of a pair of RTT MLK101s, Russian-made valve mics constructed with a combination of vintage and modern parts. The RTTs are a particular favourite for miking up guitar cabinets: "Getting that 'guitar amp in a room' sound is one of the hardest things. When you're in the room with the amp and the guitar's really loud it sounds amazing. Then you'll put an SM57 on it and go back to the control room, and the sound just isn't quite there. But put an RTT six or eight feet away from the amp, or even further, and you get this great 'fronty', middly sound." The web site www.microphones.ru is an informative, if idiosyncratic, information resource on these Russian manufacturers.

The fruits of Ebay: a Russian Lomo 19A19 microphone.

A selection of Ben Hillier's esoteric analogue gear, including his EMS VCS3 and ARP 2600 synths, Lovetone Meatball and Electro-harmonix Big Muff pedals.

rejoined them later in the project, and also renewed their acquaintance with William Orbit, who had co-produced their previous album, *13*. "We sent a couple of tunes to William to work on in his studio, working round the clock in a computer environment the way he does. He's a nutter and works all night. That was quite an interesting juxtaposition, us doing office hours then going to see William after work, just as he was getting up!"

Marrakesh Express

The decision to relocate the recording of *Think Tank* to Marrakesh, following in the footsteps of The Rolling Stones and The Beatles, was prompted by a visit to Morocco by Blur's frontman. "Damon had been out there to a music festival one weekend and was really excited by it. The musical life in Marrakesh is amazing. Live music is everywhere. It's a very important part of the culture, in a more direct way than it is over here. Here it's all strictly disseminated via radio and records, and DJs and bands. Over there live music can happen anywhere, and usually does. And they'll write songs about current situations all the time and not think twice about it. It's the sort of living folk tradition that we used to have here, and probably still do somewhere."

Considering Damon Albarn's recent forays into world music with *Mali Music* and chart-topping dub-rock with the Gorillaz project, many expected one or other of these musical directions to be reflected in the new Blur album. While *Think Tank* is certainly the band's most eclectic album to date, with some strongly groove-based tracks, not to mention some very 'un-Blur' moments, a North African flavour is less clearly discernible. Hillier sees the band's chosen surroundings as having a different kind of influence on the record. "There was quite a lot of Arabic influence in Damon's writing, though not really in an obvious way. If you listen to a lot of the scales he's using, and to the structure of some of the songs, a lot of that has come from Arabic music. A lot of the songs on the album aren't just straight verse, chorus, verse, chorus, middle eight, chorus and so on. There'll be one whole bit, then something in the middle, then it'll change into something completely different. Or it'll just keep going round the same thing and gradually building."

Before the band could take advantage of Marrakesh's attractions as a place to record, there was the small matter of transporting an entire studio there. "Basically, I hired a studio's worth of gear from a company called Tickle and put it in a truck. We needed to set up a whole studio as the plan

Like many engineers, Ben Hillier has a rack of favourite gear that accompanies him on all his jobs. From top: Auratone speakers, ADL 1500 and Smart C2 compressors (both hired from Tickle), Cranesong STC8, Tube-tech LCA-2B and Yamaha 2020 compressors, Sherman Filterbank, Drawmer DF330 noise filter, Ashley SC63 EQ, SPL Transient Designer, three old Tweed mono compressors and Hillier's Gates Level Devil broadcast compressor.

was to record and mix in Marrakesh, and we did do a fair amount of mixing out there in the end. I hired a big 40-channel Amek Mozart desk with the really nice Neve EQ on it, a pair of Genelec main monitors and three sets of mini-monitors. We took half-inch tape machines, my full Pro Tools rig, two racks of outboard from Tickle, one from 13 and my rack as well, and cabling for a whole studio. Just so we could rely on it, we ended up taking our own electricity sub-station too!"

The only thing that had been forgotton was the required clearance from the Moroccan customs: "We were out in Marrakesh for five weeks and I spent the first six days trying to get the equipment out of customs! We had a truck with nine and a half tons of gear in it stuck in customs. It didn't even strike me what a stupid idea it was until that happened. They kept on finding new reasons to stop it coming out — 'You haven't got this form or that form filled in,' or 'We need to talk to the Ministry of Culture,' and so on. Five days in we were thinking we might have to go home, so when the equipment did finally arrive we were really up for it."

The arrival inspired a new song, initiated by drummer Dave Rowntree drumming on the flight cases as they were being wheeled off the truck. The end result was 'Morrocan Peoples Revolutionary Bowls Club', one of the tracks that made it onto the album. "I started recording it with my [*Digidesign*] M Box. I had my laptop out there because we knew we'd have to wait for a while for the equipment to get set up, though not as long as we had to wait in the end! Tickle had sent a tech team out with us to help set everything up and test everything, and to fix the desk, because obviously when you move a desk that size that far, it turns up and there's loads of faults in it, so one of the guys spent a whole day wiring the desk back together. 'Morrocan Peoples Revolutionary Bowls Club' turned into a 'test the gear' tune, and as soon as we got the desk up and running we recorded some proper drums and it all went off from there."

Marble Madness

The villa the band had rented provided some interesting venues for recording. "It had this little concrete building based around a covered courtyard and we used the courtyard as a live space. Like everywhere else in Marrakesh, it had a marble floor, tiled walls and a concrete ceiling so it was really, really loud with a really banging reverb. The drum sound for 'Crazy Beat' is just the effects of that room. All the strings on 'Out

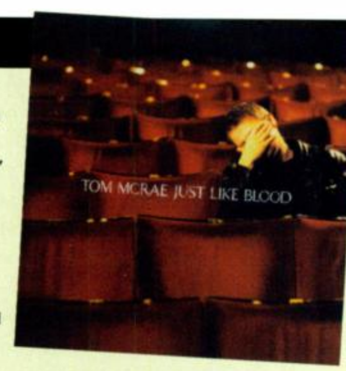
Hillier's pair of reconditioned Roland RE301 Tape Echo units and, just visible to the right on its side, an Echopet solid-state analogue delay used to create the slap-back echo effect for Tom McRae's vocal on 'Overthrown'.

Of Time' were recorded in there as well — the acoustic was really lovely for it. Unfortunately the control room had almost the same acoustic! There was a clay wall behind the main monitors that was about three inches thick and if you turned up the monitors and coughed into the microphone, it made a sound like an 808 bass drum!"

Despite these less than desirable working conditions, several tracks were still mixed on location as planned. "'Out Of Time' was mixed out there and it sounded great when we got back, but that was probably more luck than judgement. We took a load of acoustic panels out with us that we glued to the walls, which controlled the top end a bit as long as you didn't turn the volume up. We were working quietly on NS10s and Auratones, and we'd turn up the main monitors just for the vibe really, because when you turned it up it was very difficult to tell what the bottom end was doing. You just had to be careful, fly by the meters a bit, and try and guess what was happening down there."

The alternative to the cavernous-sounding live space was to record outdoors: "The acoustic in that room was mental and we wanted to get a much drier sound. The

Producing Tom McRae's *Just Like Blood*



While all the members of Blur had their own competing commitments, Hillier had a few of his own. In the lengthy gaps between Blur sessions, he was working on Tom McRae's album *Just Like Blood* in The Dairy Studios, his home-from-home in South London. Just as work on *Think Tank* began with Damon Albarn's demos, Tom McRae's own demos, recorded at home using a Digi 001 system, were the starting point for the Dairy sessions. Often elements of these demos were retained in the finished tracks. "A lot of the time we'd take the vocal from Tom's demo and then build the rest of the track around it. Then when we came to re-record the vocal we'd realise we didn't need to, because we'd built the whole track around the vocal and it fit perfectly. So a lot of the vocals were done without us even noticing! We'd go back and fix certain bits, either re-sing them or tune them a bit, but only tiny bits really. It's a very atmosphere-based record and those original takes added that bit of rawness that was great. There were a couple of tunes where we spent a long time re-recording the vocals and it didn't really pay off. Because Tom's songs are very vocal-led, that's the way we were doing the record, with whole songs built around one specific vocal take. If you then take that vocal away and put a more considered performance in its place, you lose a lot of the nuances, like breathing in a certain place, or where he's finishing each word, and it just sounds wrong. It sounds transplanted from somewhere else. So usually we'd go back to the demo vocal or the guide vocal."

Most of the tracking and overdubs on the album were recorded not in the Dairy's main studio and live room but in Hillier's small programming room, including the drums. "I was just using my little Neve 542 sidecar, which I've had modified to have individual outs on each channel. It has nice preamps and a very basic EQ, which is good becuase it stops you using too much. Then I'd record straight into Pro Tools, or through compressors and in. If I needed more ambience, I'd just open the door to the corridor and stick a mic down the end. It makes for a more colourful sound." Indeed, for all but the most way-out of sounds, he seems to avoid using effects units altogether. "I tend to do a lot of effects with the way I mic things and the way I'm compressing them. I've got an old Gates broadcast compressor which, according to the bloke I bought it from, used to belong to NASA and was used on the Apollo missions. It's a very old bit of gear. It's effectively a limiter/expander. You'd put it before your broadcast transmitter to stop the level going too high and blowing the transmitter, but the way I use it, it distorts like hell. Because it's designed to work from presets, the controls were originally at the back and I had it modified to have two controls on the front panel. It just crunches stuff up — you can hear it on the bottom end of the kit on 'Line Of Fire'. Dave Fridmann [*the Flaming Lips' producer*] has got a couple of these."

Hillier takes an equally anarchic approach to microphone placement. "For 'Stronger Than Dirt' I taped transducer mics to the drums and used a pair of Soundman OKM binaural mics for ambience. They're the in-ear omnidirectional mics designed for binaural recording but using your own head instead of a dummy head. The one problem, of course, is that if you're trying to play along to headphones you get rather a lot of those if you've got mics in your ears! So I tend to use them just gaffer-taped to a chair. You don't get the same stereo image, but they're still pretty good." He then put the output from the binaural mics through the built-in auto limiter on his Minidisc recorder, ignoring the pair of UREI 1176s sitting unused next door. Hillier has also been known to use the brick-wall limiter on a dictaphone in the same way. "There's no way to adjust what it's doing but I quite like things like that. They've got a certain sound to them and they either work or they don't." Another favourite is his ancient Brennel valve quarter-inch tape machine. "The recording head is broken, but I just use it as a preamp. The guitar sound on 'Overthrown' is just Tom's Tele plugged straight into the Brennel, via a compressor and into Pro Tools. It sounds like a really nicely miked-up AC30. It's a bit bottom-heavy, but if you compress it and use the right guitar, it sounds great. It's the best DI in the world, though quite big to carry around!

"Tom writes loads of songs, as does Damon, and it's great working with a really prolific songwriter like that because nothing's that precious — you don't have to make any particular song work. You can say 'Let's try it like that,' and if it doesn't work, try a different song. Obviously it's not always that easy. Sometimes you realise that you've got a really good song on your hands, and you've got to try a little bit harder with it. But generally I like following a song through to its logical conclusion and seeing what comes out, rather than being really Draconian about it. Because he's a solo artist, you don't have to try and get a whole band involved on every tune and we could experiment with lots of different ways of doing things."

only place to do that was outside, which is about as dry as you can get. Those were the options — either completely dry or completely roomy. There were some smaller rooms there as well, including a derelict bathroom that was great for vocals, but lots of the vocals and drum parts were done outside, sometimes with a mic inside the room so I could mix in some ambience."


A further selection of unusual gear, including a Lorenzo electric harmonium, Hillier's trusty dictaphone, and assorted guitar pedals.

Mad Dogs And Englishmen

As well as getting to grips with the quirks of their *ad hoc* studio, the band and producer had to battle the heat of Morocco in late August. "We'd had so much time off at the beginning waiting to get set up that we couldn't take any more days off, so we just worked non-stop for a month. You couldn't work that much during the day because it was so hot. Damon would be working hard writing lyrics, or the band would be working on songs, and I'd be editing and doing mixes, but not with too many people in the control room. It was just about OK with me, Jason and James in there, but it would overheat if there were any more people in there during the day."

It was not just the local climate that was causing problems: the local cuisine had a few surprises of its own. "We all got food poisoning out there as well. We all lost about a stone and no one could be more than 20 minutes away from the nearest toilet. It was 200 yards to the house where the toilets were and we had a bike outside and every now and then you'd see someone legging it back to the house."

The arrival of Norman Cook was a welcome shot in the arm. "Norman turned up at just the right point. We'd done a lot of work but it was great to have someone come in with a bit more energy and a bit more focus. He came out for five days and we worked on three songs, which was really good fun. We finished off the tune we'd started in London, 'Money Makes Me Crazy' which didn't make it onto the album in the end because it was just too happy, it was too bright and bouncy, but it's a great tune — it's the 'B' side on 'Out Of Time'. We did two more tracks which worked really well and really tied in with the rest of the album. Working with different producers on various tracks involved them taking the track away and doing their thing with it, which was what we wanted them to do, but then we'd have to wrestle it back into our realm. That's why I mixed everything — it just wouldn't have sat together as an album otherwise. William Orbit was in the middle of doing his record and didn't have the time to come and work with us, so he'd send us hundreds of tracks for each tune and we'd sift through, pick the bits we liked and blend them into what we were doing. 'Crazy Beat' [*which was co-produced by Norman Cook*] started out as a completely different song. We'd changed it round various ways and tried different things with it and then Norman came in and said 'I like that bit, I like this bit,' and we'd do that."

A closer collaboration was involved in 'Gene By Gene', the other track on the album co-produced by Cook. "That was where the collaboration worked best, because we were all there together and wrote it together. The track is based around some samples of Damon jumping up and down on an old truck that was parked round the back of the farmhouse. It had a really good squeak on it! We also used samples that me and Dave had made at 13 on a particularly slow day. I told Dave that if you hit a cymbal and lower it into a bucket of water it pitch-shifts. So we got an old fish tank and filled it up with water and spent all day hitting cymbals and lowering them into the water, sampling them and making loops. Gongs too — anything we could hit and lower into water. Norman made a loop out of Damon's squeaking sounds which fitted with the cymbal sounds and it turned into a tune."

Hillier is in no doubt that the recording sessions in Marrakesh, which might have been viewed as rock & roll extravagance or affectation by some, were worthwhile in terms of the music they produced, despite the difficulties he and the band encountered. "It started as 'Yeah, let's go to Marrakesh, that'll be great!' and of course it

Mixing To Tape

"I always mix to half-inch tape, especially now that I record everything into Pro Tools. It's nice to have a bit of tape compression in there at some point. I record everything into Pro Tools and then when I mix it I send it all out of separate outputs and through an analogue desk and do outboard compression and EQ, basically treating Pro Tools like a tape machine. Then I take the mix from the desk and record it on to half-inch tape and Pro Tools simultaneously, so I've always got the option. I can either use the half-inch mix or the Pro Tools mix. I find you can get some quite high peaks sticking out in some places when you come to mix down. When you record on analogue tape it smooths out quite a lot of those really harsh attacks and flattens it all out, but in a good way rather than having to over-compress everything. You can take a bit of the dynamic out of a tune just trying to deal with those peaks with compression. With tape, you don't even hear the compression but you'll find that those peaks have disappeared. It tightens up the bottom end as well — that's one thing you miss with not using two-inch tape any more. It tends to raise the bass a bit, making the octave above the fundamental more audible. That makes it easier to put it in the mix and means you can hear the bass on small speakers. If you're not using tape you have to use loads of other tricks to do it, like putting the bass through the VCS3 [*see box*] just to add that extra octave in. It's all swings and roundabouts, really. When we were working on tape, we used to have to use tricks to get the lower octave to come out, to give the mix more size, but now I find it's the other way around, you're having to do things to bring out the more audible octave of the bass."

was a bloody nightmare! But it was definitely worth it in the end. As it was getting more and more expensive and more and more complicated I was thinking 'Shit, maybe this wasn't such a good idea.' But when we got going it was absolutey ideal, it filled in all the holes in the record that we needed to fill."

The only problem was, the record wasn't quite finished.

Out Of Time

"We ran out of time in Marrakesh, as we knew we would, and we decided we wanted to finish all the tunes we'd started. We didn't want to kick songs out until they were done and we knew that if we didn't finish them they'd never be finished because we'd all be off again doing other things. We wanted to hold on to what we had and make the most of it with everybody there — me, the band, and the studio as well. The studio we'd hired had become part of it. It was everything we wanted but also wasn't in a studio, if you get my meaning. So we decamped to Damon's farm in deepest, darkest Devon — from bright sunny Marrakesh to rainy Devon in October. But from a technical point of view it was great. Marrakesh was difficult for me because the control room had that ridiculous reverb. We set the studio up in Devon in this 200-year-old barn that was acoustically completely dead, which was ideal. It was raining all the time, which is what happens in Devon at that time of year. So we were just bunkered down, really concentrating on the music — we had been in Marrakesh as well, but it was really hot and the record was really open-sounding because we'd recorded so much outside. We did a lot of honing down of the songwriting and lyrics in Marrakesh, so Devon was about honing down the sound of the record and mixing it. We re-recorded some of the vocals and instrumental parts. But we were out of the way and undisturbed, so we could finish it off."

At the same time, however, the band were still adding to the record. "It was very much like a mixing session but with extra recording. I don't really like to break up the making of a record into strictly distinct phases: tracking, mixing and so on. I like to be free to add stuff all the way along. But I decided eventually that the only way we were ever going to finish it, because we were all having such a good time in the studio and enjoying it so much that we were just writing new songs, was if I set a date. So I said 'At the end of November I go to Liverpool and start recording Elbow's album.' If I hadn't, we'd still be doing it now I'm sure, and we'd have 40 songs. We tried mastering the record in London but we hadn't quite decided which tracks were

VCS, MS & ARP

Hillier is the proud owner of three classic analogue synths, an EMS VCS3, a Korg MS20 and an ARP 2600, and the uses he finds for them are far from conventional. "I often record things through the VCS3 or the ARP to treat them. There are certain plug-ins I use a bit, but I tend to use the analogue stuff more. As well as its spring reverb, which I use a lot, the VCS3 is very good for tightening up a sound. Say you've got something that sounds really nice when you solo it but then, when you put it in with the rest of the track, you lose the nuances of it. If you compress it, it sounds too squashed, but if you put it through something like the VCS3, without using the oscillators, just using the filter and the output stage, it'll just make it sit in the track so you can hear it better, without it sounding over EQ'd or over-compressed."

Hillier also uses the filters to mould and shape sounds. "I use both synths a lot, either very subtly to just control a sound and push it into the right space, or for completely over-the-top treatments of stuff." Blur's limited edition single 'Don't Bomb When You're the Bomb', released anonymously only on seven-inch vinyl, features just such a treatment. "On that track we got this ridiculous bass sound by running the bass through the ARP with the ring mod on and everything. It sounds like the best bass pedal in the world ever, but it's a little bit impractical for live use!"

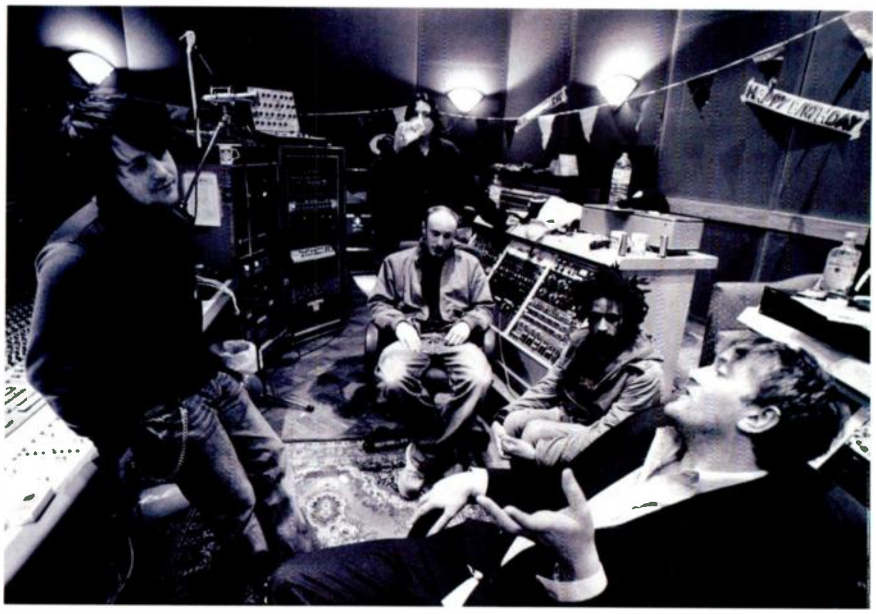
Ben Hillier's small programming room at The Dairy, with a Soundcraft Ghost desk and KRK main monitors. The funny-looking yellow box is a Geiger counter, also sourced from Ebay...

going on the record and some of the tunes didn't sound quite right. In the end I went to New York on my own and mastered it over a weekend with Howie Weinberg at Masterdisc, who's very experienced. That was the only time we went to a proper studio during the whole process!"

Elbow Room

Ben Hillier had already produced Elbow's debut album, *Asleep In The Back*, which was nominated for the Mercury Music Prize in 2001. Its success reflected the band's considerable powers of perseverance: signed to Island in 1998, they had recorded what was to be their first album before being abruptly dropped when the label shifted its priorities from alternative to pop. A lifeline deal with EMI fell through soon after. Dropped twice in the space of a few months but undeterred, the band eventually secured a deal with V2, re-recording the album that was to become *Asleep In The Back* with Hillier; and when I met Hillier he was in the final stages of mixing the follow-up to *Asleep In The Back*, *Cast Of Thousands*. "It's a terrible cliché, but 'second album syndrome' is a very real thing. You don't know how hard it is do do a second album until you have to do it. Tom McRae's album was his second and he's lucky in that he writes a lot of songs, but even then he had to deal with taking his music to another level, to the next stage.

"Elbow had written a load of songs that they thought they were happy with, but neither I nor the head of the label, David Steel, who's a big fan of theirs and does their A&R, was convinced that they had enough material. They were having to wait for me because I was doing the Blur record, so they wrote some more songs, and all of a sudden things were looking a lot rosier. So we went into the studio, and then it turned



Elbow recording *Cast Of Thousands* in Parr Street Studios, Liverpool.

that they weren't all happy with some of the tunes. They hadn't lived with them for long enough to know if they liked them, so quite a few tunes fell by the wayside. So all of a sudden we'd gone from having 12 songs to having eight. We took a two-week break and they wrote some new songs and reworked some of the old songs. We went back and sifted through the demos and picked out what was right about some songs and what we were doing wrong with others. So in the end we had to work very hard to get things right. If you're working with a band who've recorded a lot of records, they're used to having to push at it. But if you're working with a band who've only recorded one record before, no matter how prepared they think they are, they're not quite prepared for how difficult it can be. But they've worked really hard and as a result they've got some really strong stuff.

"When we went into the studio with the Blur record, we didn't have any songs at all, but no one was worried because Damon and the band have already proved that they can produce the goods in the studio. Elbow have got to prove that, and they're in the process of proving it right now. They've got a really good attitude — they will work hard on songs, they will try lots of different things, and if you're willing to do that, you can come up with a really great record."

The new album was recorded at Parr Street Studios in Liverpool — Hillier is a big fan of the studio's stone and wood live rooms. "They're a bit less moody on this record, with a broader sound palette. There's some quite full-on tunes on there as well — maybe even a couple of singles! They really want to move forward. You've got to work with a band with ambition. One of the things I hate in music is people hiding behind a front, pretending they're cool and actually not having any ambition. I hate it when people refuse to take the next step because they think it's selling out, when in actual fact if they do it and do it well it's not selling out, it's just having ambition. And there's nothing wrong with that. I don't see why music should be bad because it sells well. I think completely leftfield music should be selling really well all the time. Take The Specials, for example. If you listen to their music now, it's mental. The arrangements are all over the place, really clever, really well played and it was hugely successful at the time. Those were big pop hits but if you listen to it now it's still really clever stuff. It's got absolute dignity and absolute integrity. I think that all music should have that — dignity and integrity. And enormous sales!" Amen to that. SOS

Pedal Power

In addition to his love of Russian microphones, Hillier has a large collection of vintage, and vintage-style, pedals and stompboxes. Lovetones (www.lovetone.com) are a particular favourite. "I've got all the Lovetone pedals and they're all really good. I saw an ad in *Sound On Sound* for the Meatball when they first made them. It looked really good, so I phoned up Vlad [*Naslas, the inventor*] and he explained that they hadn't built them yet but were asking people to buy them in advance. I paid for it up front and got one of the first run six months later. Now he tells me about new pedals before they're released."

Hillier also has a selection of Mu-tron (www.mu-tron.com) and Electro-harmonix (www.ehx.com) pedals which he uses for just about everything but guitar. "I bought a lot of these in the States when I was working in LA a while ago. It was before people had quite realised that vintage pedals were as good as vintage guitars and vintage amps. I went into the Guitar Center, a huge music supermarket in LA, and I asked if they had any pedals. The shop assistant pointed to a load of bright orange and bright pink 'metal hammers'. I asked if they had any old stuff and he pointed to this tea chest that was just full of old Electro-harmonix things, all $50 each. So I came home with an armful."

The eccentric, hand-built and hand-painted Z-Vex (www.zvex.com) pedals are a new favourite. The Fuzz Probe, which Hillier describes as a super feedbacker combined with a Theremin, allows the user the manipulate the effect by moving his or her foot over a brass plate. Another pedal, the Ooh Wah, steps through eight fixed wahs, either in sequence or randomly.

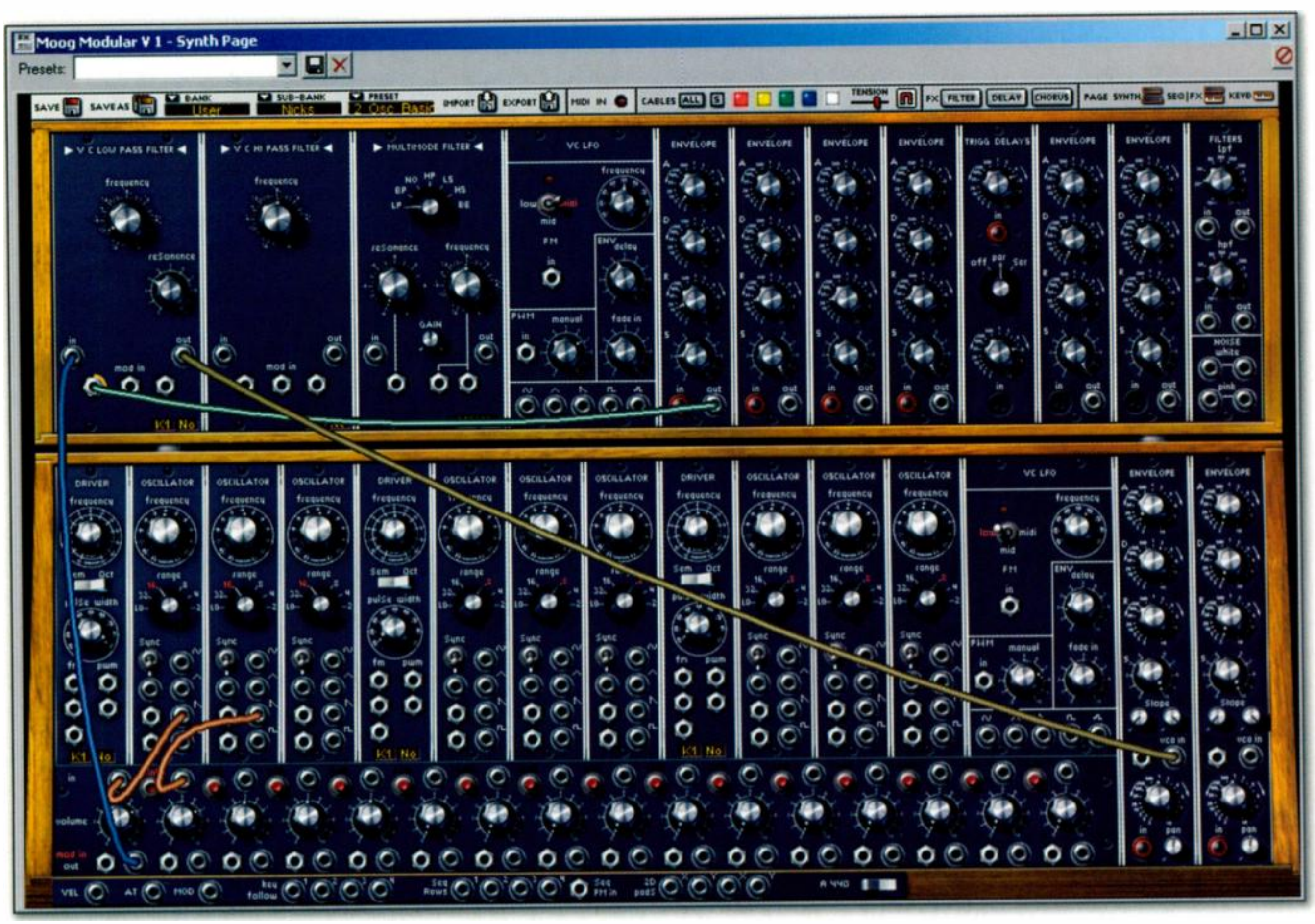
Arturia Moog Modular V

Nick Magnus

For their latest software instrument, Arturia have taken on the tough job of modelling a modular synth — *and* it's the granddaddy of them all, a Moog modular... Switched-on software or spaghetti junction? We find out.

Virtual Modular Synth for Mac & PC

The number of software recreations of 'classic' instruments is growing almost daily, no doubt due not only to the increasing scarcity of the original items, but also the ever-present enthusiasm for all things retro. Recent examples of these include Steinberg's *Model E*, a simulation of the Minimoog, Native Instruments *Pro 53*, a Prophet 5 clone, and Gmedia's *Oddity*, a recreation of ARP's famous Odyssey synth. Arturia, the French company responsible for the *Storm* studio software package, have now entered the software-synth arena with *Moog Modular V*, a virtual-analogue software instrument based on the Moog modular synthesizer systems of the '60/'70s. The first Moog modular was built in 1964 for, and in collaboration with, Herbert Deutsch. It now resides in the Henry Ford Museum & Greenfield Village, Dearborn, Michigan. This led to the 900 series in 1965, the Systems 1, 2 and 3 in 1967, and the 3C in 1969. Modulars continued to be manufactured by Moog in Trumansburg until 1971, when the company was sold to Bill Waytena, renamed Moog Music, and moved to an old gelatin factory in Buffalo. Production continued in Buffalo, culminating with the Moog 15, 35 and 55 in 1974 when the company was again sold,



The beginnings of a simple patch on *Moog Modular V*'s Synth Page.

The Modules, Controls & I/O

921A OSCILLATOR-DRIVER MODULE (x3)
Frequency, Octave/semitone adjustment selector, Pulse width, FM inputs (x3), PWM inputs (x2), Key-follow selector, sequencer-control selector.

921B OSCILLATOR MODULE (x9)
Frequency, Range (Lo, 32', 16', 8', 4', 2') Sync on/off, Sync source, Waveform outputs (sine, triangle, saw, pulse).

MIXER CHANNEL (x16)
Volume, Input, Output, Modulation input, Link (to neighbouring mixer channel).

VCA ENVELOPE MODULE (x2)
Attack time, Decay time, Release time, Sustain level, Slope time, Slope level, VCA input, Mod input, VCA gain, Trigger input, Pan.

LFO MODULE (x2)
Frequency, Range (Low, Mid, MIDI) Freq Mod input, Delay time, Fade-in time, Pulse width, PWM input, Waveform outputs (sine, triangle, saw, pulse, S/H).

AUXILIARY ENVELOPE GENERATOR (x6)
Attack time, Decay time, Release time, Sustain level, Output, Trigger input.

TRIGGER-DELAY MODULE (x1)
Delay time 1, Trigger input 1, off/parallel/series selector, Delay time 2, Trigger input 2.

NOISE GENERATOR
White noise, two outputs. Pink noise, two outputs.

NOISE-FILTER MODULE
Low-pass cutoff, input, output; High-pass cutoff, input, output. Can also be used as static filter for oscillators.

904A LOW-PASS FILTER
Cutoff frequency, Resonance, Input, Output, Modulation inputs (x3), Key-follow selector, sequencer-control selector.

904B HIGH-PASS FILTER
Cutoff frequency, Input, Output, Modulation inputs (x3), Key-follow selector, sequencer-control selector.

904C FILTER COUPLER
Band-reject/Band-pass selector, Bandwidth, Cutoff frequency, Bandwidth Modulation input, Cutoff modulation input (x2), Input, Output, Key-follow selector, sequencer-control selector.

12dB MULTI-MODE FILTER
Mode selector (Low-pass, Band-pass, High-pass, Notch, Low-shelf, High-shelf, Bell) Cutoff frequency, Resonance, Gain, Resonance modulation input, Cutoff modulation input (x2) Input, Output, Key-follow selector, sequencer-control selector.

this time to Norlin in 1975. Modular systems could still be ordered specially until 1981.

Keith Emerson is amongst the best-known modular system users with his huge System 3C, along with the likes of Walter Carlos whose pioneering *Switched On Bach* album made Moog a household name in 1968. Japanese synth luminary Tomita, Jan Hammer, Larry Fast, Herbie Hancock, The Beatles and The Rolling Stones all used Moog modular systems, and even today artistes such as Tangerine Dream, Jean-Michel Jarre and Tomita are still enthusiastic modular system owners.

Does Modular Mean Better?

In non-modular synths like the Minimoog and ARP Odyssey, the audio signal paths are internally 'hard-wired'. On instruments like these, this means that a signal's journey is preset to travel first through a mixer, then through filter and amplifier modules. Modifiers such as envelopes and LFOs interact with these modules, allowing us to change their behaviour over time to create amplitude curves, filter sweeps and vibrato, for example. This (grossly simplified) predetermined architecture, whilst still fairly flexible in its own right, does ultimately restrict the synth's capabilities. By contrast, the greatest advantage of a truly modular approach to synthesis is flexibility. Since there is no predetermined wiring between modules, it is entirely up to the user to decide how the modules will be connected. Traditionally this is done on a hardware modular synth using patch cables — the more modules you connect together, the more complex the sound can be. This is especially important when considering ways of modulating the sound. Since the oscillators' waveforms are static in their raw state, we need modulation sources to add movement and interest. The beauty of a modular system is that almost anything can be used to modulate anything else, and since the signals in analogue modulars are all simple voltages, whether they are audio or control signals, you can use them completely interchangeably. For example, not only can you use an LFO to modulate an oscillator to produce vibrato, but you could also use the output of another oscillator to modulate the pulse width of that LFO's waveform, whilst simultaneously modulating that oscillator with the output of a noise generator, whose amplitude is in turn being modulated by a second LFO... As you might imagine, the potential for complex sound creation is huge. Nevertheless, despite the apparently limitless possibilities, a logical signal flow must still be adhered to in order for a sound to be produced at all!

Compatibility & Installation

Moog Modular V (or *MMV* for short) comes on a cross-platform CD with versions for both PC and Mac. The minimum OS specification for for PC users is Windows 95, 98, ME, 2000 or XP, and Classic, 9.x, and OSX.2 or higher for Mac users. In both cases, 128MB of RAM is required, with a recommended minimum of 500MHz processor speed. The software can also be run in a wide variety of plug-in formats: PC users have the choice of stand-alone (ASIO or Direct Sound,) VST, DXi, and RTAS versions. Mac OS X users can choose from stand-alone, VST, RTAS, HTDM and Audio Units for Logic Audio, while Mac Classic users have VST, MAS, RTAS and HTDM — but no stand-alone. Installation onto my PC was straightforward and painless — *MMV* was automatically available as a stand-alone program, and I additionally chose the DXi version, as my host sequencer is *Sonar* v2.1.

Arturia *Moog Modular V* **£200**

pros
- Remarkably authentic tone.
- Attractive user interface.
- Massive creative potential.

cons
- There are some minor authenticity issues.
- A few bugs to be addressed.
- CPU-hungry in polyphonic mode.

summary
A software recreation of the famous Moog Modular systems, *MMV* has an authentic tone, a nice user interface and really brings out your creativity. Certain authenticity issues and one or two bugs need to be addressed, but otherwise this a very well-behaved piece of software that is also enormous fun.

Onscreen Layouts

When first firing up *MMV*, you are presented with the main synth programming page. There are three screen pages — the Synth page (from where all the main sound creation is done), the Sequencer/Effects page (which, as the name suggests, contains the sequencer, effects, a virtual keyboard and associated performance controls), and the keyboard and performance controls alone. All three screens have a pleasingly wood-trimmed, photo-real retro appearance, which certainly helps to conjure up a suitably Moogy mood.

User-Interface Conventions

MMV provides a number of different tools for creating patches. On the synth-editing page, the most immediate of these are the virtual patch cords used to connect modules together, without which we would hear nothing at all. These are used, unsurprisingly, in conjunction with the graphical jack sockets, which come in two varieties — smooth round sockets (signal inputs/outputs) or hexagonal ones (modulation inputs). To make a connection between two modules, just mouse-click on a socket — a cable appears, and you drag the other end of it to the target socket on another module. When you let go of the mouse button, the connection is made. In order to avoid 'illegal' connections (eg. connecting an input to an input) all the possible 'legitimate' destination sockets are helpfully illuminated with a yellow border. The hexagonal modulation inputs have an additional feature — once a connection is made to one of these, pointing the mouse to the edge of the socket 'nut' and clicking allows you adjust the input sensitivity by scrolling the mouse up and down. A yellow collar appears round the circumference of the socket to indicate whether a positive or negative value exists (as shown in the screenshot on page 58) and a momentary dialogue box appears while the mouse button is held down, allowing you to set the value accurately. As with most mouse/value settings on *MMV*, holding down the left mouse button and dragging the mouse over something gives you coarse settings, while holding down the right mouse button (or the Shift key on a Mac) and dragging allows for finer adjustment.

We all know what messy things cables can be, even if they are virtual ones — and it doesn't take too many connections before crucial sockets and panel legends are obscured, and the synth panel begins to resemble a plate of *fettucini ai funghi* that went horribly wrong. *MMV*'s solution to this is highly elegant — as you move your mouse pointer through the jungle of cables, they are swept aside as if by some unseen hand, revealing whatever lies beneath. The display of patch cables can be further customised on *MMV*'s toolbar. Cables can be visually 'muted' by colour group, the overall tension or 'sag' can be set on a slider, and the mouse-sweeping function can be turned on or off via the magnet icon.

As an alternative to the drag-and-drop patch cable, it is also possible to make connections by right-clicking on a socket and choosing the destination from a drop-down menu (again, Mac users should hold down the Shift key and click). While the majority of connections can be made either using the drag-and-drop or menu method, the trigger input connections can only be made using menus. These input types are shown as black 'S-Trig' type sockets, with two vertical rectangular slots, and can be found on the VCA Envelopes, Auxiliary Envelopes, and Trigger Delay. Their purpose is to provide a trigger source for these time-based modules — for example, a typical trigger source for an envelope module might be the keyboard, so that when a key is pressed the envelope begins its cycle on cue. Of course, the keyboard is not the only trigger source available; they may be derived from a wide range of sources, such as an individual sequencer step or an LFO. The permutations are vast; without devoting a dissertation to the possibilities, the best advice I can give is that you should experiment!

The one remaining connection on the synth page that can only be made via a menu is the oscillator sync source, from which you can choose any of the nine oscillators — you can even sync an oscillator with itself.

On a final note concerning connectivity, a general rule of thumb is that while each input can have only one source, each output has the luxury of multiple destinations. Of course, this is not so easily achievable on a hardware synth, as only one jack plug can be inserted into a socket at a time! Software, fortunately, does not have to adhere to these real-world limitations, so an LFO's sine wave output, for example, can be routed to as many destinations for which you can find uses.

Test Spec

- Arturia *Moog Modular V* version reviewed: v1.1.
- 2.4GHz Pentium 4 PC with 1GB of RAM running Windows XP.
- Cakewalk *Sonar* v2.1.

"I've had immense fun playing with *MMV*, whether creating bizarre noises, thundering Taurus-pedal impressions, Minimoog-style leads or huge, mythical nine-oscillator Memorymoog brass patches."

The Synth Page

This is the main sound-designing page (see the main screenshot at the start of this article) and is divided into two main 'cabinets' — the upper cabinet contains mainly sound-shaping modules and filters, while the bottom cabinet houses the oscillators, mixers, VCAs and various control jack sockets (for a complete list of modules, their controls and connections, see 'The Modules' box at the start of this article). Let's take a look at how the oscillators are organised, and how they relate to the mixer channels. The oscillators (equivalent to the original Moog 921b hardware module) take up a large chunk of the bottom cabinet — there are nine of them in total, grouped into three groups of three. Each group of three 'slave' oscillators has its own Driver controller (equivalent to the original 921a) which acts globally on all three oscillators in that group. With this you can set the overall tuning (±6 octaves) and waveform pulse-width of the group, as well as assign global sources for frequency modulation and pulse-width modulation. Clicking on the Driver's left-hand digital display selects pitch control from one of four programmable key-follow curves (see the section on the Keyboard Page later) or selects a fixed tuning (that is, with no key follow). The right-hand digital display is used to select which, if any, of *MMV*'s sequencer rows will control that oscillator group (more on the sequencer in a moment).

Each slave oscillator in a group has a pitch range from 32' to 2', and is fine-tuneable over ±1 octave. The additional 'Lo' setting enables any oscillator to be pressed into use as an LFO, in case the two dedicated LFO modules should prove to be insufficient. Four waveform outputs are offered, all of which can be used simultaneously. Sync on/off is individually switchable per oscillator, the sync source being selected by clicking on the jack below the sync switch.

The 16 Mixer channels, situated below the oscillators, provide the means to arrange the oscillator outputs into logical and manageable 'groups', which makes life very much easier for when you need to apply common treatments to two or more oscillators. The Link buttons that separate each pair of Mixer channels are the key to this oscillator grouping. For example, imagine you're trying to make a brass sound using the detuned sawtooth waveforms from three oscillators, to which you want to apply an overall filter, filter envelope and VCA envelope. Sure, we could make a lot of work for ourselves by sending the output of each oscillator through each of the three filters, assigning identical envelopes to each filter — but straight away, we hit a problem. There are only two VCA envelopes, and we want to use three oscillators. The way to get around this is to send oscillators 1, 2 and 3 through mixer channels 1, 2 and 3 respectively, then click the Link buttons between Mixers 1 & 2, and 2 & 3. The summed output of all three oscillators now appears at the output of Mixer channel 1, which can now be sent through one filter, one filter envelope, and one VCA envelope. Other Mixer groups can be created in the same way, but the thing to remember is that there must be at least one broken Link between Mixer channel groups to do this, otherwise all the Linked channels will appear at the output of the Mixer farthest to the left. Trust me, this is not as complicated as it looks in print! Even if you use nine oscillators and nine Mixer channels, you still have seven Mixer channels left over for other duties.

MIDI Remote Control

Once a basic patch has been created, there are further connections to consider to make the sound respond to your MIDI keyboard in the desired way. Using the modulation wheel to apply vibrato is a typical example, so you'd need to patch in the mod wheel appropriately. The row of jack plugs beneath the Mixer section provides the means to route keyboard controllers to the main synth, so in the case of the brass sound you'd firstly patch the output of one of the LFOs to the input of a spare Mixer channel. The output of that Mixer goes to the FM input of Driver 1, and the mixer output should be turned up to the point where the pitch modulation begins to take effect, then backed off until it stops. The mod-wheel output should then be patched to the modulation input of the Mixer channel, and the Mixers' modulation input sensitivity should be adjusted until the desired depth of modulation is reached when the mod wheel is pushed all the way up.

More detailed knobular control is of course possible via a MIDI control surface or hardware synth whose knobs send MIDI controller data. Most of the modules' rotary controls can be assigned a MIDI controller number, and this is easily done using the MIDI 'learn' function. Holding down the right mouse button (or the Shift key on a Mac) and clicking a knob brings up a dialogue box — you simply click on the 'learn' field, twiddle the appropriate knob on your controller, and it's done. Alternatively, click on the 'controller' field and select the required controller from a drop-down list. To de-assign the knob, simply uncheck the 'active' box and the connection is removed. Curiously, though, you cannot assign the

MMV's Keyboard page. Most of the routing and modulation options will be known to those familiar with real analogue modular synths, though only the biggest systems had all of these in the olden days!

mod wheel directly to a parameter in this way, even though controller #1 appears in the drop-down list of options. Therefore, mod-wheel control is restricted to those parameters that have direct modulation inputs, using the Mod output in the keyboard control jackfield. For example, the mod wheel cannot be used to control the resonance of the 24dB-per-octave low-pass filter, as resonance has no direct modulation-input jack. If you attempt to make this connection using the drop-down menu, it still will not work, even though filter resonance is shown as a destination. The resonance on the 12dB-per-octave Multi-mode filter, on the other hand, does have a modulation-input jack, so can be controlled from the mod wheel.

Continuing the subject of MIDI controllers, an annoying omission on *MMV* is the lack of a master output level that can be controlled by MIDI. You could of course assign the VCA gain knobs to MIDI CC#7, but they are then forced to operate over their entire range of -100dB to +12dB, the upper range of which is almost certainly going to overload your soundcard unless the oscillator mixer levels are set very low. Also, if your patch requires both VCA modules to be set to different levels, the difference in levels would be lost the moment you tweaked your MIDI volume controller. This lack of a master volume control is common to a number of software synths, so this general plea goes out to all virtual-synth designers — give us a MIDI-controllable master volume, please!

The 12dB-per-octave multi-mode filter. Note the yellow 'value' collars around the modulation inputs. The digital display in the corner shows the filter is routed to key follow 2, and sequencer row 3.

The Keyboard Page

This is the most reduced view of *MMV*, showing the virtual keyboard and various performance-related controls. The pitch-bend wheel modulates not only the pitch (with adjustable range) but can modulate the filter cutoff frequency as well. I was surprised to find that the bend wheel-to-filter cutoff assignment only works in a positive direction (ie. the filter cutoff rises with pitch), but cannot be made to work inversely. Monophonic and polyphonic modes are offered, with 16, 32 or 64 voices of polyphony, depending on your needs and available CPU power. An alternative 'soft dist' monophonic mode is also present which, according to the manual, adds 'soft distortion in the VCAs'. However, I confess finding this to be... er... extremely subtle! Release polyphony can be set separately to make optimum use of CPU power — although during the course of the review I was unable to achieve more than 16 voices, despite having adequate CPU headroom. In addition, when playing a sound with a significant release time (above around 300ms) attempting to play more than 16 voices caused *MMV* to mute its entire output until the release times had recovered, at which point it would continue as normal. Arturia are apparently working on a fix for this problem which will be implemented in a future upgrade, although nothing was available before this review went to press.

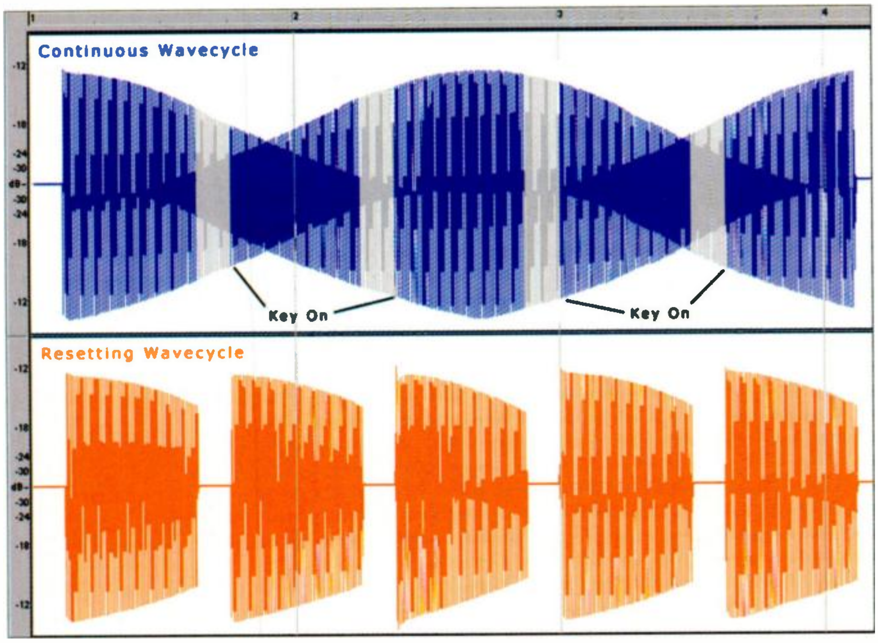
To the left of the keyboard are the portamento (or glide) On/Off switch and Glide Time knob. Glide time is not sensitive to relative note position — the time is absolute, regardless of whether two consecutive notes are a semitone apart or four octaves apart. Portamento is only available in monophonic mode, and the Legato button selects whether the portamento is full-time, or triggered only when notes are played in a legato fashion. The Retrigger button selects whether legato playing of monophonic sounds will generate single or multiple triggered envelopes.

The four key-follow generators have controls to adjust the slope, centre key, range and threshold (the lowest note at which the key-follow slope takes effect). These govern essentials such as the way the oscillator pitch and filter cutoff track the keyboard, and in fact these two functions are easily assigned using the digital displays on the Driver and Filter modules. The key follows can also be patched in via the jackfield to other more esoteric destinations that require a more finely adjustable, non-linear response to the keyboard. Note that each key follow has its own pitch-bend and portamento on/off switch, so performance tricks such as 'double-stopped' pitch-bend of one oscillator against another, or portamento 'trailing' effects between oscillators can be realised.

Some controls duplicate others on the synth page, namely ADSR sliders for the two VCA modules, and cutoff frequency knobs for each of the three filter modules, enabling on-the-fly 'quick fixes' to be made to a patch from this reduced view page. Lastly, a pair of X-Y pad controllers are provided, the axes of which can be assigned to the modules of your choice via the jackfield on the synth page. These X-Y pads can themselves have MIDI controllers assigned to them, although in practice I found this didn't work too well. The pads will only respond to one incoming controller at a time — but not both. It is just as effective (and works better) to assign MIDI control directly to the parameters you want.

The Sequencer/FX Page

This page incorporates the Sequencer, the keyboard layout described above, as well as a Fixed Filter Bank module, a Delay module and a Chorus module. The Sequencer comprises three rows, each row having eight steps. A digital display under the word 'chain' determines how the three rows will be deployed. In its basic mode (labelled '—') there are eight steps; each step of the three rows can be set to a different value, so row one might govern pitch, row two might control filter cutoff frequency, and row three might control the VCA level. Further modes allow you to chain rows together to give 16- or 24-step sequences — for example, L123 plays rows 1, 2 and 3 sequentially, L321 plays the rows in inverse order, L23 plays rows 2 and 3 only, and so on. The 'C' modes work in a vertical fashion, so C123 plays step 1 of rows 1, 2 and 3 followed by step 2 of rows 1, 2 and 3... and so on. At any time you can 'force' a sequence to play a specific row by clicking (or assigning a trigger to) the trigger switch at the right-hand end of each row. The Smooth knob acts in a similar way to portamento, by smoothing the transition from one step to the next. The Link buttons between each step allow you to skip the envelope trigger for the following step. If the patch being sequenced has a very short release time, you tend not to hear linked steps unless the pitch intervals between the linked steps are large, as the linked note is not articulated by the envelope. If there *is* sufficient release time, the effect is similar to a single-triggered (as opposed to multiple-triggered) envelope. Any step in the sequence can be repeated



The upper window shows how the wavecycle of hardware VCOs is 'dipped into' each time a note is played. The lower window shows how *MMV*'s oscillator wavecycle phase is reset each time a note is played.

up to eight times — the white button next to each counter selects whether an envelope trigger is generated for each repeat or not. Step order can be changed via the rotary knob beneath each step for some off-the-wall variations — if you set the value of any of these to random, the sequence restarts at a random step when that point is reached; similarly, if you set them all to random, the entire sequence will be randomised. Again, the red buttons below these knobs (or their associated trigger inputs) can be used to force-reset the sequence to any desired step.

Sequencer tempo can either be set manually, controlled dynamically with the output of a modulation source, or sync'ed to the tempo of *MMV*'s host MIDI sequencer. This of course means *MMV*'s sequencer can be used to provide tempo-sync'ed rhythmic effects to any patch simply by routing any or all of the sequencer rows to, say, filter-cutoff frequency or oscillator pulse width. The sequencer offers an immense amount of possibilities, and is guaranteed to keep you entertained for hours.

Effects Modules

The Fixed Filter Bank optionally applies 12 fixed frequency bands of EQ to the final patch, together with variable Q for each band, plus separate high- and low-pass filters. An overall gain control is available to tame or boost any extreme settings you might make, and a reset button thoughtfully puts all the knobs back to zero so you can start from scratch if necessary. VCAs 1 and 2 are independently selectable for treatment by the Filter Bank, from where they are passed to the two effects.

The Dual Delay can be sync'ed to MIDI if required, and produces a stereo delay with separate left/right controls for time, feedback and cross feedback. The wet and dry signals can be individually adjusted with up to +12dB of gain available.

The Chorus module has a very characteristic sound — quite unlike the smooth, glistening chorus effects of recent times. In fact, its distinctly retro quality is highly reminiscent of the type of chorus found in string machines of the 1970s, and it seems very appropriate to *MMV*. There are three types of chorus on offer — simple, medium and complex. These choices appear to generate 1, 2 or 3 chorus voices, and careful adjustment of the remaining parameters is required to get the best from them. These parameters include Rate, Delay, Amount, Stereo Width and Rate controls. The Stereo Width and Rate controls generate detuning effects of their own, combining with the general chorus settings to create extremely thick ensemble textures. It's all gloriously nostalgic, and I love it!

But How Does It Sound?

If you've got this far without jumping to this section, well done! For those of you who simply can't wait to know, the answer is... it's uncannily like a Moog. The filters sound very authentic, although unlike the real thing the 24dB-per-octave low-pass resonant filter is unable to self-oscillate in the absence of an oscillator input. High resonance settings don't quite have the same untameable energy as a Moog filter going at full tilt, but that is probably just as well for the sake of your studio monitors — *MMV*'s output can be extremely 'hot'! The envelopes are also as snappy as you could want. In short, it simply has *that* sound. Having owned three Minimoogs, a Memorymoog, a Moog Source and had access to three sets of Taurus pedals, I'm pretty convinced by the tonal authenticity of *MMV*. To help you get a feel for what *MMV* can do, it comes with several banks of presets programmed by a variety of celebrity users. These patches also serve as handy 'how-to' reference material for those moments when you simply need a bit of help. The fun starts when you begin to patch together your own sounds — and it is loads of fun.

However (there's always one of those, isn't there?) there are a couple of authenticity issues to mention, at the risk of acquiring a reputation as a picky old fusspot.

The first concerns the oscillators. On a hardware VCO synth, the oscillators are constantly running — their output is heard once the VCA opens. This means that every time you play a note, you catch the oscillator's wave cycle at whatever point it happens to be at that moment. *MMV*, on the other hand, resets the oscillator wavecycle to the same point each time you play a note. So why is this an issue? If you are listening to only one oscillator, the effect of this would be quite hard to detect. But if you are listening to two or more oscillators, slightly detuned, the phase relationship between them always snaps back to the same position with each note — in other words, the detune cycle is discontinuous. The general effect is that of repeatedly playing back a sample of a pair of detuned oscillators, and is quite noticeable, especially with monophonic sounds. The diagram above illustrates this point more clearly.

The second point concerns the envelope generators, and again this particularly affects monophonic sounds. If you are playing a patch with a significant envelope release time, *MMV* resets the envelope to zero if you retrigger the note before the release cycle has completed. If the attack time is anything other than fast, this produces a pronounced 'sucking' effect. By contrast, a real Moog envelope begins a new envelope cycle picking up from wherever the release voltage is at the time of the next note trigger, adding that voltage to the total envelope voltage. This phenomenon allowed a performance trick that was popular with many Minimoog players — by rapidly retriggering such a sound, you are able to produce dramatic volume and timbral crescendos — and all without

The Keyboard view can be seen in the lower half of this window, with the Sequencer/FX page above. The three-row, eight-step sequencer is on the left, while the basic analogue-style effects modules (fixed filter bank, dual delay, and chorus) can be seen on the right.

a velocity-sensitive keyboard.

These points may not unduly concern every player. Nevertheless they have been communicated to Arturia, who inform me that they are currently working on solutions to include in future upgrades.

Following this communication, I noticed one or two other bug-like matters:

- When you select the Blank Synth preset, oscillator 2's tuning is +5 semitones sharp, and its sync source is set as oscillator 3, when it should have no connection. This is not a problem — you simply have to correct the errors and resave the patch.
- Slightly more significant is a problem when trying to use any oscillator as an LFO with its triangle-wave output set to its 'Lo' range. Basically, it doesn't work — until you switch the range from 'Lo' to 32' and then back again. Then it only works for one note — after that it fails again.
- Listening to any oscillator's triangle wave set to 'Lo' (it should be practically inaudible) produces an audible sawtooth-like wave fixed at a pitch equivalent to MIDI note B4 which modulates in amplitude. This fault would seem to be related to the above problem.

Concluding Remarks

MMV performed very reliably during the review period, causing no abnormally unruly behaviour under *Sonar*. CPU usage for each monophonic instance of *MMV* averaged around six percent when idling, rising to 10 percent when playing (depending on patch complexity). I found that four monophonic instances, all playing simultaneously, were peaking at around 35. Polyphonic instances are rather more CPU hungry, especially when using any significant release envelope time. Typically, a three-oscillator, four-note polyphonic sound with a 500ms release time will take the CPU meter up to 20 percent with a held chord, and as high as 60 percent if you start moving around! Therefore, polyphonic patches should be used carefully unless you have a pretty serious computer.

I have to say that I've had immense fun playing with *MMV*, whether creating bizarre noises, thundering Taurus-pedal impressions, Minimoog-style leads or huge, mythical nine-oscillator Memorymoog brass patches. The successful design of particularly complex patches provides a satisfying sense of achievement! With what amounts to 11 different filter types at your disposal, it's possible to achieve vast tonal flexibility, in particular with the 12dB-per-octave multi-mode filter, which allows the creation of many distinctly un-Moog-like tones. But I suspect it's the 'Moogy' potential that will initially draw most people to *MMV*, and from the tonal point of view the results are remarkably convincing. The various authenticity issues discussed above — the resetting envelopes, waveform-cycle resetting, non-positional glide time and lack of filter self-oscillation — might be a cause for concern depending on how purist your views are. However, Arturia are keen to continue refining *MMV*, and seem anxious to produce future upgrades that will address some of these matters. All in all, *MMV* offers endless creative potential, is enormous fun, and minor authenticity niggles aside, it really does sound great. SOS

information

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Studio SOS

This month the intrepid *SOS* team travel to Wigan to address Allan Murrell's recording, monitoring and mixing problems.

Paul White

Allan Murrell has a very enviable studio setup made all the more attractive because it is set up in a spacious farmhouse near Wigan where he has a separate control room, a spacious live room and a smaller adjoining isolation room with a window onto the live room that can be used for recording vocals or drums. These are all repurposed from an entrance porch, dining room and living room — apparently to the dismay of his long-suffering girlfriend! Rather than go the computer route, he has based his system around a Mackie HDR24/96 24-track recorder, a Mackie D8b mixing console (with numerous plug-ins) and a pair of Mackie HR824 active monitors, backed up by a pair of Genelec 1029As. Just in case you think he's bought shares in Mackie, he also has a rack of outboard processors that includes a Lexicon MPX500 reverb, a TC Electronic M•One XL reverb and a TC-Helicon Voice One vocal processor, as well as some nice analogue boxes, including a TL Audio Ivory 2 valve compressor.

He also has a good selection of mics, including the budget Superlux drum set, an AKG D112 kick-drum mic, some Rode NT3s and (his current favourite) a Rode NTK tube mic. Though Allan set up the studio for his own use, it soon became clear that there was a demand from bands and artists in his area, so the studio is becoming increasingly commercial. To enable bands to get good results very quickly, the live room is set up with a Roland V-Drum kit (with real cymbals and often a real snare drum), two Line 6 Pods and one Line 6 Bass Pod, the latter three set up on stands with headphones for the players. There is also an acoustic drum kit for those that require or prefer it.

Allan called us in because he'd had some problems with his mixes not sounding as good on systems outside his studio, and he was also concerned that his small live room made vocals sound slightly coloured. Additionally, he wanted us to check over some mixes he was working on, as he felt he wasn't getting the best possible sound from all the tracks and wanted to see if he could get cleaner-sounding mixes. After a surprisingly uneventful trip up the M6, Technical Editor Hugh Robjohns and I found ourselves sitting in Allan's control room drinking coffee and negotiating a plate of chocolate cake — Allan had clearly read previous reports and was aware of the requirements regarding esurience displacement!

Control Room Troubleshooting

We decided to start with the large control room, which was rather unfortunately proportioned, being almost 14 feet square and a little less than half that in height. Having dimensions that are close to being equal to or multiples of each other tends to exaggerate room mode problems, though the two large windows and one door helped to act as bass traps here. Before suggesting anything, we listened to a selection of Allan's mixes and some commercial CDs through the system before deciding on a course of action. The room tended to produce a slightly muddy sound with less-than-ideal stereo imaging, and one or two bass notes really set off the room's standing waves.

Part of the problem was that the Mackie monitors were standing on outrigger shelves on the IKEA computer table that formed the control centre for the studio. These were not particularly rigid shelves, and even though the speakers were standing on dense foam blocks, both Hugh and I thought the bass end could be tightened up by mounting them on proper stands. We also felt that reflections from the wall behind the monitors should be killed if at all possible, something we tested by first fixing a duvet to the wall, then adding some profiled foam sheeting when we confirmed that it did indeed help tighten up the sound, particularly the stereo imaging. As Allan had plenty of spare foam sheet, we also fixed a couple of panels to the ceiling above and slightly in front of the mixing position (using spray adhesive), to damp down reflections from the low ceiling.

Comparing the Mackie monitors with the pair of Genelec 1029As Allan had for reference, we came to the conclusion that the Genelecs were sounding rather boxy, with too much bass, while the Mackie monitors were slightly bass light in their current position well away from the back wall. Setting the rear-panel switches on the Mackies to normal for the high and low ends (rather than the middle setting for the bass switch) and leaving the speaker environment switch set to 'full space' proved to work rather well. Hugh needed to set about the 1029As with a screwdriver to move the heavily recessed DIP switches to setting three so as to introduce around 2dB of low cut starting at around 500Hz. We also moved the Mackie monitors inwards towards the support side of their shelves, to try to improve their stability, moved them forward and angled them inwards, aiming at the back of the mixing chair.

Because Allan had read one of our previous Studio SOS pieces where there had been problems with sound from Genelec 1029As reflecting off the computer monitors, he had installed more foam between his monitors and speakers. However, in this situation there was actually little risk of reflections, since the speakers were slightly in front of the monitor screens with this particular setup.

Once all these adjustments were complete, the sound was noticeably tighter, and the two sets of monitors were in closer agreement as to how the mixes should sound. A duvet was hung over the door in the side wall to damp reflections, while the curtains were drawn on the window opposite to cut down on reflections from the glass (as well as sunlight causing glare on the computer monitors). We also suggested moving the entire equipment rack back a little closer to the rear wall, but as it needed to be completely stripped down before it could be moved, we left that for Allan to try at his leisure...

Live Room Acoustic Problems

The small live room presented a different problem, as our speech immediately sounded coloured when we walked in there, even though almost all the walls and windows were covered in more of Allan's profiled foam. I guessed that part of the problem was down to the foam being fairly thin and very lightweight, with a fairly open cell structure, which meant it was probably only absorbing effectively above 500Hz or so, mopping up all the high end but leaving the lower mid-range and bass frequencies to

Reflective surfaces in the control room were causing Allan monitoring problems. Following the obligatory experimentation with duvets hung in various locations (facing page), it was decided that absorption was required not only behind the monitors, but also on the ceiling above the monitoring position.

Although most of the surfaces around the vocal mic were covered with open-cell foam, the relatively thin foam used only affected the sound above 500Hz, making for a rather coloured vocal sound. Doubling up the foam thickness in this case helped to absorb more low frequencies, providing a more balanced sound.

reflect freely. Actually the large window area probably helped with the deep bass, as most of this would pass straight through, but frequencies in the 150-350Hz region were definitely dominating the room. Allan was also recording with the mic set up very close to one wall, so we suggested removing the foam from some of the window area at one end of the room and using it to double up the foam at the other end to form a well-damped corner where the singer could stand (back to the corner). We fixed up the foam in a temporary fashion, moved the mic stand, and straight away the sound was more open, with a better balance of mid-range and high frequencies.

We also felt that replacing the foam on the wall opposite the window with a thicker, more dense type (three to four inches thick) would control the low mid-range even better. However, as nearly half this wall was given over to the window looking out onto the live room, this wouldn't kill any flutter echo between the two windows.

As predicted, there was some noticeable flutter echo at the opposite end of the room to the vocal corner, where we'd removed the foam, which might be a problem when recording drums, but the vocal end was fine. As Allan wanted to get better sound isolation between the big live room and the small one, we suggested that, as the house walls were very thick, he could remove the single-glazed window dividing the two rooms and replace this with a double-glazed one, set at an angle to the window opposite. This would kill the worst of the flutter echo and significantly improve the isolation. As Allan had a double-glazing engineer as a studio client, this seemed to be a very promising avenue!

We also noticed that a lot of fine dust was coming off the foam, and this turned out to be the polyurethane breaking down due to the action of sunlight. Where such foam is to be used in direct natural light, it is best to cover it with fabric or, where it is to be used in a window opening, to fix it to hardboard or MDF with the board facing the window. Painting the board black provides a professional and neat finish for anyone looking at the window from the outside.

Fixing The Mix

Allan had been working with a rather good band that clearly had Pink Floyd/Roger Waters influences and, though his mix didn't sound bad, it didn't have the punch and clarity that he wanted. My first experiment was to try the Acuma Labs *Final Mix* plug-in on the D8b to attempt some mastering-type processing, which involved low-ratio, low-threshold three-band compression and a touch of the inevitable 'air' EQ (gentle wide boost at 16kHz in this case).

I reset the processor's crossover frequencies to 150Hz and 5kHz so as to leave the mid-band intact and used the gain settings in the three compressor bands to fine-tune the tonal balance. This opened up the mix quite noticeably, but there were some track EQ and balance issues we felt we could improve on, so all three of us went through the mix a channel at a time and checked the quality of the basic sounds while looking at the processing Allan had used.

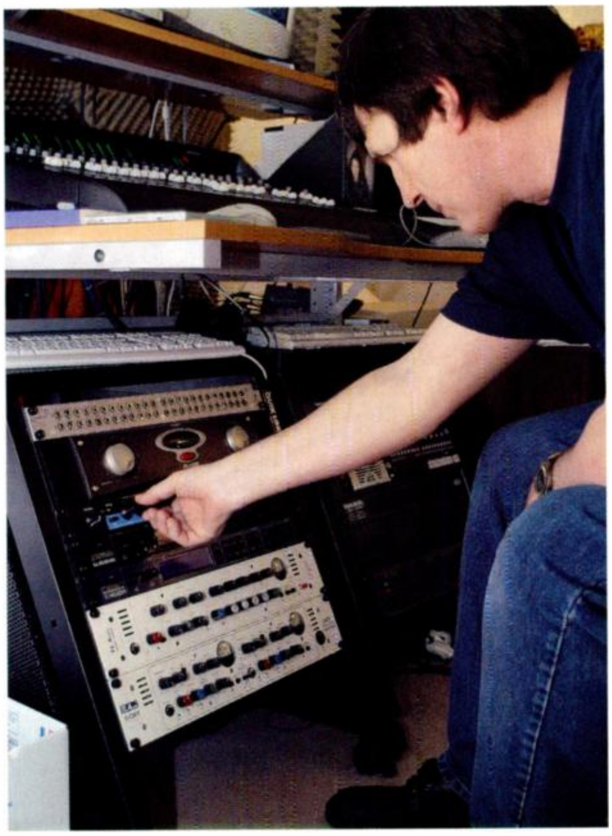
Allan felt he had got into a bit if a rut with his EQ settings and was applying some 'by habit' curves in situations where they might not be the best thing to use. Furthermore, he had used some compression and EQ while recording and, while this can be perfectly valid in some circumstances, it made it very difficult to approach the mix with a clean slate. Our suggestion was for Allan to make future recordings with flat settings and minimal compression, at least until he got a feel for what could safely be done at the recording stage.

One of the first changes was to edit the Plate reverb setting being applied to the drums via the Lexicon MPX500, primarily by

Some *Auto-Tune* Tips

Allan also mentioned that, although he had the *Auto-Tune* plug-in for the D8b, he'd never really got the hang of using it, so we called up a vocal track, established its key by playing along with a guitar, and set about adjusting the controls after setting the key input in *Auto-Tune*'s plug-in window. The most important control is the one that sets the speed at which the pitch is corrected, and once I'd figured out that this moved the opposite way from its VST plug-in counterpart (fastest correction when fully down), it was easy. The most natural correction was achieved by setting the correction speed slider about a third of the way from its top end (slowest position) where the pitch-correction display confirmed that corrections were only being made on sustained notes and that natural pitch variations were being allowed through intact. Allan was quite pleased to see this working so well as he had worried that it would sound unnatural.

HDSP 9652



The Plate reverb setting which Allan had used on his drum tracks was muddying the sound, so Paul reduced the bass reverb decay time and increased the pre-delay to get a tighter sound.

reducing the bass multiplier value from 1.0 to 0.6 and increasing the pre-delay to around 60ms. This took a lot of the muddy mid-range out of the reverb, resulting in a crisper drum sound.

The first sound we felt needed improving was the fretless bass, where Allan had used some chorus, compression and EQ. The chorus sounded fine and the compression wasn't too far off, but the EQ he'd set up had a deep notch in the low mid-range, which meant there was plenty of deep bass but no real definition. Removing this dip and adding boost centred at around 250Hz immediately reinforced the woody quality of the instrument. Once the rest of the mix was brought back in, it was obvious the bass sound was sitting much better. It sounded better defined and was punchy without sounding overblown — and it could still be heard clearly with the monitors turned down, when deep bass tends to disappear.

Still at the bass end, the kick drum was also lacking definition, something we remedied by adding quite a large amount of boost centred at around 4.5kHz and balancing this with an 80Hz hump. There was also some rather muddy ringing so we invoked the channel gate and adjusted the release time until we had what we felt was a tight, punchy kick sound.

This was the first time I'd used a Mackie D8b in anger and, though the EQ sounded very musical, it seemed to need far more cut or boost to get the job done than an equivalent analogue EQ. Typically I might use 2-3dB of boost on an analogue EQ, but to fix this kick sound I found I was adding more like 10dB of boost! This is quite common with digital desk equalisers in my experience, although some of the later algorithms are a lot more analogue-like in this respect. Allan had used a dual-mic approach for the kick drum, with a D112 inside close to the beater head and a Superlux kick mic a little further out. Again, we had to EQ this second mic heavily in the 4-5kHz region to give it some attack, then we mixed it in with the main kick sound, but a few decibels lower in level, to provide a fuller and more rounded sound than the close mic could provide alone.

The snare had also been dual-miked with Shure SM57s, one mic above and another with the signal phase reversed on the bottom head. The upper mic needed some EQ boost at 6.1kHz to make it crisper, then the low head mic was boosted at 7.4kHz and brought up in level until we had a suitably convincing snare sound. Boost was also needed to freshen the hi-hats, in this case at 8kHz with a corresponding 140Hz high-pass filter to reduce the low-end leakage from other sources.

Acoustic & Electric Guitars

The 12-string acoustic guitar was peaked at 11.5kHz and dipped at 200Hz, the latter to cut boxiness from the sound, while a six-string acoustic was boosted at 10kHz to add shimmer. The compression Allan had applied to both these acoustic guitars was reduced until the gain reduction meters were showing no more than around 5dB of gain reduction, and the attack time was set in the region of 20ms to allow the attack of the sound to come through clearly.

The track also featured a flanged electric guitar that was tending to clutter up the middle frequency region of the mix, so it was thinned by notching it at 150Hz and adding a 4.5dB boost at 2.2kHz. A high-pass filter was set at 50Hz to take out unwanted low end, then the guitar rebalanced in the track. A lead guitar that was also taking up a lot of space in the mix was similarly treated, but with a 210Hz notch and a 2.2kHz peak. The result was a powerful guitar sound that didn't stomp all over the vocals.

Moving back to the percussion tracks, the Tambourine was treated to heavy boost at 5.1kHz plus low shelving cut below 150Hz, and then the digital trim control had to be backed off by about 5dB because the HF boost was causing the channel to overload on peaks. A shaker was brightened by

Although the digital EQ in Allan's Mackie D8b mixing console sounded musical enough, much more drastic settings were required to solve his mix problems than would have been required using analogue EQ.

adding a relatively narrow boost at 9.5kHz. While all the EQ settings seemed quite drastic, they were needed only because that was the nature of the digital EQ being used.

The outcome of our endeavours was a much more transparent and open-sounding mix with better-defined drums and bass, complemented by lively acoustic guitars and percussion. It was brighter than Allan's original mix, but sounded more natural and airy, and no longer required the mastering EQ we had used to 'polish' the original version. The last job was to try to maximise the level of the mix so that it would sound loud when compared with commercial CDs, a job we attempted using the *Final Mix* software by setting the compressors to an infinite ratio and fast attack time to act as limiters, and also engaging the soft-clipping function. This bought us a noticeable amount of extra level, but wasn't as intuitive to set up, or as effective, as dedicated mastering software/hardware with a separate limiter.

Mixing Multitrack Drums

Allan also brought out a multitrack recording of an acoustic drum kit which sounded a little unnatural, partly because of the way the kit was tuned and played, but also because he'd EQ'd the toms to remove what he considered was excessive high end, presumably to minimise cymbal spill. Though most of the energy in a tom resides between 100Hz and 250Hz, the high end needs to be there to preserve stick definition, so we restored the missing high end and also beefed up the kick drum at 80Hz and 4kHz to give it depth and attack. Apparently the drummer wanted a ringy piccolo snare sound, but this wasn't really in the character of the drum. Nevertheless, by cranking up the mid-range boost, then sweeping it between 2kHz and 4kHz, we found a sweet spot that improved the sound. Once located, the boost was pulled back to a more practical level.

It proved to be impossible to gate the toms (to try to clean up the mix a little) because of excessive spill from the snare which caused false triggering. This seemed very odd given that the tom mics were apparently mounted just over the tom head rims, and implied that either the drummer didn't hit the toms very hard at all (unlikely!), or that they had been recorded with some compression in an attempt to create a fatter sound. Once again, this shows that it is often better to record the original signal raw, then apply gating and compression during the mix.

The kit had been recorded with stereo overhead mics, and Allan had taken the low end out of these during recording. All they needed was a bit of high-end encouragement to lift the cymbals out, something easily achieved using boost in the 6-8kHz region. However, while this approach was fine for the intended purpose, it also meant that there was no possibility of using the overheads to form the basis of a complete kit sound, with the close mics then being added for extra definition, because the low end of the drums had effectively been EQ'd out. In other words, the mixing options had been severely limited with no real gains — another example of why it usually pays to record flat and with minimal compression, to keep all options open.

Allan had also gated the snare-drum mics quite hard, which affected the attack and resonance very audibly. The overall sound improved considerably when the gate range was set to around 10dB rather than infinity. This allowed a low level of spill to survive, which seemed to help the overall kit sound gel with the overheads, and it also improved the attack of the drums because some of the original transient could still be heard even though the gate took a short time to open fully.

In A Nutshell

The room problems we encountered were typical of those experienced by many project studio owners, and again we demonstrated that the worst of these problems could be improved at a relatively low cost. However, the control room still had a somewhat uneven bass response, largely due to its unfortunate geometry, so we suggested Allan fit two foam corner bass traps to each of the rear corners, in addition to finding a tidier and more permanent solution to the layer of foam and duvets we'd fixed to his front wall. We also felt that some one-metre-high speaker stands, ideally filled with sand, would tighten up the bass sound further.

Once the monitoring was sorted out, the reasons for the mixing problems became more evident, and it transpired that Allan had been perhaps too enthusiastic with his EQ and compression settings during recording, which had forced him into a bit of a corner when mixing. In a lot of cases, Allan had also been applying (almost as a habit) small amounts of lower mid-range cut at similar frequencies to try to reduce the perceived mid-range clutter in his mixes, rather than using the EQ to bring out the important elements of each source individually.

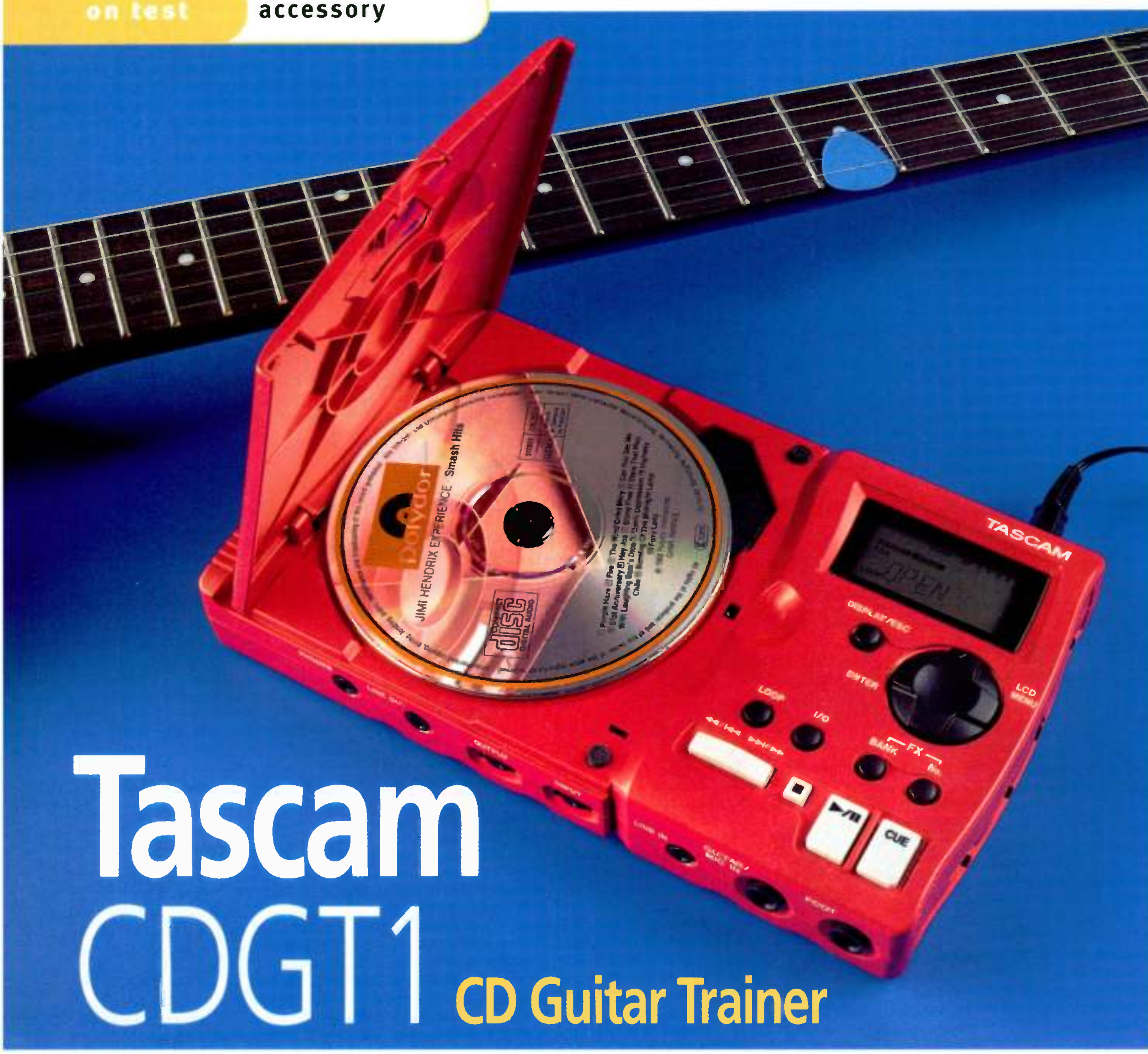
During mixing, we tended to use less compression than Allan had (over-compression can rob a sound of punch and clarity) and our final EQ settings were determined entirely by ear, simply by sweeping a parametric EQ across the frequency range while set at full boost to determine the best frequencies to cut or boost. Once these key frequencies have been identified, the cut/boost and bandwidth can also be adjusted by ear to give the best subjective sound. We also stressed the importance of making the final EQ adjustments with all the tracks playing, as what sounds great in isolation might sound quite wrong when the whole mix is up.

As a rule, EQ boost should be as broad as possible for a natural sound, though, in the case of drums and percussion, you can sometimes get a musically useful effect by making it narrower. EQ cuts on the other hand should be made as narrow as possible while still getting the job done, so that you're not taking anything more away from the sound than you need to.

On an artistic level, we also pushed the drums and vocals a little further forwards in the mix to get a more modern sound, while keeping any 'thickening' sounds further back in the mix to avoid congestion. A lot of modern mixes have most of their energy in the vocal, drums and bass parts, with everything else sitting lower in the mix, so choice of sound rather than sheer volume allows those parts to remain audible and distinct. SOS

Allan's Session Notes

"I must admit that I was a little apprehensive about the visit, fearing that the guys might highlight too many imperfections in the setup, but far from it. I asked Paul and Hugh if they could concentrate on the acoustics of all the rooms and have a listen to some of my mixes to identify any fundamental mistakes I was making. A few changes to the positioning and settings of the monitors, addition of acoustic tiles and heavy drapes here and there and the room was sounding much better. More importantly, they suggested ways in which I could improve my mixes and, I have to say, since the visit, my clients and I have noticed a massive difference in the quality of the mixes."



Tascam CDGT1 CD Guitar Trainer

Paul White

Tascam's colourful new CDGT1 lets you easily practice playing or singing alongside your favourite CD.

Tascam have clearly identified a need for a device that will help guitar players or singers work out songs from CDs by allowing them to slow down the music without changing its pitch. However, their CDGT1 Guitar Trainer goes rather further than that, as it also allows the key of a piece of music to be changed without affecting its tempo. The guitar input has its own effects section to allow you to play along with the CD you're trying to learn.

Seeing Red

Solidly moulded from alarmingly red plastic and slightly larger than a VHS video tape case, the Tascam CDGT1 can be powered from four AA batteries or from an optional external mains adaptor. It has a small but perfectly adequate LCD window to handle the CD play functions (as well as parameter editing) and a Gameboy-style four-way rocker switch for parameter access and value changing. The Display button doubles as an Escape button during editing, while Loop and I/O buttons allow two points to be marked within a song while it is playing so that these can form the start and end of a loop for rehearsing a defined section of the song. Bank and Number buttons are provided for accessing the onboard effects banks and user memories, and the regular CD transport controls are augmented by a Cue button that returns the playback position to wherever playback was last started. Cue can be set up so that the player pauses when Cue is pressed or it can go straight to the playback point and start playing. Provision is made for an optional footswitch, which can be set up in the menu ▶

SOUND ON SOUND

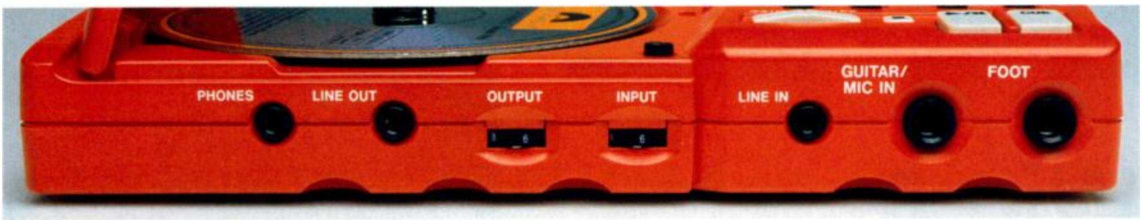
Tascam CDGT1 £149

pros
- A great all-in-one practice station.
- Can change pitch and tempo independently over a useful range..
- Pitch-shifted audio quality remains surprisingly acceptable on most material.

cons
- The guitar overdrive effects are somewhat basic.

summary
A useful and portable tool for anyone who needs to practice or learn new material.

TASCAM CDGT1



The audio I/O and level controls occupy the front edge of the unit, alongside a footswitch jack which can control the cue function or the internal effect selection.

either to act as a Cue controller or to step through the effect presets.

Connections

The stereo phones output, line input and line output jacks (3.5mm) are on the front edge of the case, with thumbwheels to control the input and output levels. The stereo line input is mixed with whatever else is going on in the CDGT1, enabling external audio sources to be mixed in. For example, you may wish to plug in your favourite modelling guitar preamp rather than using the onboard processing. The mic/guitar and footswitch sockets are conventional quarter-inch jacks, where a switch on the right-hand side of the case toggles between mic and line input modes. There's also a Hold button that locks the current control status when the machine is on to prevent accidental interference with the controls. Curiously, there's also a recessed Guitar power switch that powers up the mic/line input and effects section. This is in addition to the main power switch close to the power adaptor socket, and it must be on to use the mic/guitar input. It's my guess it is included to extend battery life when the mic/guitar input isn't needed, as the digital effects chip and preamp circuitry probably draw a significant amount of power.

The CD player behaves pretty conventionally until you dip into the menu and select one of the special modes. Key mode enables the key of the recording to be changed by up to six semitones either way (in semitone steps) without affecting the tempo. SSA mode does the opposite, and allows the tempo to be reduced by up to 50 percent without affecting the pitch, while Pitch mode acts rather like the varispeed on a tape recorder and lowers the pitch and tempo together, once again by up to 50 percent, in one-percent steps.

The onboard effects are arranged into two banks, where numbers one to five in each bank can be used to store user settings. There are 43 presets in all, 30 designed for guitar and 13 for vocals. As you might imagine, the degree of editing is fairly limited and values are changed using the up/down sections of the four-way rocker switch, but you're provided with adequate adjustment to control things like guitar distortion amount, compressor gain, tone, degree of pitch-shift and amount of chorus, flanger, phasing, tremolo, delay or reverb. The vocal presets, on the other hand, have only one adjustable parameter. There are also auto-wah effects, panning and vocal de-essing. Guitar presets combine four effects blocks (headed up by either distortion or compression). There's also a handy guitar tuner that works via a row of dots in the display. When the pitch shows in the middle of the row, it's correct. Normally the note detection automatically recognises which string you're tuning, but, for oddball tunings, you can opt to set a note value manually and then tune to that instead.

Training Session

So, how well does the CDGT1 stack up as a 'guitar trainer'? After several hours of using it, the guitar still couldn't do somersaults, fetch slippers or jump through hoops, so I'm afraid it's *nul points* in that area! However, as a *guitarist* trainer it's a bit of a gem. The audio quality is comparable with any decent portable CD player, though, as you might expect, it takes on a hint of the characteristic pitch-shifter warble when you use the Key or SSA modes. Even so, it's perfectly acceptable to play along to, and the slowed-down settings make it a lot easier to figure out what's being played. Used as a straightforward CD player, it behaves conventionally, but seems reassuringly resistant to skipping caused by bumps or vibration. It's also happy playing CD-R audio discs.

The guitar effects are fine for practice purposes, though the overdrive sounds are rather too 'fizzy' for my liking and I wouldn't choose to record using them. I preferred the clean and chorused guitar presets as these sounded rather more musical to my ear. The vocal effects fare a little better, as they don't usually have any distortion settings and tend to use mainly effects such as reverb or delay, though there are some interesting lo-fi frantic panning and megaphone presets for the more adventurous.

Red Giant Or Red Dwarf?

Although the CDGT1 costs rather more in the UK than a typical portable CD player (and is arguably less portable), it manages to double as a high-quality music playback system and a comprehensive practice aid for vocalists (using an unbalanced mic with a jack lead) or a practice/learning aid for guitarists. While the guitar effects are a little unrefined compared with the latest in modelling technology, they are absolutely fine for the purpose intended and the only trick I can think of that Tascam have missed is in not adding a central image voice canceller, which would have allowed singers to attenuate the vocal part on the CDs they want to sing along to. SOS

The right-hand side panel carries switches for selecting mic or line input circuitry; engaging hold mode; activating the preamp and multi-effects facilities; and powering up the whole unit.

information

- £ CDGT1, £149; optional power adaptor, £12. Prices include VAT.
- T Tascam +44 (0)1923 438880.
- F +44 (0)1923 236290.
- E info@tascam.co.uk
- W www.tascam.co.uk
- W www.teac.co.jp

What Mic Should I Buy First?

We answer one of the most common queries from those just starting out with home recording.

Paul White

When it comes to setting up a home-based project studio, deciding which microphone to buy first can be pretty daunting, not least because of the huge number of models and types available. The good news is that today we have a choice of some extremely good, low-cost microphones, many imported from China or the former Eastern Block, all of which perform significantly better than anything we could afford when home recording first took hold in the late '70s. This strong competition has also resulted in European and US manufacturers launching budget mics, something made possible by the higher volumes of sales generated by the growing project studio market.

Dynamic Or Condenser?

Before trying to pin down what to buy, it pays to be aware of the main differences between dynamic mics and capacitor mics. Dynamic mics work on the moving-coil principle, rather like a loudspeaker in reverse, and have in their favour mechanical durability, cost-effectiveness and a solid, punchy sound that works well for guitar, bass and drums, as well as other loud instruments such as brass. They also need no phantom power to work (see the 'Phantom Power' box for details), which makes them very popular for live use, but working against them is the fact that their high-end response isn't so good as a typical capacitor mic and they are also relatively insensitive. In practical terms, this means that sounds relying on a lot of high-end detail, such as cymbals, acoustic guitars, pianos and even some voices can sound restricted in the upper frequency ranges if recorded via a typical dynamic mic, though there are exceptions which have a more extended frequency response. Typically, though, dynamic mics are good up to around 15-16kHz, above which their sensitivity tends to drop off quite drastically. The overall sensitivity of the mic determines how much gain you have to add on the mixer or mic preamp to bring the output up to the required level and, while dynamic mics are adequately sensitive for close-miked vocals and fairly loud instruments, they struggle with more distant sounds or quieter acoustic instruments. In these situations you have to add more gain at the mixer and more gain invariably equates to more background hiss.

Capacitor mics (or at least those used in studios) fall into two categories — true capacitor mics and back-electrets. A true capacitor mic uses a very thin film to form the diaphragm, coated with a conductive metal such as gold, and, because there is no heavy voice coil attached to the diaphragm, it puts up less resistance to being moved at high frequencies. The diaphragm forms part of an electrical capacitor and is charged via a polarising voltage enabling it to convert movement to a change in voltage. Phantom power is needed to drive the on-board preamp electronics and to polarise the capsule.

A back-electret mic may use a similarly constructed diaphragm and can produce the same level of performance as a conventional capacitor microphone, though models designed for use with batteries are usually less sensitive than their 'phantom power only' counterparts, especially if they are designed for live use. The principle of operation is similar to that of a conventional capacitor mic, except that, instead of needing an external polarising voltage, the capsule's back-plate is covered with a ▶

Peaks & Presence

A microphone can be engineered to produce a very flat frequency response extending from below 20Hz to over 20kHz, though this isn't always desirable other than for measurement applications. Most models have either an in-built low-end roll-off to keep unwanted deep bass out of the signal chain or a switchable low-cut filter. At the high end, the response curve may be shaped to produce a particular sound. A boost at 3-4kHz provides presence, while boost above 10kHz gives more of an airy high end. Some capacitor models may also be rolled off gently at the high end to give them a warmer sound. For general-purpose use, a mic with a fairly flat response is the most flexible, though a gentle peak at 8kHz or above is usually pretty benign. I'd only recommend choosing a mic with more extreme coloration if you intend to use it mainly with one voice and you've found a particular model to suit that voice.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The AKG C1000 back-electret mic can be powered either from a normal phantom supply, or from a 9V battery fitted inside the mic casing. The option of battery power could be very useful if your mixer or recording channel has no phantom powering option.

material that carries a permanent electrical charge within a highly insulating film (the electret material). A voltage is still required to run the onboard preamp, though, so power may come from phantom power and/or batteries depending on the model. An example of a popular back-electret mic that offers the same performance as a regular studio capacitor model is the Audio Technica 4033. Dual battery/phantom models, such as the well-established AKG C1000, are often less sensitive, so that they are able to match the requirements of live sound close-miking. Capacitor mics are used in the studio for most vocal and acoustic instrument recording as well as for drum overheads. They may also be used for recording electric guitar, where they deliver a useful alternative to the dynamic-mic sound.

Polar Patterns & Diaphragm Size

Although mics can be bought with omni, cardioid or figure-of-eight pickup patterns (and every stage in between), the cardioid response is the most useful in a project studio, as it excludes more of the room sound and spill from off-axis sources. Omni mics have a more natural, open sound than cardioids, but pick up equally in all directions and so tend to be used only when the room has a particularly supportive sound or where there is little risk of spill from other sources. Cardioid and figure-of-eight mics both exhibit a bass tip-up when used close to the recorded source, called the proximity effect, but omni mics don't suffer from this and so may be used very close to the sound source without the tone becoming more bass heavy.

Much is made of the way a mic's diaphragm diameter affects its tonal attributes, but this is more subtle than you might expect. As a rule, large-diaphragm mics have a slightly fuller sound, but are less accurate when picking up off-axis sounds. For example, in omni mode, a large-diaphragm mic may suffer noticeable high-end loss when used 90 degrees off axis. Conventional wisdom has it that large diaphragm mics (around one inch in diameter) are best for close-miked sounds, such as studio vocals and guitar amps, whereas smaller diaphragm mics (typically 0.5 inches or less in diameter) are the preferred choice for recording ensembles at a distance or for recording acoustic instruments. Having said that, either type of mic can produce perfectly acceptable results in either situation, so where you can only afford one mic, large-diaphragm cardioid models are a safe bet, as they should give good results most of the time.

Tubular Balls?

So, what about tube mics — they cost more, but is it worth it for that magical tube warmth? That depends on what you expect tube warmth to sound like, because the tube's contribution to the sound is subtle in a well-designed tube mic. To my ears, a well-designed tube mic has a slightly denser sound than a conventional solid-state model, almost as though subtle compression has been added, while the high end can be more open and detailed but without harshness. If the sound is audibly distorted or dull, someone has tried to design in a particular sound rather than rely on the natural mechanism of a good tube circuit to do the job. This is quite an acceptable thing to do on an artistic level, but be aware it may sound quite different to the classic tube mics so revered by those engineers privileged enough to use them on a regular basis.

The same is true of some tube circuits that run high-voltage tubes from a low voltage. There are special tubes designed for low-voltage use, such as some military devices or those developed for use in hearing aids, but most tubes are designed to run with a plate voltage of between 250V and 300V, which is why the vast majority of tube mics include their own power supply. In fact the only exception I know to this is the new Audio Technica 3060, which uses a low-current, low-voltage hearing aid tube and works from regular phantom power. It

Phantom Power

Capacitor microphones require a power supply to polarise the capsule and to drive the preamplifier which is normally located inside the microphone body close to the capsule. Back-electret mics and active DI boxes need power to run their onboard preamps. To avoid the need for a separate power supply, the majority of capacitor microphones (other than tube mics, almost all of which need a separate power supply anyway) and most studio-quality back-electret models use the universally standard 48V phantom powering system, though some models will operate at lower phantom power voltages at the expense of some loss of headroom.

The term 'phantom' was coined because no additional wiring is needed to carry the power from the power source to the microphone — it is passed along the microphone cable using the same wires that carry the audio signal. Phantom powering is only possible with balanced wiring systems (the cables have two cores and one outer screen) and most studio mixing consoles have their own internal phantom power supplies that feed their microphone inputs. Depending on the mixer, the phantom power may be globally switched or switched per channel. Separate phantom power supplies are available (some running from multiple batteries for mobile use), and all serious microphone preamplifiers also have integral phantom powering.

Dynamic microphones do not require phantom power, though where the phantom power on a mixer can only be switched globally it is safe to connect balanced dynamic mics provided that balanced cables are used. This is because the same phantom voltage is present on both the hot and cold pins of the mic cable and so no current flows. However, those mics not wired for balanced operation could be damaged if phantom power is applied.

is worth noting that some mics also include transformer output stages and these often influence the sound in a positive manner, but simply buying a mic with a transformer output is no guarantee in itself of a warmer or more musical sound. With both tubes and transformers, you have to listen to the specific model you're interested in and make up your own mind.

You Pays Your Money...

Having set out the background, how do you set about choosing that first mic? My first observation would be that, while most studio gear seems to become obsolete shortly after the credit card statement arrives, a good mic will last you a lifetime, so buying the best you can afford for your particular application, then looking after it, is a good investment.

There's little to be gained from buying a dynamic mic as your first choice for recording these days, as a good one will cost as much as a budget capacitor model and will usually deliver a lower-quality vocal sound. Its lower sensitivity may also show up noise problems in budget mic preamps. The exception is when you find a model that suits a particular voice — for hard rock vocals, a dynamic can work well, as it often delivers a more focused, punchy sound than a capacitor model. Naturally it still pays to buy a good dynamic mic rather than a cheap one as it will always remain useful, no matter how many other mics you buy afterwards. Although there are lots of good dynamic mics to choose from, the Shure SM57 is still a great all-rounder. That said, the Sennheiser MD421 is probably the model I'd choose if I had to pick one dynamic mic that would work acceptably well on everything, from vocals and guitar to brass and kick drums. If I was ever stuck on a desert island with no phantom power, the MD421 is the mic I'd choose!

Here you can see the relative sizes of typical large-diaphragm and small-diaphragm mic capsules.

Because large-diaphragm mics will give acceptable results on just about everything aside from a kick drum (I can't bring myself to subject a capacitor mic to those kinds of SPLs!), there's little point in buying a small-diaphragm mic as your first choice unless you intend to specialise in some sort of acoustic instrument recording, such as solo acoustic guitar. In such situations, there may be an advantage in a small-diaphragm model, and in most instances the same mic will still produce perfectly acceptable vocal recordings if used with a pop shield. It's just that large-diaphragm models are often more flattering to vocals than the relative honesty of a small-diaphragm model.

If you're only going to buy one mic, then a cardioid pattern is the clear choice. Figure-of-eight mics are most useful when you're recording two sounds at once and need to separate them as much as possible — and for this you'd obviously be working with two mics. Nevertheless, if you feel this may be a mic technique that you'll need to explore in the future, it could be worth spending a little

Although a condenser mic will usually be the best first buy for the home studio, if you decide to go for a dynamic mic then the Shure SM57 and Sennheiser MD421 are both industry-standard microphones with good all-round performance.

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▶ more on a mic with switchable pickup patterns.

...But How Much Should You Spend?

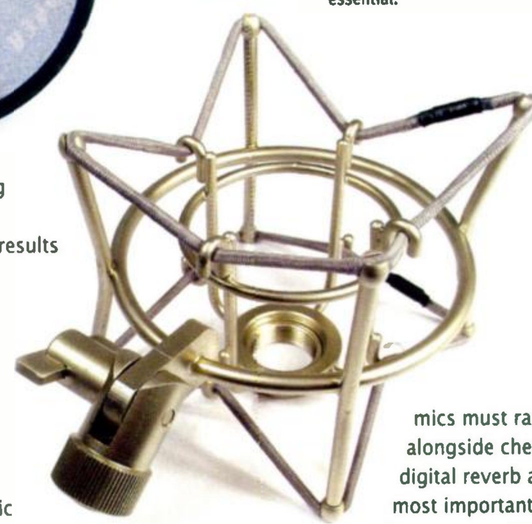
If, like many *SOS* readers, you're doing vocals in a bedroom studio, then pretty much any of the budget large-diaphragm cardioid capacitor models (typically under £150 in the UK) will do a good job — you're likely to run into the limitations of your acoustic environment long before those of the mic itself. Most of these mics are sensitive enough that they'll give good results even with relatively inexpensive mixers or preamps, but always use a pop shield and try to ensure that the space you use to record your vocals is free from excessive room reflections. If used reasonably close to the source, the mic will turn in a good noise performance, even with budget mixer preamps, provided that you take care not to choose a low-sensitivity back-electret model designed for live use. Also, it's worth paying the extra for a proper shockmount and pop shield (though many mics now come bundled with a shockmount) as these make a big difference to the quality of vocal recordings.

The same large-diaphragm cardioid condenser mic will serve you well for recording acoustic guitar, electric guitar and percussion. If you're recording a complete drum kit, a single capacitor may give you good results if you have a nice kit in a good-sounding room, but in most situations you'll need multiple drum mics, so if you can buy two of the same model, at least they'll double as stereo overheads. Similarly, identical pairs may also be used to record acoustic piano and I've made a lot of really nice-sounding recordings at my daughter's school, where all they have are four budget Superlux capacitor mics, a Behringer mixer and a Minidisc recorder. Better mics undoubtedly turn in a better performance, but the difference is often less than you imagine unless you happen to be working in a very good acoustic space — the law of diminishing returns applies as much here as anywhere else.

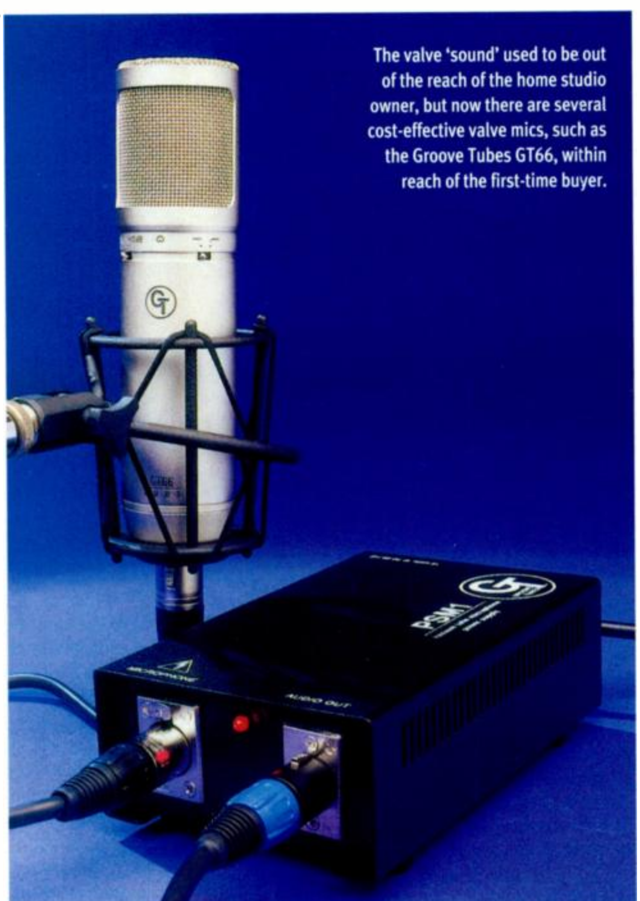
We get lots of enquiries about upgrading microphones to get a better vocal sound, but in most of the cases that we've investigated the most serious limitations to the sound quality are imposed by the recording environment rather than by the mic. You really can make pristine vocal recordings using capacitor mics costing as little as £100 and, in my view, the introduction of affordable capacitor mics must rank alongside cheap digital reverb as the most important factors in enabling project studio owners to make professional-quality recordings on a budget.

If your studio is a step up from the basic bedroom setup and you have a good space for recording vocals, then it's probably worth comparing these lower-cost mics with the mid-priced solid-state models or lower-cost tube mics such as the Rode NTK, Audio Technica AT3060, Groove Tubes GT66 or SE Electronics Z5600 to see if there's a noticeable benefit. Furthermore, if you're choosing a mic for yourself or for a particular vocalist, try to borrow or hire a small selection of mics within your price range so that you can try them in your own surroundings and then pick the one that works best for you. Once you start working with this class of microphone or above, it's probably worth considering getting a good-quality preamp to go with it, as you'll eventually reach a point where the benefits of a better mic are partially negated by the limitations of your mixer's mic preamps.

To be honest, I'd only recommend buying a mic worth £1000 pounds or more in the UK if you are running a commercial studio or if you are preparing vocal tracks in a very well-designed home studio that will be used to complete a commercial release. Having said that, the fact that a good mic can last for several decades if looked after might be reason enough to buy something a little bit special in exactly the same way as a weekend guitarist might buy a top-of-the-range Les Paul or Fender guitar. SOS



Large-diaphragm condenser mics tend to be quite sensitive to vibration and handling noise, so make sure that you use them with a decent suspension shockmount (below) — many of the newer mics come with one as standard. And if you're planning on recording vocals, a pop shield (left) will be pretty much essential.



The valve 'sound' used to be out of the reach of the home studio owner, but now there are several cost-effective valve mics, such as the Groove Tubes GT66, within reach of the first-time buyer.

Photos: Mark Ewing

WRH

Synapse Audio's *Orion* is now supplied platinum-coated and, with only a modest price rise over the *Pro* version, is also available in a cardboard box.

Synapse Audio Orion Platinum

Software Studio For Windows

John Walden

SOS last looked at *Orion* in September 2002 when we reviewed *Orion Pro* version 3 (read the review on-line at www.sospubs.co.uk/sos/sep02/articles/orionpro.asp). The overall impression was very positive: *Orion Pro* 3 offered an effective, *Reason*-like virtual software studio system, at a price that was significantly cheaper than its Propellerhead competitor. Of course, there were a few negatives. For example, support for audio tracks was very limited (as in *Reason*) and, because the product was only available as a download, few presets were provided for the *Sampler* generator module.

A new version, *Orion Platinum*, is now available. Amongst a number of improvements, the highlights include a new *Ultran* synth/sampler generator, a Master Section for the Mixer and, courtesy of being available in a cardboard box via retailers, a CD-ROM containing several hundred Megabytes of *Sampler* presets. A further change is in company name: Sonic Syndicate have now become Synapse Audio Software.

Of course, while *Reason* is the established market leader in this particular software niche, *Orion Platinum* also faces other credible competitors. For example, Image Line's *FL Studio* v4 and Cakewalk's *Project5* will both be available by the time you read this. For those on a budget, however, the pricing of the three flavours of *Orion* — *Platinum*, *Pro* or *Basic* — may be an important consideration.

Platinum Potential

Orion Platinum will run under Windows 98, ME, 2000 and XP, and minimum hardware requirements are a 400MHz PII, 64MB of RAM and 100MB of hard disk space, although 800MB is required for full installation of all the *Sampler* presets supplied. As with all CPU-intensive software, a faster processor and more RAM is likely to result in smoother performance. *Platinum* supports MME, Direct X and ASIO drivers, and the majority of this review was conducted under Windows XP Pro using the ASIO drivers of the Echo Mia installed on the test system.

The boxed version of *Orion Platinum* includes an install CD-ROM and a useful printed Quick Start Guide, which covers installation and configuration of *Platinum* as well as providing a series of short, introductory tutorials aimed at new users. More detailed documentation is provided via an on-line Help system — but it would still be nice to see this also available in PDF format for those who like to print a hard copy! During installation, the user can choose to

---

SOUND ON SOUND

Synapse Audio *Orion Platinum*
£130

pros
- Good set of tools for dance music production.
- Useful additional features including the *Ultran* synthesizer.
- Boxed version is good value for money.

cons
- Audio track facilities still limited.
- Documentation of some features could be improved.
- Very occasional stability issues on the test system.

summary
Orion Platinum offers some good additional features over the *Pro* version, and still provides a very useable and affordable environment for dance music production on the PC.

copy the 600MB of Soundfont (SF2)-format *Sampler* programs to the local hard disk. These address one of the criticisms made of *Orion Pro* 3 in my earlier review.

Orion Revisited

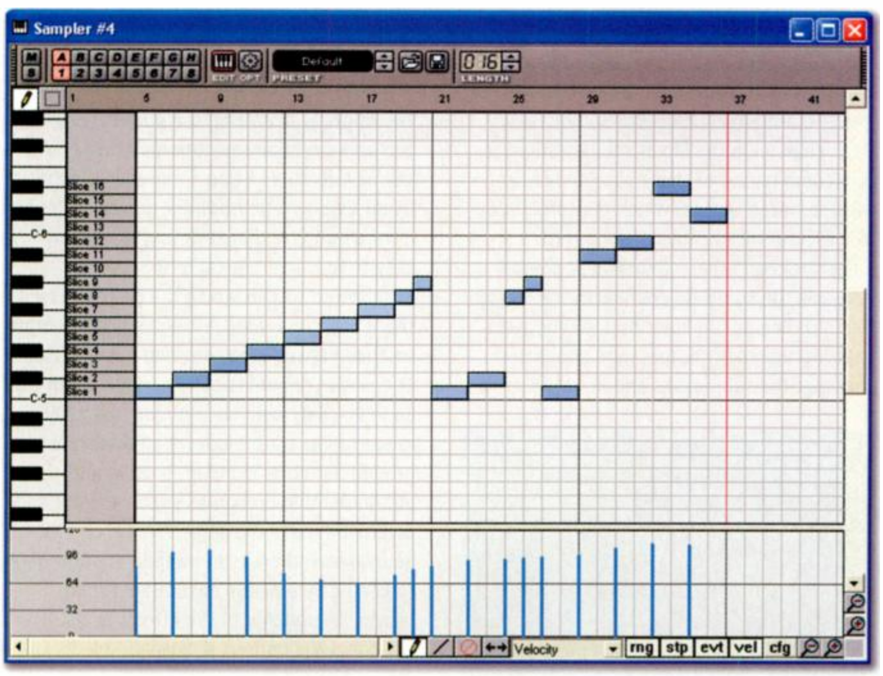
As the core functionality of *Orion* was covered in the earlier *Pro* review, a brief recap of the key features is all that is needed here. As can be seen from the screen shots, *Orion Platinum* looks rather like '*Reason* without the rack'. Sound is created by a selection of Generators (see the Generation Of Power box for a brief list) including various synths, drum machines and a sampler, much as in the *Pro* version. While *Orion Platinum* provides access to both Direct X and VST plug-in instruments and effects, the application also includes a set of proprietary effects and the number has been expanded over and above those in *Orion Pro.* Effects can be used via one of the four send-return busses or as channel inserts, with a maximum of two inserts per channel.

The main methods of music entry and song construction also remain little changed from *Orion Pro.* Sequencing is primarily pattern-based and each Generator includes 64 patterns. Pattern length and step resolution can be set on an individual basis. Well-specified piano-roll editing makes for a fairly flexible sequencing environment. Once patterns have been created, the Song Playlist is used to arrange patterns for each Generator along the timeline, all of which is very straightforward.

New Stars

Top of the additions to *Platinum* is the new *Ultran* Generator. The documentation describes this as a "wave-morphing sampler/synthesizer, capable of playing multisampled waveforms like the *Sampler*, but with additional blending features for multiple waveforms across each other". There is no doubt that the supplied presets include some very impressive programs, including pads, sound effects and analogue-style patches, while the combination of oscillators, LFOs and envelopes provides plenty of flexibility. On the down side, the documentation doesn't do a particularly good job of explaining how this potentially powerful synth operates, so if you like to build your own sounds from first principles, expect to have to do a certain amount of head-scratching.

The *Wavefusion* Generator is essentially an updated version of the wavetable-based *Wavedream* Generator provided in *Orion Pro*, and has undergone a graphic overhaul. Again, some nice pads and lead sounds are



Orion's piano-roll editing environment is both easy to use and well featured.

The new *Ultran* sampler/synth is capable of some excellent sounds, though the documentation could be better.

Test Spec

- *Orion Platinum* v3.8.5.
- Pentium 4 1.6GHz PC with 1GB RAM, running Windows XP Pro.
- Echo Mia 24, Yamaha DSP Factory and Yamaha SW1000XG soundcards.

Generation Of Power

***Orion*'s sound production is achieved using a range of Generators. In brief, these are:**

Ultran
New to the *Platinum* version, *Ultran* is a wave-morphing synth/sampler with four oscillators, two LFOs and three envelopes.

Tomcat
A monotimbral drum synth.

Sampler
A sampler module with multisample support, layers, two LFOs and three envelopes. Kurzweil (KRZ) and Soundfont (SF2) files are supported, and WAV and AIFF files can also be loaded.

Wasp
An FM-style synthesizer with three oscillators, two LFOs, two envelopes, six filter modes and an integrated effects unit.

Drums
A sample-based drum machine (although there is nothing to stop you using it to trigger other sample types) with 12 tracks.

Monobass
Analogue-style bass synth — simple but effective.

XR909
A combination of analogue drum synth and sample playback.

Wavefusion
This is a wavetable-based synth with three oscillators, three LFOs, three envelopes and three filter modes.

Plucked String
A basic physical modelling synth that does exactly what it says on the tin.

The MultiFX Control allows some creative effect manipulation to be achieved.

provided amongst the presets. Improvements to the *Sampler* Generator are mostly under the hood, with program editing now much more comprehensive, and in contrast to *Ultran*, program creation/editing for the *Sampler* is documented quite well. The *Sampler* now also loads Akai S5000/S6000 formats, and its *Recycle*-style Groove Slicer can now deal with stereo as well as mono audio files. However, the real advance is the presets provided on the CD-ROM, which include various acoustic instruments, pads, percussion, synths and basses. While the selection is fairly modest in number, what is here demonstrates that the *Sampler* is a perfectly capable unit. Soundfont programs are widely available for download via the Internet, so users could easily expand this collection.

Three changes to the effects provision are worth highlighting here. The new MultiFX Control provides a means of chaining up to four effects, either in series or parallel. In the latter mode, an X-Y graphic controller allows the levels of the four effects to be blended as required. As with all the synth and effect controls within *Orion*, movement of this controller can be recorded for real-time mix automation, either using the mouse, an external MIDI controller or via the appropriate Event Editor. If crunchy and grungy is your thing then the new *Lo-fi* effect might be useful, offering as it does bit reduction and the emulation of vinyl dust, clicks and noise — great for giving a well-recorded drum loop an authentic 'aged' feel! The *Rotary Speaker* effect is also worth a mention, and while it offers nothing too revolutionary (sorry!), it is capable of creating both subtle and not-so-subtle movement in sounds.

A further major improvement in *Orion Platinum* is the Master Section. This now provides return channels for each of the four send-return effects and four sub-busses. Each channel in the Mixer can be routed to any combination of these sub-busses or the Master stereo output, which is useful if you want to submix various drum-based Generators and then control their overall level via the sub-buss channel in the Master Section. The documentation also suggests that multiple ASIO outputs are supported and that the sub-busses and master stereo output can be sent to different physical outputs if required. However, I couldn't actually find any way within *Orion* of routing a particular buss to a particular ASIO output pair, and the documentation did not detail how this might be done — oops!

The new Master Section (on the right) provides greater control on send-return effects as well as providing four sub-busses, allowing submixing of selected channels from the main Mixer (on the left).

Orion Options

The upgrade from *Pro* to *Platinum* is more evolutionary than revolutionary, but all the key new features bring some useful options. Using *Orion Platinum* is not greatly different from *Orion Pro*; that is to say, it is pretty easy to use, and critical to this is the uncluttered way in which the pattern-based sequencing operates — there's just the right balance between features and ease of use. The same comments can be made about the Song Playlist, which is a doddle to use. The range of Generators covers all the same ground as those found in *Reason*, and although I think modules such as *NNXT* and *Maelström* have the edge over *Orion*'s equivalents, there is more than enough here to keep dance-heads on a budget quite happy. The quality of the effects is also good and the various delays add plenty of creative possibilities.

Like *Pro*, the *Platinum* version of *Orion* does include an Audio Generator, but the functionality is still fairly limited. Essentially it allows pre-recorded audio files to be played back alongside the rest of the arrangement but, as no audio recording is possible, if you want to add more than just a few pre-recorded vocal samples, then you are going to need some other software environment to complement *Orion*. This said, the same is also true of *Reason*.

My only other complaint about *Orion Platinum* in use is that I experienced a small number of crashes during the review period. I couldn't isolate any one particular cause for this, and neither adjusting latency levels of my Echo Mia's ASIO drivers, nor moving over to the ASIO drivers of my Yamaha DSP

The Very Latest

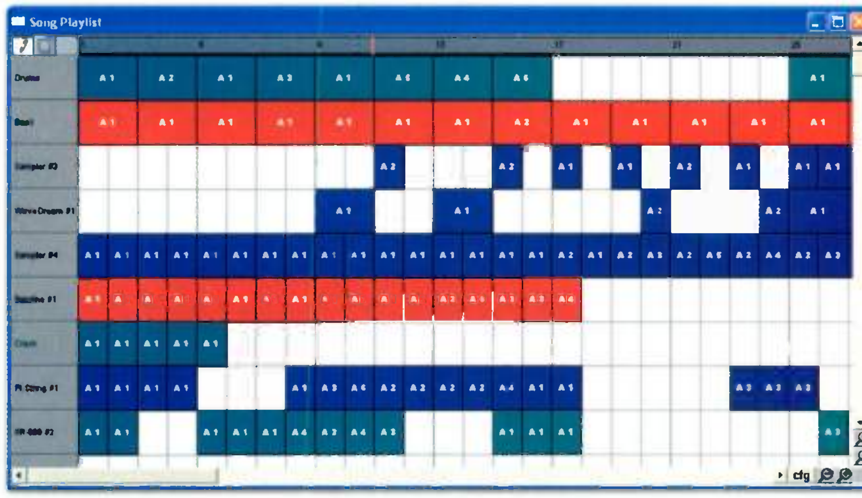
As this review was going to press, Synapse Audio Software announced details of *Orion Platinum* 4 and it should be available by the time you read this. The *Wasp* synth has undergone some cosmetic and functional changes including LFO delays, variable velocity amounts and keyboard tracking. A new effect, the *Harmonic Resonator*, has also been added. Other new features include parameter readouts on all controls, and MIDI export of song files is now possible. As well as some routine bug-fixes, both the *Plate Reverb* and *Ultraverb* are also a little less CPU-hungry. Version 4 will be a free download for registered users, and full details are available from both the Synapse Audio and Dark Horse web sites.

Factory/SW1000XG combination in the test system didn't seem to stop this happening on occasion. This certainly was not a major concern and, as a demo version of *Orion* can be downloaded (admittedly as a 15MB file!), it is possible to test it fully on your system before flexing your plastic.

Conclusions

It is impossible not to make a direct comparison between *Orion Platinum* and Propellerhead's *Reason*. Labelling *Orion Platinum* as '*Reason* on a budget' is very tempting and is probably not that far from the mark. This should not, however, be seen as a criticism. *Reason* is an excellent music production environment and, in *Orion Platinum*, Synapse Audio have struck an interesting balance, producing an application that is not that far removed from *Reason* in function, but has a considerably lower price.

If money was not an issue, yes, I'd go with *Reason*: in its particular niche, it is the industry standard and, with an ever-increasing range of Refills available, it is capable of working in a diverse range of musical styles well outside the dance genres for which it has become so popular. However, whether money is an issue or not, *Orion* is a credible alternative to *Reason* and potential purchasers should certainly give the demo version a thorough test drive before making a decision. Some may well prefer the stripped-down working environment offered by *Orion*. *Reason*, *FL Studio*, *Project5* or *Orion Platinum*? A tough call, but better to have that choice than not! SOS



The Song Playlist makes pattern arrangement very straightforward.

information

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- E info@dark-horse.biz
- W www.dark-horse.biz
- W www.synapse-audio.com

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Demeter RV1

Dual Spring Reverb

Paul White

A high-spec spring reverb for the digital age.

While most of today's reverb comes from digital processors rather than echo rooms, plates or springs, there's no denying that the old analogue systems have a character that digital systems find very hard to duplicate. Perhaps the magical warmth is down to the infinitely dense reverb field created by physical systems — most digital reverb units start by generating individual reflections. For example, if a spherical sound wavefront hits a flat wall, you don't just get one reflection, but an infinite series of them as the various parts of the curved front reach the wall. These scatter off in all directions and this reflected energy behaves similarly when it encounters other surfaces, building up into a complex sound energy field where the concept of individual repeats or reflections becomes irrelevant. The word 'homogenous' perhaps best describes natural reverb, and while plates and springs aren't direct counterparts to what happens in nature, they do share that homogenous characteristic.

Plates are large, expensive to build and have to be isolated from noise, which means they're less than ideally suited to project studio use. Springs are less costly and far more compact, but they've attracted a lot of bad press over the years because they are often noisy and have a tendency to sound fluttery. While springs work well with electric guitar, relatively few serious studio spring systems were ever built. Now James Demeter believes he's cracked the most serious technical problems and, in the process, designed a good-sounding spring reverb unit that deserves a place in the modern studio.

RV1 Real Reverb

Housed in a 1U rack, the Demeter RV1 Real Reverb is based around two long Accutronics spring 'tanks', each of which contains three springs (actually six strung together to form three long springs). The receiver end of each tank is shielded with mu-metal foil to reduce interference from magnetic fields, though it is still important to mount the unit away from strong magnetic fields to ensure hum-free operation, as the coils of the pickup transducers are quite sensitive. One spring tank has a 1.5-second decay time, while the other has a 3.5-second decay time.

Part of the problem with spring systems is getting anything like a flat frequency response out of them, but the circuit design used here helps the transducers operate more consistently over the frequency range, both at the drive and pickup ends of the spring. The audio ins and outs to the unit are also fully balanced (on both jacks and XLRs) using Analog Devices 2142 and 2143 amplifier chips, while the power supply operates at 18V to provide plenty of headroom. A toroidal transformer is used in the PSU circuit along with a couple of standard regulator chips, and all the circuitry (which is chip-based) is mounted on a narrow PCB at the front of the unit. The cables leading to the rear panel are clipped down to prevent contact with the spring tanks and some kind of mastic is used

Demeter RV1 £699

pros

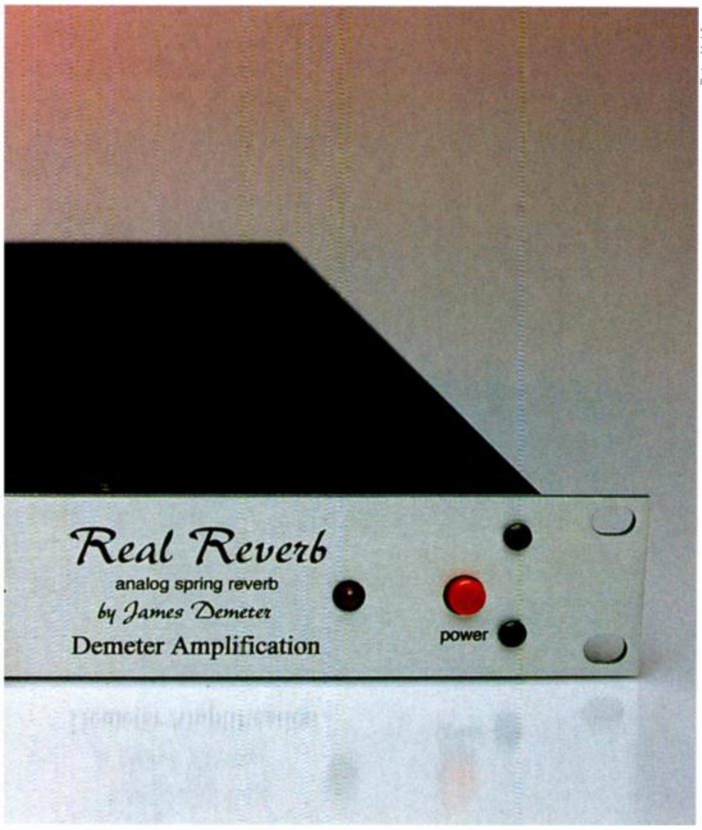
- Low noise and minimal flutter.
- Can be used in stereo or dual-mono configurations.

cons

- The decay time is fixed.
- In stereo mode the left and right decay times are different.
- Not cheap!

summary

This is one of the best-sounding spring units I've tried and should appeal to anyone interested in a more retro sound. However, it seems expensive when compared with multi-functional digital units.



Photos: Mark Ewing

where the tanks sit in their suspension cradles to prevent the suspension springs from becoming dislodged.

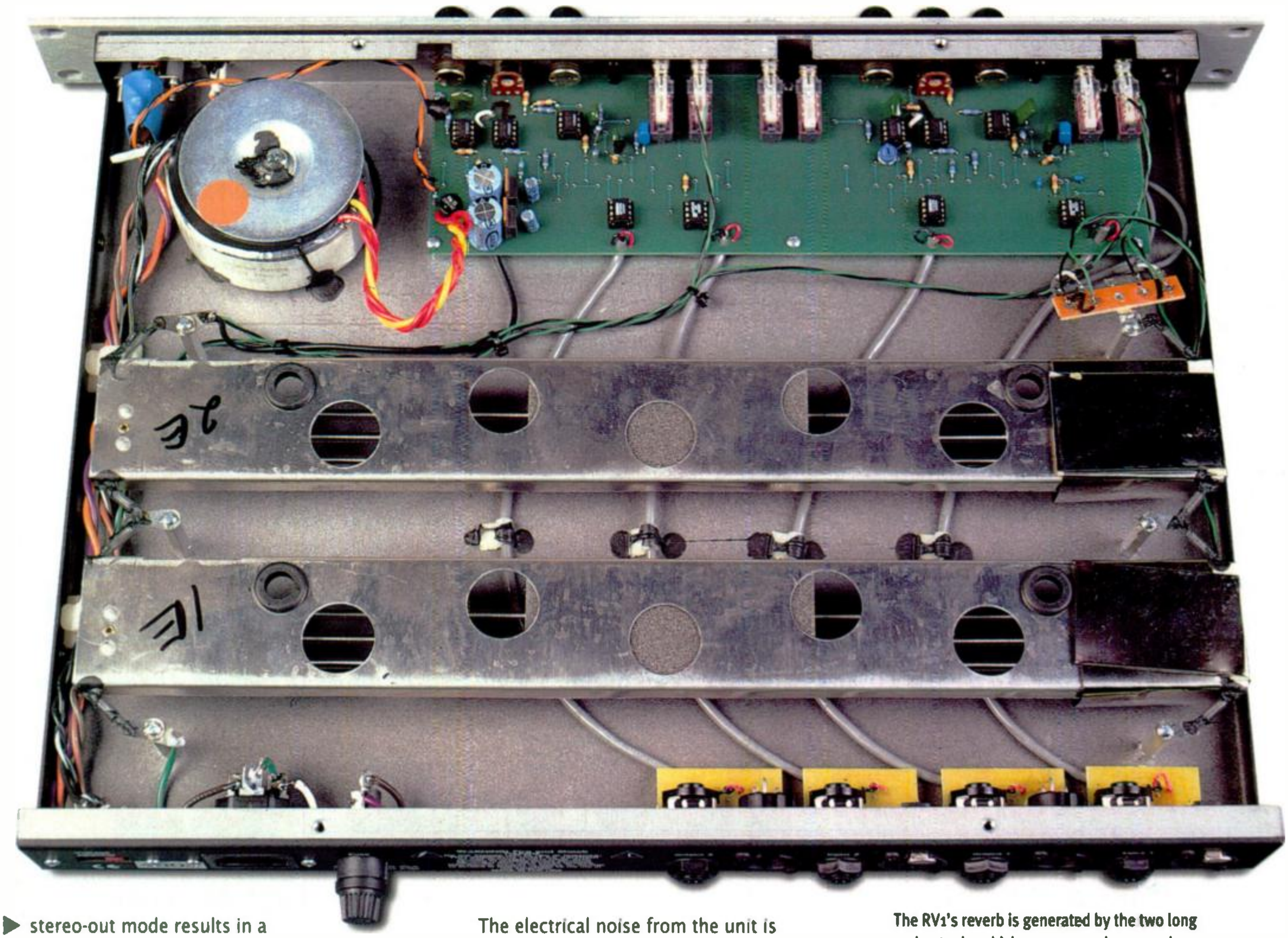
Each spring has its own set of controls and can be used independently as a mono-in/mono-out reverb processor, but there are also separate link switches for the inputs and outputs so that the unit can work as mono-in/stereo-out or mono-in/summed mono-out. Two of the three rotary controls are to do with level, the Input control being used to set the optimum audio level feeding the springs. An overload indicator turns red if the input gain is too high. The other level-related control is Output, which simply sets the level feeding the next device in line. The Mix knob balances the relative levels of reverb and dry signal, though this would normally be left set fully clockwise when used in an effects send loop.

In addition to these three simple rotary controls are two switches, one for phase and one to introduce a low-cut filter into the reverb path, the latter to remove muddying low frequencies when treating sources that have a lot of low end. Phase can be used to put one reverb spring tank out of phase with the other, which provides a slightly different tonality, and because the outputs from the two springs aren't correlated in any significant way, there's no obvious comb filtering or heavy cancellation of specific frequencies. The effect is also slightly different if you flip both phase switches, as the way in which the wet and dry elements combine changes slightly, adding a bright shimmer to the sound.

Testing Time

I tested the Real Reverb using a number of instrument sound sources as well as vocals, and found that, provided that it was mixed at a sensible level relative to the dry sound, it produced a warm and musically satisfying reverb. Because the spring tanks have different decay times, using the unit in ▶

DEMETER RV1



The RV1's reverb is generated by the two long spring tanks which you can see here running across the entire width of the rack unit.

stereo-out mode results in a faster decay time on one side than on the other — although this is not musically unpleasant, it's not really what you expect to hear. Unlike digital reverb units that invariably start off with a series of artificially generated discrete early reflections, springs feature a very fast build-up of density with no noticeable early reflections to speak of. The result isn't quite like a real space, but it is musically sympathetic in the same way that plate reverb is. If you listen to the reverb signal in isolation, it sounds somewhat less impressive, and although the flutter element that's common to all spring reverbs has been tamed pretty well, it's still there if you feed in a percussive sound of any kind. Nevertheless, once the dry signal is added, the imperfections seem to dissolve, leaving a warm, smooth reverb decay. You can even use the Real Reverb with drums, though you have to keep the reverb level fairly low to avoid giving the game away.

The electrical noise from the unit is surprisingly low from a spring device, provided that you set the input gain as high as you can without getting the overload light to flash, and though you do get the familiar 'thwaaang' noise if you knock the unit, it's well enough behaved once mounted in a solid rack.

Interestingly, the Real Reverb works quite well in conjunction with simpler reverb plug-ins, most of which are sadly lacking in density when used on their own. If you feed the reverb signal out through the Real Reverb, the missing density is replaced, but you still retain some of the character of the early reflections. If you want a longer reverb time, you can use the digital reverb unit to create the length you need, then use the Demeter Real Reverb to add density and mellowness.

Puts A Spring In Your Step?

Given that good digital reverb units can be bought relatively cheaply, the Real Reverb may seem quite expensive in the UK, but it does offer an alternative reverberation character that some people may prefer in specific situations, not least for electric guitar or vocals. It will be of particular interest to anyone seeking to create an authentic retro sound, and though there are cheaper spring units, this is the best-behaved one I've tried to date. Given that a good digital reverb unit will satisfy most of the people most of the time, I can't put my hand on my heart and say the Demeter Real Reverb is a 'must have' effects unit, especially given the inflexible nature of springs insomuch as their decay times are fixed, but it does offer a musically attractive alternative that works well on its own or in conjunction with digital reverb. SOS

information

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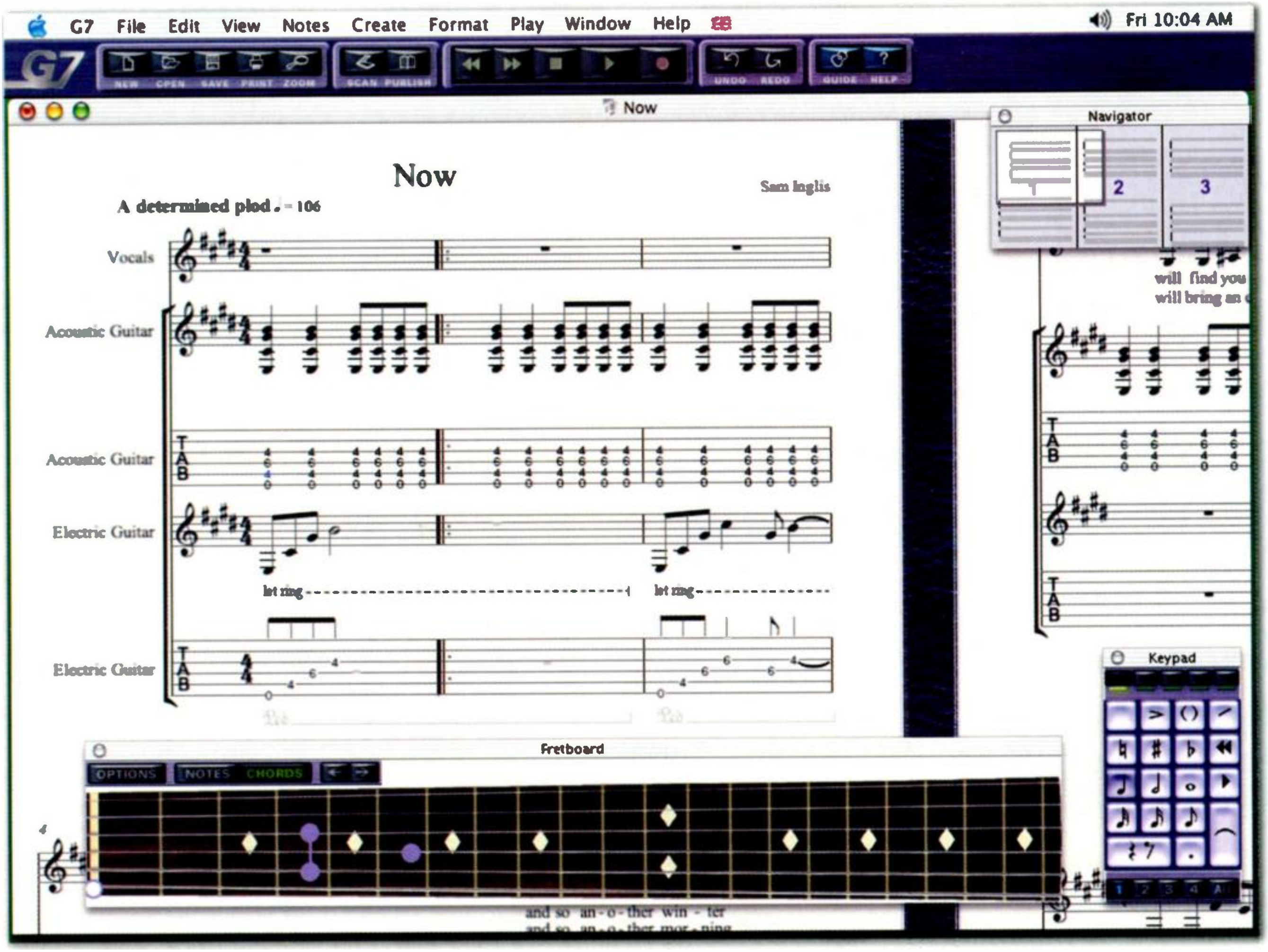


Sibelius G7

Mac & PC Notation Software For Guitarists

Sibelius Software have won over the classical music community with their well-regarded score-writing software. Can their new *G7* package do the same for guitarists?

Sam Inglis

Although there are other worthy contenders in the market, few would dispute that Sibelius Software's eponymous score-writing package is the most advanced product of its kind. For a long time, it was available only for the Acorn RISC OS operating system, but has survived the transfer to Windows and Macintosh platforms with its reputation and user base intact. Version 2, released last year and reviewed in March 2002 (read the review on-line at www.sospubs.co.uk/sos/mar02/articles/sibelius2.asp), added some ambitious auto-arrangement and Internet publishing features, along with support for Mac OS X, and remains the notation package of choice for most computer-based composers and arrangers.

All instruments make their own particular demands of a notation package, and the guitar family throws up more challenges than most. As well as being one of the few genuinely polyphonic instruments, the guitar also has a wealth of different playing styles and expressive techniques such as string bends, hammer-ons, slides and harmonics. Guitar-based music thus requires a battery of special notation conventions, and is often written as tablature rather than on standard five-line staves.

The full version of *Sibelius* has always featured comprehensive facilities for notating guitar music in both conventional and tabbed form, but was primarily aimed at composers who are used to working on a piano or with pencil and paper. Recognising that not everyone composes at a keyboard, Sibelius have adapted their flagship program to create *G7*, a notation package targeted exclusively at guitarists.

In essence, *G7* consists of a cut-down version of *Sibelius*, augmented with some new features designed to simplify and streamline score-writing for guitar players. Most prominent of these features is a virtual guitar fingerboard: as in *Sibelius* itself, you can still enter note data using a MIDI or QWERTY keyboard, but *G7* also allows you to do so by clicking on on-screen frets. Also exclusive to *G7* is the Guitar Guide, an Acrobat-based educational resource providing information on different kinds of guitar, playing techniques, and so on (see box elsewhere in this review). The bundle is topped off with Neuratron's *Photoscore Lite*, which allows you to scan sheet music and turn it into a form *G7* can understand, and Sibelius's *Scorch* web browser plug-in.

Getting Started

The full version of *Sibelius* is justly renowned for its 500-page printed manual, and *G7*'s 200-page effort is also very welcome. It covers all of the program's features, it's clearly written and well illustrated, and someone has actually proof-read it, all of which makes a pleasant contrast with some software packages.

Installing *G7* is straightforward. The first time you load it, you're prompted to enter your serial number, and you also need to register the program within five days; if your music computer is connected to the Internet, *G7* will link to Sibelius' web site itself, but you can also register by 'phone. Past the splash screen and a burst of ear-splitting music (which can, thankfully, be disabled in the preferences), and it's onto the Quickstart dialogue, where you can decide whether you want to open a recent score, start a new one, and so on.

If you choose to start a new score, you're given a choice of templates to choose from. These range from the obviously useful (electric guitar tab, classical guitar notation, guitar duos, acoustic guitar plus vocal, and so on) to the more esoteric (mariachi band, anyone?), and you can add or remove instruments from them to suit your needs. The Add Instruments dialogue reveals the most important sense in which *G7* has been cut down compared to the full *Sibelius*: it provides a comprehensive selection of guitars, basses and other fretted instruments, along with a fairly wide range of percussion instruments, but most of the orchestral instruments are missing, and there's no facility to define your own instruments. You can have lead and backing vocals and a few different types of keyboard, plus 'strings' and 'brass', but there are no special facilities for notating transposing instruments, and no tenor or alto clefs. I suspect that anyone wishing to produce serious brass arrangements will find reason to buy the full version of *Sibelius*.

You can have up to 16 staves in one *G7* score, and it's worth noting that all the staves in a score are always independent of one another, even if two of them are supposed to represent the same instrument. For instance, if you want to show a guitar part in both tab and conventional notation, which is often useful, there's no way of linking the two staves so that they automatically mirror one another — you need to manually copy and paste between them to make sure that the notes are the same in each.

Using *G7*

Sibelius is one of those programs that becomes second nature after a time, and power users can enter complex scores very quickly. However, I think it's fair to say that if you're used to typical word-processing, graphics and DTP packages, you might find that you need to un-learn some of their conventions in order to get used to *G7*'s way of working. For instance, you can use the cursor left and right keys to select the object to the left or right of the current cursor position, but the up and down keys move the currently selected object up and down the staves, rather than selecting the object above or below it. If you have a note or rest type selected in the floating Keypad palette, positioning the mouse pointer over a stave and clicking deposits that type of note; to use the mouse pointer to make selections or scroll the page, you need to press Esc to clear the note palette. It's easy to get confused until you become familiar, and fortunately, *G7* offers up to 10,000 levels of Undo (the exact number is set as a preference). Unusually, and happily, making selections with the mouse pointer or arrow keys is an Undoable action.

Using the mouse to position notes on staves is only one way of entering data into *G7*, and for most purposes the others are more useful. If you have an attached MIDI device such as a keyboard, you can play in your parts in real time, and if you have Flexi-Time switched on, *G7* will intelligently interpret any timing variations in your performance. This works well enough with a MIDI keyboard, especially if you play the part in slowly. However, I was disappointed to find that there are no special facilities for cleaning up MIDI guitar input, even though the program is targeted at guitarists. Unless you use some third-party MIDI filter utility between the guitar and *G7*, *G7* will faithfully notate all the spurious low-velocity notes that are inevitably generated when you play a MIDI guitar.

If you don't have a suitable MIDI input device to hand, the most efficient way to enter note data into *G7* is probably to use the computer keyboard. The Keypad palette, as in all data-entry modes, is used to select the lengths of each note and rest, along with other features such as accidentals, slurs or ties, slides and so forth. As you can see from the screen shots, each key on the Keypad corresponds to a key on your computer's numeric keypad, with the zero key toggling between notes and rests. (The Keypad is a fundamental part of the

Importing Tab

Web sites such as OLGA (the On-Line Guitar tab Archive) and the Cowpie Song Corral contain hundreds of thousands of songs transcribed into tab format by fans and guitarists. Sibelius have realised that these tab files provide an invaluable resource for guitarists who need to learn a song without having to transcribe it themselves from CD, and so they've built into *G7* the ability to open ASCII tab files as well as Standard MIDI files. The process is inevitably a little hit-or-miss, as there's a lot of variation between the ASCII files produced by different guitarists; *G7* will do its best to cope with the variations and ignore spurious text, and Sibelius say that it will usually get the notes and chords right, but not the rhythms. This is fair enough, as few ASCII tab files include detailed rhythm information in any case. Sibelius suggest that you might benefit from cleaning up ASCII files in a text editor before attempting to import them, but even so, I had limited success with a random selection of files downloaded from OLGA — of half a dozen I tried, only one would actually open at all in *G7*.

SOUND ON SOUND

Sibelius G7 £149

pros
- Contains all of *Sibelius*'s excellent features for notating guitar music.
- Supports almost every conceivable instrument within the guitar family.
- Decent, printed manual.
- Can import ASCII tab (sometimes) and publish on the Internet.

cons
- Guitar Guide and auto-arrangement features are superficial.
- Fretboard can be frustrating to use, and is not very well integrated into the program.
- No dedicated MIDI guitar features.
- Doesn't handle capos very well.

summary
At heart, *G7* is a streamlined version of *Sibelius*, and that means it's a powerful and flexible package capable of producing professional-quality scores. However, its new features don't entirely succeed in making this power more accessible to guitarists.

program, and I wouldn't like to use *G7* on a laptop without a separate numeric keypad.)

When you have a note length selected, pressing any of the letters 'A' to 'G' creates a new note. Pressing a number key on the main part of the keyboard adds an additional note to the currently selected note or chord, separated by whatever interval that number represents. So, for instance, pressing 'A' then 'C' would create two separate notes in sequence, but pressing 'A' then '3' would create a single two-note chord. Once you get used to this method of entering note data, it feels very efficient, and you rarely need to use the mouse except to revisit selections for editing, or to add special symbols such as pauses.

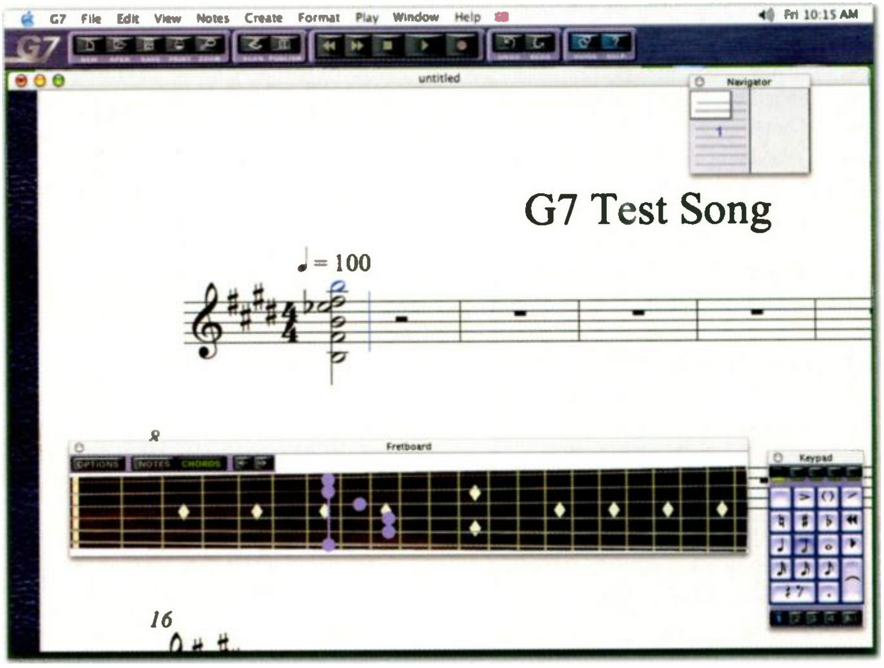
Something I did find counter-intuitive in all the data-entry modes is that there's no way to delete a note or rest and have everything to the right of it shift along to close up the gap. Deleting or cutting a note always inserts a rest of the same length, while attempting to delete a rest just makes it go grey, indicating that it has been 'hidden' and won't print out. This has the advantage that making corrections early in a piece doesn't constantly cause the rest of it to be shifted around, but can be a pain if you discover you've accidentally entered too many notes and need to remove one.

The Fretboard

Flexi-Time, mouse and QWERTY note entry will already be familiar to existing *Sibelius* users, but in *G7* the Keypad is joined by another floating window: the Fretboard. As you might expect, this is a graphical representation of a (right-handed) guitar or bass fretboard, on which you can click with the mouse to tell *G7* what notes you want to enter. The on-screen fretboard follows whatever tuning you've chosen in the currently selected stave. You can select a variety of appearances, and you get the choice of four-or five-string basses as well as guitars; clicking on a staff of bass guitar tab in your score automatically changes the Fretboard to show the bass neck, but for some reason, this doesn't happen when you click on a bass guitar staff that uses conventional musical notation. Guitarists are limited to a six-string neck, and if you're entering notes for an instrument with more than six strings or courses, such as a seven-string guitar or a lute, the on-screen fretboard corresponds to the six lowest strings on that instrument.

Test Spec

- *G7* v1.0.
- 800MHz Apple G4 iMac with 256MB RAM, running Mac OS 10.2.



The song is in E major, and I've entered a simple B major chord on the Fretboard... so why has *G7* notated the D sharp as an E flat?

A more annoying restriction is that there's no way of adding a virtual capo to the Fretboard. For pieces where the capo is used across the whole neck, it's not difficult to enter the notes in the first position and simply add a text label telling the player where to put the capo. However, it's not unknown for guitarists to use a capo over, say, the top five strings only, and there's no way to adapt *G7*'s notation or tabbing to reflect this. It would be similarly difficult to notate properly for a four-string bass guitar that offers a low D beyond the normal nut.

The Fretboard has two modes of note entry: Notes and Chords. If Notes is selected, clicking on the Fretboard creates a single note and advances the cursor to the next position on the stave, so that you can enter another note. In Chords mode, by contrast, each click adds a new note at the same cursor position, thus building up a chord, and you need to click the right or left arrow buttons to move on to the next or previous chord.

The Fretboard looks neat, but in practice I found it rather frustrating to use, for a number of reasons. Almost every aspect of *G7* can be controlled by an elegantly constructed system of keyboard shortcuts, but you can only switch between Notes and Chords modes by clicking on the relevant buttons with the mouse. Although you can use the 'R' shortcut to repeat the last note or chord, there's no library of Fretboard chords or chord shapes, so every time you enter a new E-shaped barre chord, you need to click six times on the Fretboard. Worst of all, if you're entering notes and chords onto a five-line stave (rather than a tabbed one) from the Fretboard, *G7* seems to have difficulty understanding fret positions in the context of the key of your song. For instance, if your song is in E major and you enter a simple B major chord on the Fretboard, *G7* wrongly notates the D sharp as an E flat. Things got worse when I experimented with a song in A flat major (four flats): *G7* soon became confused and started to notate every note in the A flat major triad enharmonically (as G sharp, B sharp and D sharp). If this is artificial intelligence, it's not exactly the musical equivalent of HAL 9000.

A Symphony Of Voices

One of the main differences between tablature and conventional notation is that while tab can indicate the rhythm in which all the notes are to be struck, it can't show how long each note should be sustained. In classical notation, by contrast, each note has a definite length, so it's possible to indicate both the playing rhythm and exactly how long each note should be held for. When you come to notate almost any fingerstyle or classical guitar piece in this way, it's necessary to show some notes being struck while previous notes on other strings are still ringing. In a simple fingerstyle folk arrangement, for example, you might want to have each bass note sustained through a whole bar of finger-picked arpeggios on the higher strings.

This is where notating guitar music gets

SIBELIUS G7

This classical piece requires three *G7* polyphonic voices to notate. When selected, voice 1 appears blue, voice 2 green, and voice 3 orange.

a bit more complicated, because you need to start thinking of it in terms of polyphonic voices. In the example above, you would need at least to treat the bass notes and the arpeggiated picking as separate voices on the same stave. Unless you play in your parts from a MIDI guitar or keyboard, you have to do the work of breaking your guitar parts up into polyphonic voices as you enter them. Four buttons at the bottom of the floating Keypad are used to select and indicate which voice you're currently writing to, and the cursor and any selected notes change colour to reflect this. Voices one and three are represented by default with 'sticks up', voices two and four 'sticks down'. It's straightforward enough once you get used to it, but you will need to get used to it if you want to work with classical notation as well as tab.

Since the guitar has six strings, it would perhaps be technically possible to compose something requiring six separate voices of polyphony, but as the manual says, the four voices per staff offered by *G7* should be adequate to notate any existing classical guitar piece. Because of the way *G7* defines a voice, however, playing in a guitar piece from a MIDI guitar or keyboard will tend to produce something that needs quite a lot of tidying up.

Take, for instance, a simple picked chord over six strings, where each note is played a quaver apart and sustained. This isn't polyphony in any musically interesting sense, but although each individual voice in *G7* can contain chords, every note in the chord must have the same length and must start at the same point. If you want to have one note begin or end while an earlier note is sounding, they must be assigned to different voices, so the most natural way of notating the spread chord described above would require six *G7* polyphonic voices. As it is, the user is forced to compromise. The conventional solution is to notate every note as a quaver, and perhaps add an instruction such as 'Let Ring'; this looks all right and is easy enough for a human player to interpret, though if you want *G7* to produce sustained notes when playing back your score or exporting a MIDI file, you'll need to add hidden sustain pedal messages. When you play in or import a MIDI part containing a spread or arpeggiated chord, however, *G7* takes the alternative route, which is to represent the sustains using tied notes. This plays back correctly, but usually looks horrible — particularly as *G7*'s algorithm seems designed to use as few voices and as many tie-lines as possible.

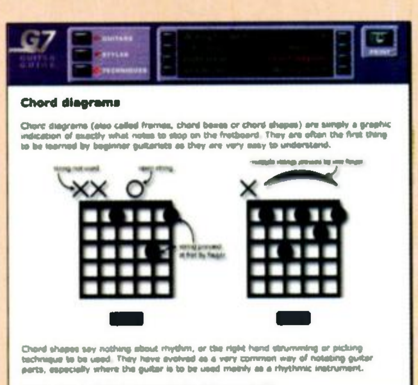
This may well be the only practical approach from a programming point of view, but it does mean that anything you play in to *G7* will need a fair bit of work before it looks like a proper score. It would have been nice to have a tool for thinning

Guitar Guide

One of the *G7* features Sibelius have been most keen to trumpet is its Guitar Guide. Clicking on a small guitar-shaped button in *G7*'s toolbar launches Adobe's *Acrobat Reader* and brings up the front page of the Guide. The Guitar Guide is divided into three sections: Guitars takes you through the different varieties of electric and acoustic instruments that exist, Styles covers eight different musical genres in which the guitar is prominent, and Techniques describes playing techniques and notation conventions. It's navigated by clicking buttons on a virtual rackmounting front panel at the top of the window, and each page contains concisely worded text, a few drawings or score examples, and short audio examples accessed through Play and Stop buttons.

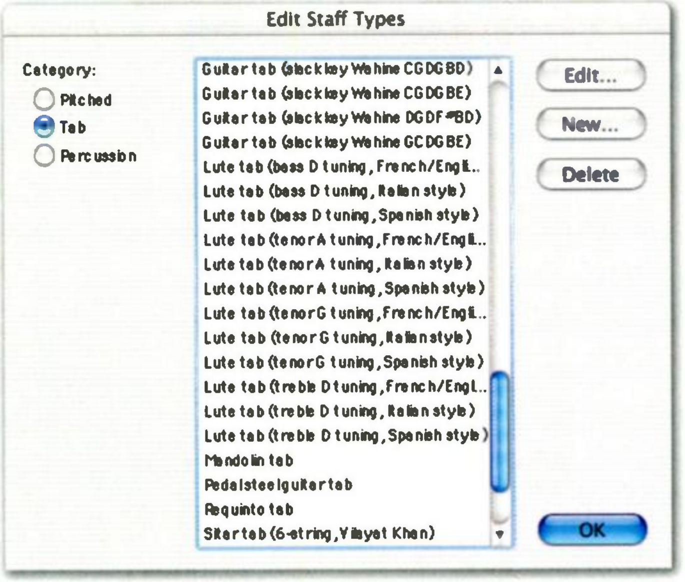
The Guide is clearly written, but I'm slightly baffled by its inclusion in *G7*, as much of the information in it is very basic. I can't imagine many potential *G7* users really needing a series of audio examples to explain what the open strings sound like on different guitars, for instance — especially when the steel-strung acoustic has been recorded with a bad piezo pickup and sounds nothing like an acoustic guitar. With its large-print text and cartoon drawings, the Guide feels more like a resource for schoolchildren and beginner guitarists than an accompaniment to a sophisticated score-writing package.

My guess is that *G7* itself will appeal most to experienced guitarists who want to develop their ability to produce professional-looking sheet music, and as such, I feel that it would have been more useful to have a detailed guide to notating for guitar. The manual does a good job of explaining how to use the program's functions, and how to access advanced symbols such as bends and harmonics, but the two included tutorials are quite brief. I would have welcomed more advice on topics such as how to represent a fingerpicked guitar part using polyphonic voices, which is something most users will need to understand in order to use *G7*.



out those bunches of tied notes and adding appropriate MIDI sustain information instead. If this was the full version of *Sibelius*, with its ManuScript programming language, an enthusiastic user could perhaps create a plug-in to do so, but *Sibelius*'s plug-in support has gone by the wayside in *G7*.

Scoring polyphonic pieces is another area where a proper MIDI guitar implementation would have been very welcome. Most MIDI guitars can output notes from each string on different channels, so why not have the option to force notes from different strings to separate *G7* voices, or at least to write notes from each string to the correct line in a tabbed staff?



G7 supports an impressive array of guitar tunings; if yours isn't one of the many presets, you can also create or edit another.

Playback Time

Although *G7* is by no means a fully specified MIDI sequencer, it can be used to play back scores on Quicktime Musical Instruments or any attached MIDI instrument, and will intelligently interpret and follow tempo markings, dynamics, articulations and so forth. The more instructions you give it, the better the results are likely to be, and you can easily hide any that you don't want to appear on your printed output. If your score originated as a performance on MIDI keyboard or guitar, you can use the Live Playback option to play it back exactly as it was performed. If, on the other hand, you've created your score by entering notes or tab onto staves, you can choose from a variety of performance styles which aim to impart a degree of human feel. The results are unlikely to be mistaken for John Williams, but can be surprisingly effective.

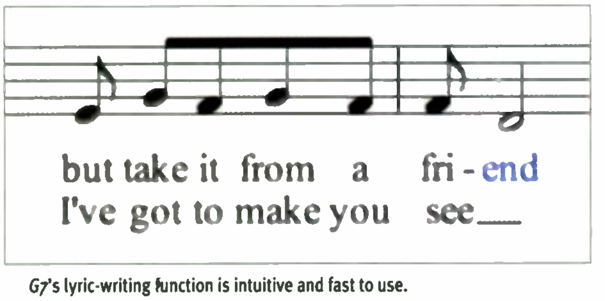
Nevertheless, there are a couple of ways in which *G7*'s playback could be improved. First, all editing is suspended while *G7* is in playback mode, even if it's paused; I would have liked to be able to make selections, at least. Second, *G7* doesn't interpret capo markings on playback. Third, the program uses a single MIDI channel to play back all the polyphonic parts for an instrument, and seems to use MIDI pitch-bend to recreate slides and string-bends. The result is that slides and bends get imposed on all notes that are sounding, even if they're only notated on one string. It would have been nice to have an option to play back each polyphonic part or each string on a separate MIDI channel, and to be able to export MIDI files in this format.

G7's implementation of polyphony also undermines another of the Fretboard's features. When you play back a score, dots appear to show you where to put your fingers at each step. However, the Fretboard is capable of displaying only one polyphonic voice at once, which makes it pretty useless as a learning aid for complex pieces.

Lyrics And Chords

Songwriters will welcome the fact that *G7* has inherited *Sibelius*'s impressive lyric-writing features. Up to five lines of lyrics can be added to a stave, and the program does an excellent job of aligning words with the notes in the melody. All the user has to do is remember to insert hyphens between syllables when one word is broken over several notes; the rest is handled automatically and with consummate ease, although I did once encounter a curious glitch where every time I tried to edit a word, it reverted to the original.

Sibelius's excellent handling of box-type guitar chords has also survived intact into *G7*. Individual box chord diagrams are added to the score manually from the Chord Diagram dialogue. This includes an extensive chord library covering every conceivable guitar tuning (except those involving capos), which you can augment with your own shapes if necessary, and includes intelligent search facilities — so you can, for example, view only C minor chords with an E flat bass note and a finger spread of two frets or fewer. Just as in *Sibelius*, this is a very powerful feature, but it's a shame that it hasn't been integrated at all with the new Fretboard window: it would have been very handy to be able to access the libraries of chord diagrams when inputting chords from the Fretboard, or to have chords entered via the Fretboard appear automatically in the Chord Diagram library for your song.



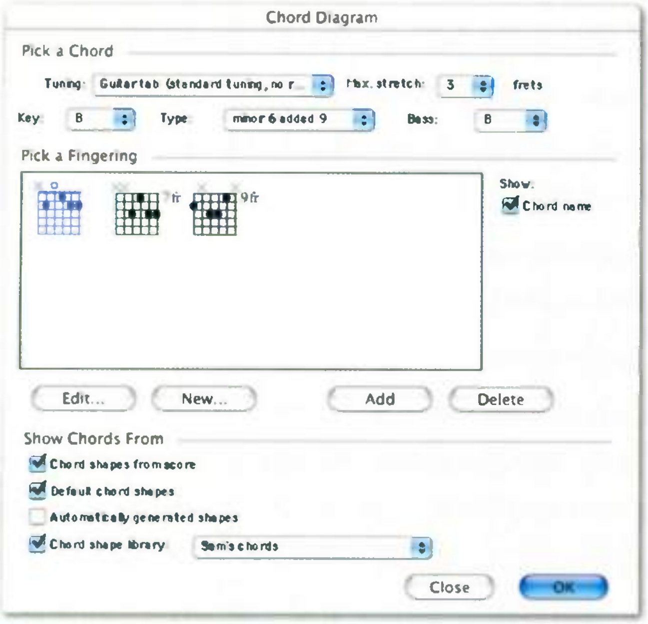
G7's lyric-writing function is intuitive and fast to use.

Automatic Writing

Another *Sibelius* feature that's present and correct is the facility to select a staff of tab or notation (for any instrument, not just a guitar) and have *G7* automatically add chord names. This works very well, and for some reason seems much cleverer than the Fretboard at understanding notes in the context of the key of your song. You can also select a staff of five-line notation and paste it into a tab staff, whereupon *G7* will make a decent stab at working out the easiest fingering. Sadly, though, there's no facility to turn your tabbed chords into box diagrams, and the chord names feature only works on block chords, not arpeggiations.

Additionally, unlike *Sibelius*, *G7* boasts a reverse chord-naming feature. You feed it a series of chord names or symbols, and it will generate a simple guitar part, either in tab or classical notation, or a piano part. This is all right as far as it goes, but doesn't go very far: you're basically limited to producing block chords or simple arpeggios.

G7's new user-friendly arrangement tools also include a drum pattern generator. This is certainly easy to use: all you do is select an appropriate style from a drop-down list, decide whether you want an intro and/or an outro bar, and tell *G7* how often you want it to insert fills. After mulling for a while, *G7* will then produce a drum part on a new staff. Again, this works without a hitch, but is pretty unsophisticated. Each 'style' basically consists of nothing more than a simple repeating drum pattern — there doesn't seem to be any attempt made to follow either the structure of your song, or the



The Chord Diagram dialogue allows you to choose from your own libraries of chords, or an array of automatically generated shapes.

rhythm of your guitar or bass parts. I suspect that most people will find this feature useful only as a very basic starting point for creating drum parts.

Conclusions

Having spent a fair amount of time with *G7*, I find myself with slightly conflicting opinions about it. The many features that it has inherited intact from the full version of *Sibelius* are undeniably impressive, from the comprehensive Chord Diagram dialogue to the massive array of special notation symbols and the beautifully thought-out system of keyboard shortcuts. Considered as a cut-down, more affordable version of *Sibelius*, it will enable many musically literate songwriters, transcribers and bandleaders to do everything they need to, without forking out extra cash for advanced orchestration features they won't require. If you already know what you're doing, you can easily produce scores to the same standards as you might find in guitar magazines or songbooks, while the built-in on-line publishing facility provides an interesting alternative means of getting your music to the masses.

However, if the aim is to open up score-writing to a new group of musicians — people whose primary instrument is the guitar, and who might not be experts in music theory or the conventions of classical notation — I feel that some opportunities have been missed. Neither the Guitar Guide nor the new auto-arrangement functions offers enough depth to be genuinely useful. The Fretboard is a nice idea, but gives the impression of having been 'bolted on' to the core program, rather than integrated effectively with the existing features. Little has been done to help the novice understand the process of representing guitar parts using polyphonic voices, and the limited capo handling is an odd restriction in a program that copes with so many guitar-related instruments and tunings. Finally, and perhaps most unfortunately, *G7* could have been the perfect application for use with a MIDI guitar, but this potential has been wasted.

At £149, *G7* represents excellent value compared to the full version of *Sibelius*, but it's not without competition: Sibelius's rivals Coda Music offer an even more aggressively priced guitar-oriented version of their *Finale* score-writing software, which retails for only £80. This, however, lacks *G7*'s Mac OS X version, ASCII tab import, Fretboard input and Guitar Guide — and, of course, isn't based on *Sibelius*.

In short, guitarists who already know that they need a notation package will welcome *G7* with open arms. The crucial question is whether it can convince the other 99 percent of us. *G7*'s powerful feature set, good manual and slick appearance are bound to help here, but there are areas in which it doesn't fulfil its potential. SOS

Information

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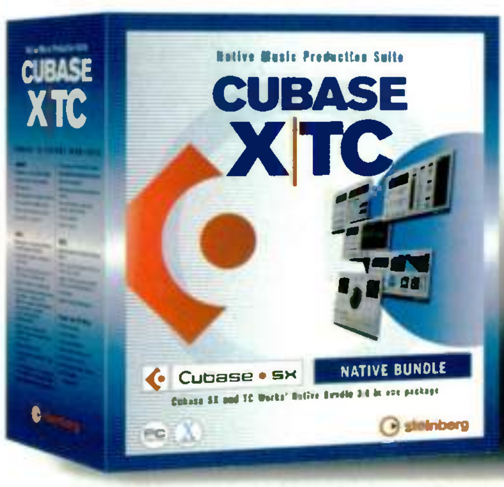

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Will Gregory and Alison Goldfrapp.

Goldfrapp

Will Gregory: Recording *Black Cherry*

Nigel Humberstone

It seems that almost everyone who's heard Goldfrapp's debut album *Felt Mountain* has fallen under their spell. Now the duo have returned with a second, *Black Cherry*, and a harder, synth-based sound.

Taking inspiration from classical music, movie soundtracks, '60s French pop and decadent Weimar Republic cabaret, Alison Goldfrapp and Will Gregory, as Goldfrapp, fashioned a mesmerising debut album *Felt Mountain* in 2000. Pooling together their eclectic visions, the album was composed using a range of instrumentation, including cello, violin, viola, double bass, French horn, baritone ukulele and melodica, all intertwined with edgy leftfield electronica — not forgetting a certain amount of whistling, at which Alison is particularly adept, and which helped produce one of those unforgettable melodies utilised in the equally memorable Orange TV commercial featuring Gary Oldman. It became Goldfrapp's calling card.

Will Gregory, the man behind Goldfrapp's unique sound, is in many ways a musician with split personalities. With a background steeped in music composition for TV and films (*ID*, *Rabbit-proof Fence*), he is a string arranger and saxophonist of merit: but juxtaposed with all this is an underlying passion for all types of electronica. Close friend Adrian Utley of Portishead started him off on the vintage synth-collecting bug, which hasn't diminished. The difference between the two collectors is that Utley knows when to stop!

"We had a bit of a race at one time, but he's more sensible than me and gets rid of stuff," concedes Gregory. "But now I've got a room which is stacked full to the ceiling with all this stuff, which is a bit alarming.

"The way that I look at it is that if you get a synth and get a song from it, or it inspires something or gives character to something, then it's valid and justifies itself. A lot of them may only make one or two sounds that I really love, but for that reason I can't bring myself to get rid of them."

These characteristic instruments are undoubtedly the driving force behind Goldfrapp's latest album *Black Cherry*. Synths set the tone and define the agenda —

having too many is not an option. "When you making an album it's somehow justified and I walk willingly into that danger," admits Gregory. "I get freaked out when I think about us all sitting down at 10 in the morning in front of *Logic* set at 120bpm, 4/4. I think we all need something that's our own, that you feel is special, and I think finding these old things, probably quite erroneously, makes me feel like that."

Agenda

Black Cherry started taking shape as an album at the beginning of 2002, after Gregory's 'accumulated' home studio had been moved to purpose-built premises in Bath. Based around a Mac G3 and *Logic*, the initial demos and pre-production were controlled through a Yamaha 02R digital desk, used more as a neutral router and automated mixer than for its effects or EQ. For these purposes, Gregory relies on outboard devices like old Audic channels for mic pres, and a valve compressor built by his girlfriend's enthusiast brother. Augmenting these is Gregory's collection of synths, including an ARP 2600, EMS VCS3, Korg 700 and MS20, and Moog Minimoog as well as three Farfisas, Hammond, Casio and Philips Philicorda organs.

Gregory distinctly remembers the start of the album following the 2001 Christmas break: "On 7th January we sat down in a cold room looking across a few synths at each other, contemplating what we would do. When we started *Black Cherry* we made this big list on a huge bit of paper, listing what we wanted to do. Like we wanted to do a disco track with just strings, we wanted to do a track with a lot of white-blouse-wearing dulcimer players. And we didn't actually do any of these things!

"We do love to make little dogmas for ourselves. Like saying 'We're not using any guitars' — but in fact we did use guitar. Adrian [*Utley*] played guitar on 'Train'; he's the fuzz noise that comes in every once in a while. But generally I think with guitars that we find they come with a lot of baggage that just moves things into a recognisable zone that we can't quite handle.

"We also said 'We're not using any samples.' We don't use sampled loops anyway — they come with a little bit of borrowed vibe, but you pay that back later on because you're not completely in control of the character of what you're doing. And we don't use bits of records — we've figured that it's more entertaining to go and make your own loops."

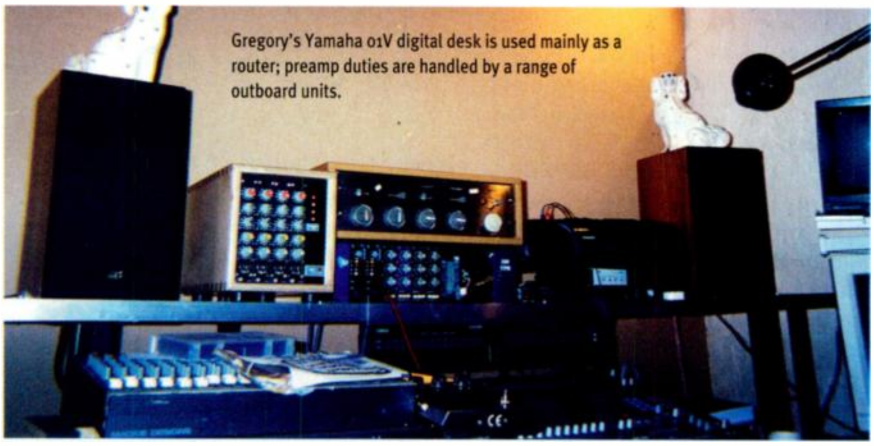
These organic loops and textures, which formed the basis for the majority of the new tracks, were created from improvised jamming sessions involving Gregory, Alison Goldfrapp, Mark Linkous from Sparklehorse, and Adrian Utley.

"Basically we did a 10-minute jam, and two weeks later that was 'Train'. That was one of the jamming days that we had," recollects Gregory, "where there were four of us twiddling knobs or guitars or whatever we were doing, all at once — which made the editing side of it rather complicated.

"I think we did more jamming on synths with this album. On *Felt Mountain* we would tend to set up pads with a few other sounds and improvise over that. This time we'd set up a basic beat for a couple of them and improvise with the synths wildly over it, and then go back to find things we liked rather than go for a purposeful melody over a chord. There was improvisation on *Felt Mountain*, but this time it was improvising at a much more basic level — just making noises, really, or events or accidents or things that we liked the sound of that we could loop up. So there was a more fundamental approach, building from sound events. But apart from that I think we just wanted to turn everything up and get a bit more raucous and dirty — have a bit more fun."

Commercial Interest

A number of tracks from *Felt Mountain* have been used for commercials and in films, and Goldfrapp were even commissioned to write music for a new commercial (the Renault Clio 'Va Va Voom' campaign). So were they conscious when making it that tracks from



Gregory's Yamaha 01V digital desk is used mainly as a router; preamp duties are handled by a range of outboard units.

Black Cherry might be used in that context? "No," insists Gregory. "It's a weird thing that when you're writing music you get a phone call saying they want to do a mobile phone ad using your music... The other side was that we weren't getting any radio play for *Felt Mountain*, so we thought it might be a good way for people to get to hear the music. So we did end up licensing one or two things, for that reason more than anything else."

This, according to Gregory, did not influence the way in which *Black Cherry* was conceived as an album. "I think we followed our intuition more this time. We needed to write something that wasn't like *Felt Mountain*, so that was a mission, and we had some ideas about the vocals. Alison's very versatile and has a lot of other voices in her than we got out on *Felt Mountain* — so we really wanted to explore how we could develop that. We wanted to do vocal harmonies, use layers and different vocal effects. For example, on 'Train' all the harmony is in the vocals, there's no harmony in the track. So we kind of slipped ▶

The String Orchestra

Despite *Black Cherry*'s heavy reliance on analogue sounds, a number of tracks, including 'Black Cherry', 'Tiptoe', 'Forever' and 'Deep Honey', feature live strings recorded with a 27-piece orchestra at AIR Studios by Steve Orchard and orchestrated and conducted by Nick Ingman. As an experienced arranger, Gregory himself had made the initial efforts. "We'd worked them out and Alison listened to them on MIDI strings, and then I'd go away and try to turn it into something that people could play. I had help from Nick Ingman, orchestrator and conductor, just to dot the i's and cross the t's. Then he would conduct the session so that I could be sitting in the studio relaxing — no, listening really hard. Total megalomania time! You know you can keep running in and out saying 'More vibrato!'

"With five titles to record and only four being allowed within any one three-hour session we had to resort to some 'creative' scheduling. But string players nowadays are not like they were in the '60s and '70s when they were slipper-wearing 50-year-olds. Now they're all young, vibey and into pop music — and they know how to play in time as well."

that round a bit."

Whereas *Felt Mountain* had been sporadic in its construction, *Black Cherry* was an intense affair, with Alison Goldfrapp relocating to Bath in order to be involved from the beginning. Lyrics were often pre-written, as she keeps a book of ideas, but they also developed along with the tracks. "A lot of the time she's doing guides or improvising over stuff," explains Gregory, "making sounds rather than words, and then sometimes they crystallise into words. So she writes almost by making a weird translation of the sounds she's made."

Vocals were almost exclusively recorded using a 1970s AKG C12 microphone, routed through an Audic channel and via a MOTU interface into *Logic*. Gregory came across this particular model of microphone via producer David Lord, who had a mic called a Percy Bear: "Essentially it's a mono C12, which really suits Alison's voice, and it's got a really lovely glassy top end. I found one and that's what the album was done on."

Attempting to describe the music on *Black Cherry*, especially tracks like 'Train' and 'Strict Machine', is no simple task — but try to imagine David Bowie's 'Jean Genie' merged with the double drums of the Glitter Band, topped with a little Donna Summer (Giorgio Moroder and Pete Bellotte era) and you're halfway there. It's like a mutant disco stomp propelled by surging analogue bass sounds.

"A lot of the bass lines were put together from chance jams," recalls Gregory, "they're not even melodic lines. But if you loop them they have a rhythmic groove to them that we liked and it's kind of like that 'concrete' principle of Steve Reich, where he was taking loops and music would appear out of them through sheer repetition. You know you could probably dance to the sound of someone coughing if you looped it up — you'd find some element of music in it.

"Last time [*on* Felt Mountain] I was using more orchestral sounds, and this time we went for naked synths a lot more — and organs. For example, at the start of 'Hairy Trees' there's a little Farfisa organ that I've got. It's got this weird knob that says 'Slalom', with a picture of someone skiing down a hill, and basically it gives you pitch-bend, which is quite unusual for an organ. And also you can think about skiing whilst you do it, which is quite entertaining!"

Upfront Drums

The move to more analogue textures and sounds also necessitated a different approach to drums. Rowan Oliver, Goldfrapp's drummer for their live shows, played on some tracks, as did Damon Reece from Spiritualized, but the majority of drum parts were programmed. "I think we quite like and favour electro drums," explains Gregory, "rather than going for a rock-out real kit. It's difficult — we're new to drums and we find that if you go with too much reality it starts to sound like a pub gig. We had this really nice machine called a Machine Drum [*Elektron SPS1*] — a Swedish thing, and we used some of that. It's a lovely analogue thing with knobs everywhere, a bit like a Simmons meets a 909 because you can filter and sweep and edit the synthetic parts so the drums sound really well. We also wanted to get the drums more present — I think we felt we needed to learn how to do that. Whereas before we'd gone for this very background '60s big-band approach to drums, where they're just supporting things in the background, this time we tried to move them to the front and see how that worked. So that was another mission."

Processing is another important part of the Goldfrapp sound. Gregory employs the usual tricks of feeding signals (including Alison's vocals) through old amplifiers as well as via synths such as his VCS3 and ARP 2600. "It processes beautifully so we shoved drums through it, voice, anything really. It's an instant '60s vibe machine because of the spring in it and the filter and the little bits of distortion that you get from it. Very often you can put stuff through it to give you a kind of sepia effect on everything.

"Funnily enough, I've got a Minimoog and I didn't use it. I can't get my head around the Minimoog at the moment. I find it hard to make it sit in the track, but I love my [*Korg*] MS20 a lot, and I've got this weird Russian thing that I found in East Germany called a Polyvox, and that is a bit of a beast. On the front panel everything's in Russian Cyrillic script, so I don't know what anything does. I've just about worked out where the pitch-bend is! But also, things don't do the same thing twice. There's a knob on there that says Ron and one below called 2 Ron, and as far as I can tell, if you touch those everything goes mental. It's out of control but lovely."

The Polyvox is a duophonic analogue synthesizer, with a 49-note keyboard, which was made and sold in Russia during the 1980s by the military. In this instance it added the distinctive, submarine bass sound to 'Black Cherry' and a throbbing, metallic growl to the aggressive, retro-tinged 'Train'. But whilst analogue synths are ever-present on *Black Cherry*, they are carefully, and very cleverly, controlled so that every texture and nuance has its own place in the overall mix.

Will Gregory's much-used EMS VCS3 and Roland SH09 synths, with a Wurlitzer EP200 electric piano.

"It was achieved through taking time," admits Gregory. "It's a very slow process when you're building music out of sound events as opposed to a guitar, bass and drums. So it's very much like a collage or mosaic — very finicky, you're piecing it all together, little elements, dotting them in there. For example, on 'Hairy Trees' there are something like 12 bass parts that make up 'the' bass part from different elements. I wish we could find a quicker way, because sometimes it gets a little bit crazy, especially when we come to mix it. Tom Elmhirst's eyebrows would go up when I'd give him the track sheet and there would be 12 basses! I usually had to do a comp so that I could get his eyebrows back down again!"

Demo Vibes

"I don't know what the ideal mixing route is, but we had this tortuous route where we demo-mixed everything and then went to 'properly' mix it with Tom Elmhirst, who's very good. But he got a little bogged down, because a lot of the time the guide mixes would have a vibe that was kind of irretrievable and so sometimes he'd be scratching his head to figure out how to approach it. In the end we did this bizarre stem mixing between the demos and what he'd done — which I actually thought was really good, because you got something that was better than either mixes on their own. Sometimes you could build them by crossing over from one set of stems to another and you'd get this bigger sound as a result. It was very painstaking but that's just how we are. The mixing wasn't any different from the writing in that tortured and anal kind of way.

"Pro Tools is really good for stem stuff. When I did my guide mixes I did stems from them, and then Tom did the same from his mixes so that we could compile stems from both mixes and you could cross over element by element as you went along if you so wished. It was quite tortuous but as a result we got something better than either of us would have done without each other."

Will Gregory at work in his Bath studio.

Yet flexibility at the mixing stage is not without its drawbacks. Without imposing arbitrary deadlines, just when is a track deemed to be complete? It's a question that Gregory ponders thoughtfully. "To really know when a track is finished you need to have the luxury of time. You have to be able to listen to it a week or month later and not feel differently about it from the first time you listened to it. So the only way is to give yourself some distance from it. But the good thing about giving them some distance is that they tend to develop a rosy glow — they're more likely to survive if you've left them alone.

"Nick Batt helped a lot when we were writing and helped us to develop some of the percussion parts. Having Nick around was also handy when we were doing the demo mixes. It's funny how when you play something to somebody, even if they don't say anything or move a muscle — just the fact that you're hearing it in the room with somebody else makes you feel differently about it. And that can be very useful."

Another useful collaborator was Dave Bascombe, brought in to mix a number of tracks on *Black Cherry*. "Dave's great. We got a bit stumped with a couple of tracks, 'Hairy Trees' and 'Strict Machine', and he kind of clarified them a little bit. 'Hairy Trees' didn't seem to have enough richness to it at the time and he made a few crucial changes, like turning the vocals up in places where they'd always been too quiet. He also added some overall compression, which tightened it all up so that everything was more inside the mix rather than all over the place. But that's what mixers do, isn't it!" SOS

Click Tracking

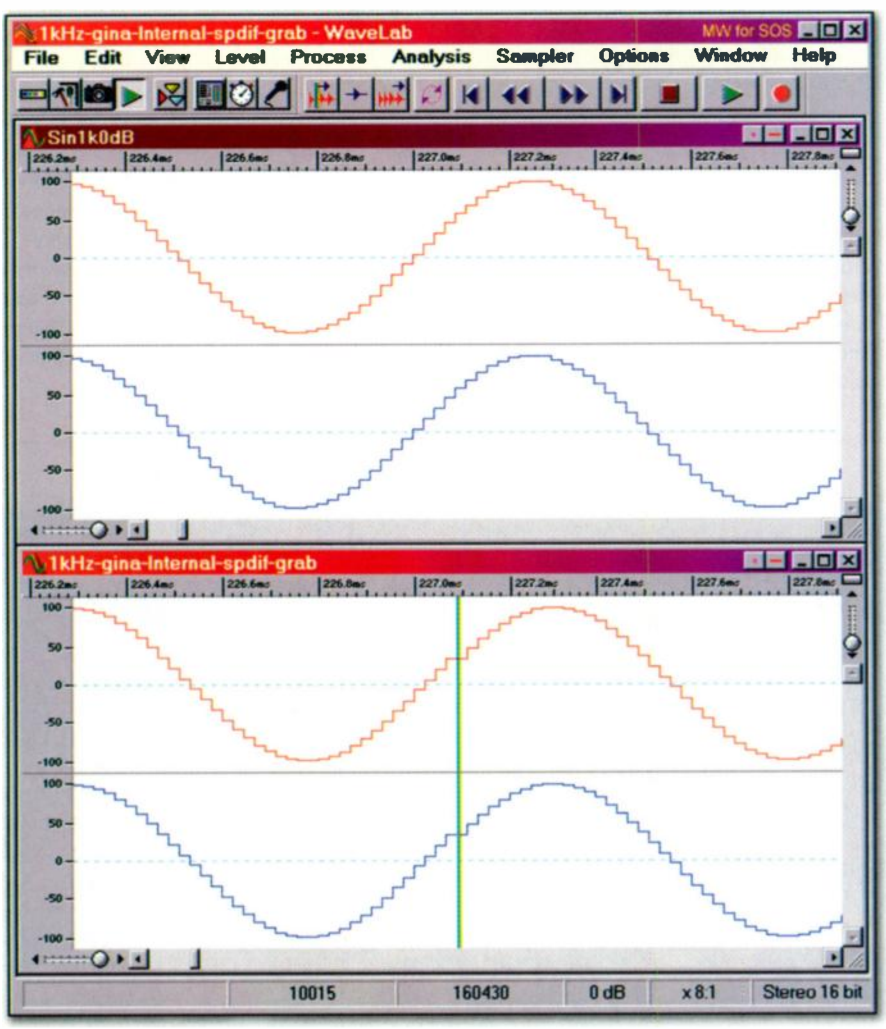
Eliminating PC Audio Glitches

Few things are more frustrating than unwanted clicks and pops in your recorded audio. But what are the main causes, and how can they be eliminated?

Martin Walker

Audio clicks and pops have been the bane of some musicians' lives ever since soundcards first added digitised audio recording and playback to their arsenal of features, and it's natural to assume that it must be the soundcard's fault when any glitch is heard. However, there are plenty of other possible reasons, some of which are quite mundane, and it makes sense to eliminate these first before delving into the virtual innards of your PC. One is faulty or intermittent leads, while another possible external cause is incorrect clock settings. If you're transferring audio from Minidisc or DAT and hear the occasional click in your recordings, try recording a 10kHz sine wave, and then transfer this. Since even a single missed sample will then be fairly obvious as a tell-tale 'tick', this will test your digital cables and settings. If the recording isn't locked to an external clock you may hear a fairly regular ticking, while not using proper 75Ω (S/PDIF) or 110Ω (AES-EBU) cables to transmit electrical digital signals can produce faults ranging from the occasional tick to no signal getting through at all.

Audio clicks and pops that have nothing to do with the soundcard or Windows settings can also occur during track grabs from Red Book audio CDs. These are often characterised by missing or repeated sections (see screen shot, right), and are due to 'jitter' when joining the various data chunks together. If you have a CD drive that suffers from this problem, most modern software has special extraction settings that can overcome it. However, before you resort to these, do make sure the disc itself isn't scratched, marked, or otherwise flawed, and that the CD lens doesn't need cleaning. Anyone who has Ahead's *Nero* can run the DAE Quality Test of its CD Speed utility, which will detail the number of read faults for a particular CD.



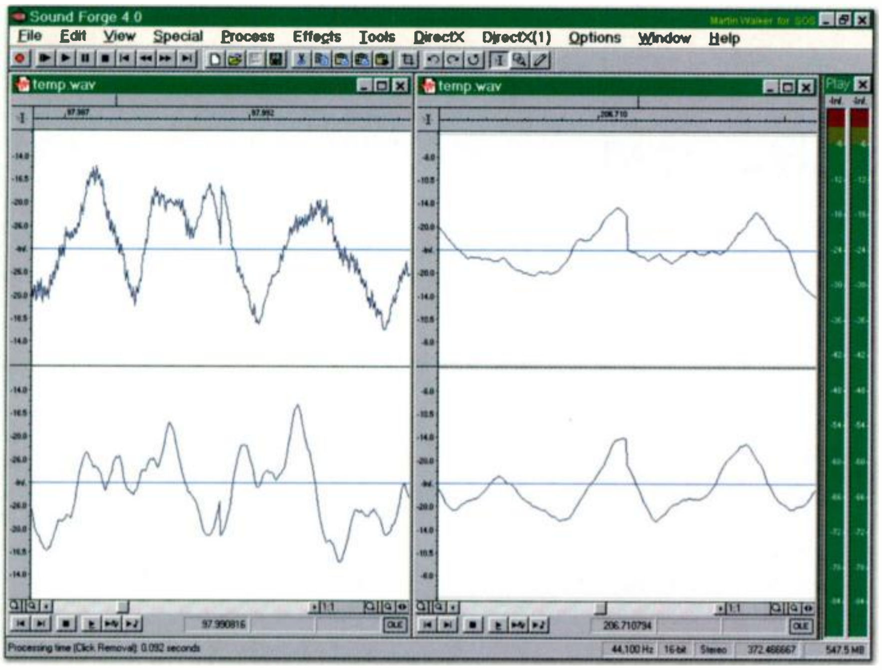
When transferring Minidisc or DAT recordings, glitches due to incorrect clock settings can be tracked down by recording a simple 10kHz sine wave signal, which makes hearing the tell-tale 'ticks' due to missed samples a lot easier (the 1kHz sinewave shown here makes it easier to see the missed sample, as in the lower window, but faults will be far easier to hear at 10kHz).

Narrowing Down The Problem

Before you start delving into Windows settings and adjusting arcane parameters in the nether reaches of your soundcard's software utility to solve clicks and pops, there are certain things you should and shouldn't do. First, don't be tempted to indulge in overclocking unless you really know what you're doing, since this can exacerbate any problems, particularly if you attempt to run the PCI buss faster than 33MHz.

Try to note down details about what's happening when the clicks occur, and in particular what software you're running at the time. If your PC has been running happily for months and then pops and clicks appear, try to remember what you changed or installed just before you noticed the problem.

However, don't assume that if your sequencer glitches on audio playback, this guarantees that it is to blame. Trying other basic audio software like Microsoft's Media Player to see if this suffers from a similar problem may help prove innocence or guilt, although a better test if possible is to run a



Don't confuse glitches like these with those caused by soundcards — I captured both of these while grabbing tracks from Red Book audio CDs, and both are caused by jitter. On the left a section of data has been repeated, while in the right window, the vertical jump in both channels indicates that a section has been missed out altogether.

stand-alone soft synth using the same type of audio driver, running with a similar latency. If this too exhibits clicks then your problem is more likely to be system-related rather than software-related, and even if you don't hear clicks, it doesn't necessarily mean that your sequencer is to blame — MIDI + Audio sequencers stress your hard drive, processor, and RAM in quite a comprehensive way. Remember, by the way, that sequencer CPU meters generally poll activity once every second or two before displaying the result, so a very short CPU overload may not register at all, despite causing a click and sometimes stopping your sequencer altogether.

If you actually capture some clicks and pops in your recordings, another way to narrow down the cause is to zoom right in on them in a waveform editor and examine them more closely. The classic 'digital dropout' has vertical sides and zero data throughout the interruption (see screen shot on the next page), is due to some interruption in the smooth flow of audio data, and can usually be tracked down to faults inside your PC. If, on the other hand, you find the clicks less predictable or more 'analogue' in shape, they are more likely to be due to external problems such as faulty synths, central heating boilers and the like. This sort of click can often be traced by noting when it happens, and then seeing if the heating has just switched itself on, or if the problem disappears after you've temporarily unplugged the fridge/heating/synth. It may even be arriving piggyback on the mains supply, especially if you're close to a local sub-station.

However, the simplest way to narrow down possible causes is to search the Internet for users complaining of similar problems, to see if you have any software or hardware in common. If you post your symptoms, you may even be lucky enough to get someone who recognises them and replies with some helpful suggestions. This approach can also weed out soundcard driver problems, and if enough users of a particular model have the same problem, the manufacturer almost has to take notice and do something about it.

Fundamental System Problems

Some PC components are notorious for compatibility problems that cause audio glitches, including early Via and AMD motherboard chipsets such as the KX133, AMD 750 and AMD 751, and USB click problems with some early Intel BX motherboards with two USB ports. In some cases like these, it's almost impossible to solve the problem, however you tweak your PC. Some people now say that motherboard chipset problems are a thing of the past, but it still pays to visit your soundcard manufacturer's web site to check on compatibility with your specific chipset before buying a new card.

If your clicks and pops are due to a fundamental motherboard problem, they won't respond to Windows tweaks, so you should always make sure you have installed the latest motherboard chipset drivers. Anyone with an AMD chipset should run the latest Driver Pack, Via chipset owners should make sure they have the latest 4in1 drivers installed, and most Intel chipset users should install the Intel Application Accelerator, as discussed in *SOS* April 2003. Opinions are more divided about updating the motherboard BIOS, since you can prevent your PC booting up at all if anything goes wrong during the procedure. If you have a specific problem that may be solved by whatever features are updated in a newer BIOS then follow the step-by-step procedure very carefully. If not, leave well alone.

Many specialist retailers who advertise in *SOS* set up their PCs in Standard Mode, perhaps because various M Audio cards seem to prefer it to ACPI mode. However, I still maintain that XP is best left in ACPI mode, especially if you have a PC with an APIC chip, unless you're suffering from clicks that don't respond to other Windows tweaks.

While we're on the subject of fundamental system problems, make sure your various hard and CD drives are connected in the most appropriate way to whatever IDE channels are available on your PC, to prevent them jockeying for supremacy. I discussed the options most recently in *SOS* January 2003. It also pays to make some considered choices about which expansion slot to use ▶

Preventative Measures

A completely separate music-only partition does take some extra effort to set up in the first place, but the chances of it developing problems in the future are far smaller when you're not browsing the Internet, playing games, and generally indulging yourself on a much bigger general-purpose Windows installation. You'll also make your life a lot easier if you regularly take images of your hard drive, so that if any audio fault does develop you can backtrack to a known state. I always make an image just before I install any major application, just in case of problems, and this is yet another reason to divide your drive into system and audio partitions, as it makes the image files much smaller. I've never had to create a partition larger than 2.5GB to hold Windows and its applications, especially with a multi-boot PC where each Windows partition only holds a subset of applications, and with the lossless compression options of suitable utilities like Powerquest's *Drive Image*, my image files rarely exceed 1GB.

Placing your audio files in a separate partition also reduces the likelihood of glitches, since you can regularly defragment this to ensure the smooth flow of its data during playback, and the smooth saving of data being recorded, without the read/write heads having to dart about between every available nook and cranny.

for your soundcard and any other PCI devices, to avoid the sharing of IRQs at motherboard level. Most modern soundcards will happily share with another device, but others won't, and nor will some other PCI cards. Look no further than *SOS* May 2003 for a comprehensive guide to this often confusing area.

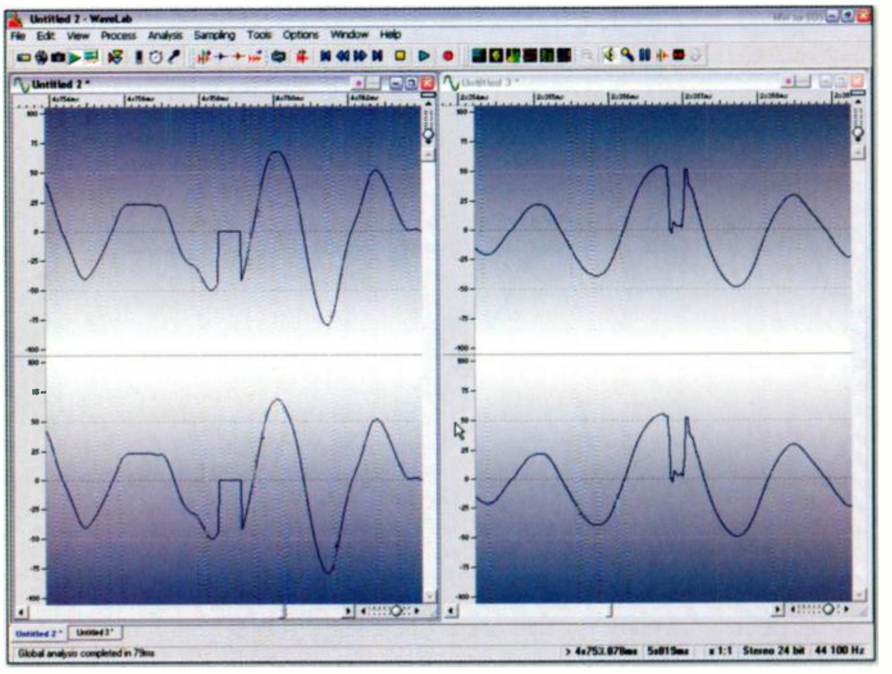
With laptops, additional factors can come into play, including BIOS and power management settings for long battery life that are unsuitable for hard disk audio recording, infra-red devices that are regularly polled for activity, and CPU fans that cut in when required, all of which can cause audio glitches (some Dell laptops are particularly bad on the last front). You may not be able to change these settings as easily as on a desktop PC, and sometimes not at all, since many laptops provide limited BIOS options. Since every laptop model can behave differently, be guided by other users before deciding which one to buy for audio work, and read my feature in *SOS* January 2001 for some more background information.

Last but by no means least, make sure you have no system conflicts, by having a look in Device Manager. Having an exclamation mark next to a non-audio device may or may not be relevant to your audio problems, but it pays to make sure before you tear out your remaining hair.

Unsuitable Windows Settings

Only now that you've removed any possible external causes of your clicks and pops, and confirmed that your PC doesn't contain any rogue devices, is it worth tweaking your Windows settings. I've covered general Windows tweaks in some depth in past PC Musician features: you can find extensive details about setting up Windows 98 in *SOS* January 1999, and for Windows XP in *SOS* March 2002. Moreover, you can find various web sites with all this information presented in checklist form, one of the best being www.musicxp.net. So let's see how to employ this information in tracing and eliminating the sources of audio glitches.

First up, and normally easiest to deal with, are interruptions due to graphics. You may be suffering from these if your glitches occur when you do something specific such as opening a new menu dialogue, quickly moving your mouse, or particularly when the screen is either scrolling or flipping from one image to the next. If you suspect this to be the case, try to minimise such opportunities by temporarily disabling screen updates in your sequencer, and leaving the mouse untouched. If this lessens the problem, or it goes away altogether, you can implement the various related tweaks. Extra CPU power will be used every time you use unnecessary



If your recording has a dropout with clean vertical sides as in the left window, and with zero data during the fault, this is almost certainly a PC-related fault that interrupted the recording process. The right window shows a similar PC dropout from a soft synth, but this time recorded through a soundcard's analogue input; despite the sides not being quite vertical, and having a slight overshoot, it's still likely to be a PC-related fault, since it once again has a value of zero throughout the interruption.

graphic frills, such as scrolling, fading, or exploding window transitions, balloon tips, and so on, so if these are the culprits it will be fairly obvious, as clicks will occur when this graphic activity is happening. Screensavers can also cause occasional clicks, and are best disabled.

Less obvious is the CPU overhead due to drawing or updating the main graphics. You should check that Graphic Acceleration for your graphics card is left at full, despite what some web sites tell you, since using other settings will simply increase CPU overhead as your processor has to do the work instead. In general, 16-bit colour will give the best combination of graphic excellence and low CPU overhead with modern music software, although some applications like *Wavelab* 4.0 specify 24-bit or higher.

If you're running a song that is already displaying a high CPU figure inside your sequencer and you start to get the occasional click or pop that doesn't respond to increasing the soundcard's buffer size, you can test whether the graphics card is at fault by changing the graphic colour quality in the Settings page of the Control Panel Display applet.

Hogging The Buss

Another cause of audio glitches gives no visual clues, and sometimes happens only very occasionally: hardware devices hanging on to the PCI buss and stopping the otherwise smooth flow of audio or MIDI data. For instance, anyone with a Promise FastTrack PCI controller may find their audio clicks and pops disappear after they open the Promise FastTrack Monitoring Utility, and in the 'PCI Bus Utilization' section on the Options page, move the slider from High to Low.

Sometimes it's the graphic card that may be causing the problem, so make sure you have installed the latest drivers for it, even if you don't hear audio clicks and pops. This can make a great deal of difference, as I found in *SOS* October 2002, when occasional jumps in MIDI latency disappeared after I abandoned the default Windows XP drivers for my Matrox G450 card and replaced them with the latest ones

Other PC Musician Resources In *SOS*

- January 1999: Setting up Windows 98.
- January 2001: Making music with laptop PCs.
- December 2001: Resources & background tasks.
- March 2002: Setting up Windows XP.
- September 2002: MIDI latency & drivers part 1.
- October 2002: MIDI latency & drivers part 2.
- January 2003: IDE connections.
- April 2003: Motherboard drivers.
- May 2003: Choosing PCI slots & interrupts.
- June 2003: Monitoring CPU & RAM consumption.

downloaded from the manufacturer's web site, as well as unticking their buss mastering option. Buss hogging is sometimes still a problem, but most cards have such options to improve matters for the musician. Although most of us now use AGP graphics cards, anyone considering adding a PCI card to their PC so they can attach two monitors should think again, since this will increase the likelihood of buss hogging.

Excessive CPU activity can also be caused by incorrect Bus Master DMA settings for your hard drives, which can cause intermittent clicks even though once again the audio application doesn't display high CPU readings. Windows 98/ME users need to set their drives manually, though 2000/XP users should find this done automatically, as will any user of Intel's Application Accelerator (see *SOS* April 2003) or specialist IDE controller chips. Sometimes Bus Master status isn't available to check, but there's an easy way for any musician to find out — just download *Dskbench* from www.sesa.es and run it. CPU readings lower than about 5 percent prove that your drives are correctly set up.

USB audio devices can also be prone to occasional dropouts, partly because they consume quite a bit of the USB bandwidth, but you can minimise such problems by ensuring that you avoid hubs. Use them for other USB devices by all means, but make sure your USB audio peripheral is plugged directly into its USB port. If possible, disable all other USB devices when you're making music, and it's also recommended that you disable network or modem cards while running USB audio.

Drive Activity & Housekeeping

Another class of pops and clicks can be caused by extra and unexpected drive activity when you're already pushing your PC near its limits. If you spot the activity LED on your PC's front panel flashing in sync with clicks and pops, in addition to its normal pattern of activity, then you should investigate further. Possible causes include Auto Insert Notification for audio and data CDs, which sometimes causes a regular extra flash every few seconds, and swap or page file activity, which is likely to be visible as a sudden burst of flashes lasting several seconds. This can mean the computer is saving data that can't be fitted into system RAM — in which case a RAM upgrade may be in order — or that it is resizing the swap file. Most musicians set the swap file to a fixed size or disable it altogether if they have sufficient system RAM (512MB or greater).

Another major class of audio glitches is caused by extra tasks cutting in that are completely unrelated to your audio application. These don't always cause no extra drive or CPU activity visible from within your sequencer; they may be caught by Microsoft's System Monitor, as I explained last month, but since they often last a very short time, its regular polling of overall CPU overhead may miss them as well. The most obvious candidate is Task Scheduler, which can defragment your drive, clean up unwanted files, or indeed run any desired task at a time of your choosing. However, while amassing some housekeeping duties to be performed every Friday at midnight may initially seem like a good idea, you won't think so when you happen to be recording late one Friday evening. Make sure it's permanently disabled and run these tasks on demand.

Windows 98SE users have more settings to adjust than 2000/XP users, and most are now fairly well known. Resizing of the cache RAM can cause untimely audio interruptions, but can be prevented by setting a fixed size for the vcache, either by hand or using *Cacheman* (www.outertech.com). Other occasional system functions to disable are Power Management, Automatic Update, and System Restore, all of which run in the background. Permanently disabling all System Sounds is also advisable, since some of the WAV files they use may be 8-bit or of low sample rate, which may cause your soundcard or Windows to implement sample-rate-conversion algorithms.

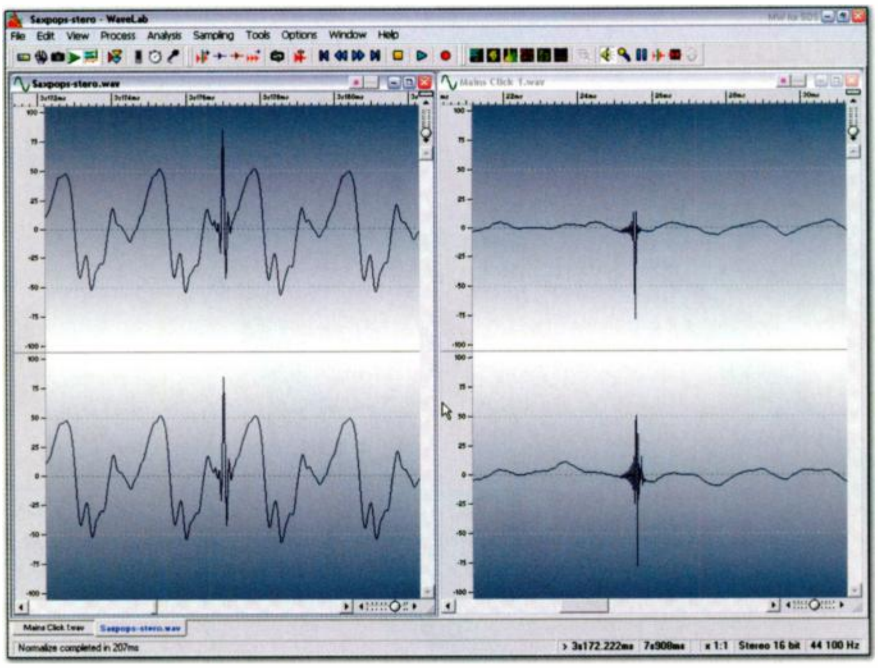
Other tasks may get installed behind your back by applications and expansion card driver installations. Some run only once when you start up your PC, and may be benign, but you can get a list of these by using the Start menu's Run option and typing in 'Msconfig'. Others may also be lurking in wait for the perfect audio performance before they cut in and ruin it for ever with a glitch, dropout or unexpected halt.

I discussed the majority of culprits in *SOS* December 2001, but remember that you can always get an up-to-date list of all the processes currently running on your PC by using the three-fingered salute (Ctrl, Alt and Delete keys). Even if you're not currently suffering from audio clicks and pops, the chances are that by dealing with unwanted background tasks you may be able to drop your latency to a lower value without glitching.

Soundcard & Music Application Settings

There's no point trying to tweak your soundcard or music application settings to achieve lower latency until you're happy that your PC hardware is properly installed with the latest drivers, and your particular version of Windows has been set up to suit the particular requirements of continuous audio recording and playback.

It's routinely recommended to increase the size of your soundcard buffers if you suffer from clicks and pops, as this gives your PC more time to recover from any interruption. Disk buffers don't normally cause as many click and pop problems unless you're running loads of audio tracks, in which case you can increase their size and



External clicks captured during an analogue recording can have many causes, but they usually differ in shape from PC-related ones by oscillating either side of the zero line. The left window shows a mains glitch or vinyl click, while the right window is the infamous Echo Mia glitch that used to emerge randomly from its analogue output, cured with the version 6.0 firmware update.

number; applications like *Cubase* provide a separate meter for hard disk transfer load to help you choose the optimum setting once you've eradicated any DMA problems as described earlier.

So, only when Windows has been tweaked and the interruptions minimised is it time to carefully lower your soundcard's buffer size until you start to hear clicks and pops, and then increase it to the next available setting. You may have to increase it still further when playing back more tracks and running more plug-ins and soft synths, and when simultaneously recording, but you should soon find a setting that works for most purposes. Although you can often cure clicks and pops by whacking the buffer sizes right up, you may just be masking the real source of the problem, and by tracking it down and curing it, you could be able to run your soundcard at a much lower latency. In fact, you can sometimes tell how well 'tuned' your PC is by listening to the glitches that you get at very low latency values. If there is a regular scattering of small dropouts then this shows that you don't have any major interruptions. However, if you get one random bad one every few seconds then it's more likely that there's a single cause.

It's particularly tricky when you attempt to run two music applications simultaneously, since you can't predict when each will demand its maximum resources, particularly when both are running on different stereo output pairs on the same soundcard.

If you get occasional clicks when your MIDI + Audio sequencer is already running with a high CPU overhead, try altering the Colour Quality to see if this improves matters.

Final Thoughts

Whenever and whatever you're in the process of fault-finding, it pays to be systematic, and to make some notes, so you can backtrack if necessary. If you simply blanket-bomb the problem with every Windows tweak you will possibly make the problem worse, or degrade your overall performance. Unless you suspect an unresolved system conflict that may require a fundamental change such as a new motherboard, once you've worked through all the main contenders it's always worth trying the ultimate overhaul — starting afresh from a newly formatted partition and installing a clean version of Windows and your music applications. SOS

Yamaha MG16/4

Mixing Console

A new entry-level mixer for simple recording and monitoring tasks.

Paul White

Along with many other audio equipment manufacturers, Yamaha have developed a budget range of mixing desks to compete in the lively 'cost-effective' end of the market. The 'origin' label affixed to the mixer indicates that Yamaha have taken advantage of low-cost Chinese manufacturing to achieve this feature set at such an attractive UK price. The MG series includes 16-channel and 12-channel versions featuring four-buss architecture, making them suitable for basic recording applications or for use as the control centre of a soundcard studio.

The 16-channel MG16/4 under review here follows the traditional wedge-shaped format, with all the connectors other than the headphone outlet on the rear panel — the phones output is at the top of the front panel. A neat blue and black paint job plus extruded alloy end cheeks give the mixer a very tidy and well-finished appearance, and the end cheeks can be removed to fit rackmounting strips if needed. Power comes from an external adaptor, with a sensibly heavy locking connector, and all the jack sockets are metal types bolted directly to the chassis. In common with many other current mixer designs, the channels are divided between mono mic/line channels and stereo line-only channels — in this case 12 mic/line channels and two stereo line channels. All the mic/line channels have balanced XLR mic inputs with globally switchable phantom power, balanced line inputs on quarter-inch jacks and insert points on TRS jacks. The line channels have two inputs (left and right) on both jacks and phonos, while the main output is available on both balanced XLRs and jacks. Two further impedance-balanced jacks carry the two group outputs, and there are phonos for tape in and out. Separate jacks are provided for the control room output, the two aux sends and the two aux returns. The latter are normally stereo, but can be used in mono by connecting to the left socket only.

A Glance At The Specs

On paper, the mixer is well specified, but without being esoteric. The mic input noise figure is -128dBu EIN, which is within a decibel of what virtually every other mixer claims for its mic inputs. The frequency response is pretty flat (+1dB/-3dB) from 20Hz to 20kHz, but there's been no attempt to extend the frequency response into the stratosphere. With a maximum output level of +24dB on the main stereo output, there's plenty of headroom for driving level-hungry soundcards, and the distortion with the gain control at maximum is just 0.1 percent.

Channel Facilities

The mono mixer channels themselves are reassuringly conventional, headed up by a gain trim control and an 80Hz switchable low-cut filter. The line channels have no input trim or filter, so the nominal level needs to be adjusted at source. The EQ has three bands, with shelving sections at 10kHz and 100Hz plus a fixed mid-band peaking at 2.5kHz. All have a ±15dB range. There are two aux send controls per channel, the first switchable pre/post-fader and the other fixed post-fader for use as an effects send. There's no EQ bypass, but all the EQ controls are centre-detented. Mono channels have conventional pan controls, while the line channels have stereo balance controls.

An illuminated On button allows individual channels to be muted or made active, and there's also an illuminated PFL (pre-fade listen) button for each channel, but not for any of the sends. PFL allows the pre-fader channel signal to be monitored in isolation when setting up the mixer. As usual, operating the PFL affects only the control room and phones outputs, not the main outputs, and the PFL level is shown on the main level meters. Each channel also has a single routing button which, when pressed, sends the channel signal to the two group outputs, rather than the master outputs. This is very useful in a recording situation, as you can use one buss for feeding signals to a recorder and the other for monitoring the outputs. Channel and master level control is via 60mm faders, all of which have a smooth, positive action.

Master Section

That leaves the master section, which has been kept as simple as possible.

A recessed, illuminated button switches on 48V phantom power for all the mic inputs, and there's a dedicated level control for the two-track inputs. A pair of 12-section meters monitor the main output, and there's another rotary control for the Control Room output level, which also controls the level sent to the headphone jack. The control room out can carry the main stereo output, the group output or the two-track return depending on how the two switches below the level control are set. If any PFL button is used, this overrides any other control room setting in the time-honoured fashion. Stereo faders are fitted for both the main mix buss and the group outputs, and the group fader is accompanied by a rather lonely looking 'To ST' button that can be used to route the group mix into the main stereo mix, something you might want to do when ▶

Yamaha MG16/4 £229

pros

- Good cost/performance ratio.
- Practical choice of facilities both for live and recording applications.
- Quiet, neutral sound.

cons

- Limited EQ, with fixed-frequency mid-band.
- No EQ bypass switch.

summary

A neat-looking little mixer that performs well and meets the needs of many 'small systems' users, both live and in the studio.

creating a drum or backing-vocal subgroup.

That leaves the auxiliary section, where the two master send level controls are self-explanatory. The aux return is a little more involved, as it not only includes a stereo level control, but also send controls for the two auxes. If the aux two control is used when the return is carrying the output from an effects unit fed from aux two, this is a recipe for feedback, but it's also possible to use the returns as just another stereo input, and in that case having both aux sends available allows effects to be added just as with any other channel.

When tested in real-world conditions using capacitor microphones, the mixer was quiet and transparent sounding, imposing no obvious character on the sound unless the EQ was brought into play. The high-end EQ was particularly nice, providing an airy quality to the sound, while the low EQ did what it purported to do with no surprises. Having a fixed-frequency mid-range control can be a little limiting and on some vocal tracks the 2.5kHz region sounded rather nasal or hard if boosted, but by the same token it was also possible to cut these frequencies where the sound being treated was too dominant in that area. Like most well-designed budget desk equalisers, this one sounded fine if used sparingly.

The general design of the mixer is practical without being too complicated, and the four-buss architecture lends itself well to small-studio applications, where you might need to record just one or two discrete tracks at a time, but monitor and mix, say, eight soundcard outputs. Having aux one switchable pre/post-fader is also useful, as you can use it in pre-mode for setting up a monitor mix, then flip it to post when mixing to use as a second effects send. A few tricks have been missed, such as not including a 'two-track to mix' button, but when designing a small mixer you have to decide what you can put in and what you have to leave out to meet the price point. On balance, the features that are provided meet the needs of most small-studio operators or those wanting a small live sound mixer, and I have no real criticisms other than perhaps the lack of sweep mid-band EQ controls and EQ bypass switches.

Verdict

Every mixer built to a price is a compromise, but Yamaha have made some good decisions here by creating a design that has practical applications both for live sound and for recording. It doesn't have direct outputs on the various tracks, but its buss structure makes it possible to either record up to four discrete parts at once or to record two parts while monitoring multiple outputs from a soundcard or recorder. Indeed, you could record four parts (five if you used the pre-fade send as another way into your recorder) while simultaneously monitoring the stereo output from a soundcard via the two-track input and control room outputs. Those outputs that aren't fully balanced are impedance-balanced, which works well enough in small studio setups.

As to the sound and general performance, I'm quite happy with the quiet, neutral sound of the mixer, though the fixed mid-band EQ control is quite limiting and I'd probably try to use it as little as possible. Artefacts such as distortion and crosstalk are low enough to ignore for all practical applications and there seems to be plenty of headroom in the output stage, which is useful when driving soundcards that are calibrated to peak at around +18dB. Ultimately then, the MG16/4 is built to a price and isn't without stiff competition, not least from other Chinese-built mixers, but it offers a good range of features, it is solidly built, it sounds good, and it looks the part. You can't really ask for much more than that. SOS

information

- £ MG16/4, £229; MG12/4, £159. Prices including VAT.
- T Yamaha-Kemble Brochure Line +44 (0)1908 369269.
- F +44 (0)1908 368872.
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11-pin Leslie cabinet connectors, labelled Main and Echo. The main output is exactly as its name suggests, but if a second Leslie speaker is required (a common setup for organ jazz combos) it can be plugged into the Echo socket (Echo being standard 'Leslie-speak' for a second speaker output). Usually a second Leslie switch pod is added to supplement the normal Leslie slow/fast switch, with Main/Ensemble/Echo switching, allowing either or both Leslies to be used, as required. Another point worth making here is that the Leslie 122XB rotary speaker intended for use with the New B3 includes a Stop (Brake) mode as well as the normal Chorale (slow) and Tremolo (fast) settings.

The two motor controls aren't strictly necessary on the new, digital B3, but the effect of the original controls has been modelled so that players who used the original switches creatively when playing don't miss out.

A metal plate under the organ console provides the MIDI output socket (there is no MIDI In or Thru) plus four quarter-inch sockets which provide an external stereo line input, and a stereo line output for the complete organ (including reverb if selected).

Technology

In some ways, the New B3 employs a 'back-to-basics' approach in a concerted effort to achieve the intended goal of producing a direct and indistinguishable replacement for the B3, retaining all the character and subtle nuances of the original. The heart of the original B3 is its electromechanical tonewheel generator, and the heart of the New B3 is, wait for it... A 'Digital Tonewheel Sound Engine.'

It seems obvious really, but Hammond have found the best way to recreate the characteristics of the original design is to reproduce each tone source individually and simultaneously — and it is therefore generating 96 individual tones at all times. These are not audio samples of tonewheels, but bespoke tones digitally generated and designed precisely to replicate the various artefacts of the original electromechanical system. There are two different sets of tonewheel waveshapes, called 'B-type' and 'Mellow' which replicate the slightly different tonalities of different vintage tonewheel organs.

Most modern keyboards only generate the specific required note when a key is pressed, and there is a limit to how many notes can be produced simultaneously — whether that is 32, 64, or whatever. The arrangement used in the New B3 means it has total polyphony — all notes can be played at once, just as in the original, and the sound is output the moment a key is pressed, not several milliseconds later after the required waveform has been found in the system memory.

Another aspect of the New B3's operation is that it employs a similar key-switching arrangement to the original instrument. There are nine palladium switch contacts operated by each key, one for each drawbar, plus a 10th contact to generate MIDI note and velocity data. The output from the appropriate tonewheel generator is routed as an analogue signal through the corresponding drawbar contact for each keyboard note, and then on to the associated harmonic drawbar mixer — and the audible key-click is actually generated as a by-product of this mechanical key-switching, just as on the original (apparently, the whole generator and key-switching system is the result of over seven years of painstaking research and development).

Consequently, there is no facility to adjust the amount of key-click, other than with the

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02

tone controls — what you get is what is created by the contact switching. Whether because of this keying arrangement, or because of the way the tones are generated, there is audible crosstalk from all the other tone generators when any note is played, just as on the original.

The signal-mixing system has also been engineered in such a way that the 'loudness robbing' which is such an essential part of the original Hammond sound has also been maintained. In other words, although the volume increases as more notes are played, it doesn't build linearly, but instead there is a degree of gentle compression, producing a fatter sound than might otherwise have been expected. This attention to subtle detail is obviously paramount when trying to produce a modern replica, and it can be seen all over the New B3. The volume curve and frequency response changes that are an integral element of the action of the expression pedal (foot volume controller) have also been reproduced extremely accurately (with three alternative settings to replicate different models of organ), for example. It's a subtle point and not widely known, but the amount of key click changes very slightly when the chorus/vibrato is switched on in an original B3, and the New B3 shows exactly the same effect — now that really is attention to detail!

If you have already read the box below about the history of the original B3 instrument, you might have noticed a discrepancy in the number of tonewheels.

A Little History

The B3 was introduced in October 1954 and production continued with only very minor (and mainly cosmetic) changes right through to the end of 1974. Well in excess of 275,000 B3 and related organ models were produced in that period, and a large number are still in daily use. Consequently, there is a thriving spare-parts and servicing market for these instruments all around the world.

The B3 is an electromechanical rather than electronic instrument, with closer technological ties to an electric guitar than an electronic synthesizer. The instrument is housed in a huge wooden case supported on four spindle legs, and a separate power amplifier and speaker system is required (typically a Leslie rotary cabinet such as the 122). The main case of the B3 alone weighs around 310 pounds and the complete ensemble with pedalboard and bench is close to 425lbs (roughly 190kg), making it less than popular with roadies!

The musician is faced with a pair of 61-note keyboards with square-fronted 'waterfall' keys, and a removable 25-note flat radial pedalboard. The various controls are arranged across the full width of the instrument above the top keyboard, with various rocker tabs, drawbars and switches to control the tonality of the beast. Each keyboard also has an octave of reverse-colour notes at the left-hand side which are actually latching switches to access one of nine user-programmable presets or one of the two sets of nine drawbars associated with each keyboard (programming requires a screwdriver and the moving of sets of nine wires on various metal busbars in a frame inside the organ).

Although the B3 is probably the best-known Hammond model, there were several near-identical siblings, differing mainly in their case design with some minor feature-set changes. For example, a more ecclesiastical case style was available as the C3 model, and built-in reverb, amplification and speakers were added for the A100. For more classical applications there were also the RT3 and D100 instruments which had a 32-note concave pedalboard and some additional pedal voice-generation circuitry (the RT3 used a C3-style case while the D100 was similar to the A100 with built-in amplification and speakers).

The nine drawbars are the musical heart of the organ, allowing the sound to be built up using individual pure tone (near sine-wave) harmonics, each at any of nine volume settings (Off and 1 to 8). Based on pipe-organ parlance, the fundamental drawbar is an 'eight-foot' voice (the dimension represents the length of pipe required to produce a middle 'C' pitch). There are also 16-, 4-, 2-, and 1-foot drawbars producing one octave below and consecutive octaves above the fundamental (sub-octave, 8th-, 15th- and 22nd-scale intervals respectively). In addition, 5 1/3-, 2 2/3-, 1 3/5-, and 1 1/3-foot drawbars allow harmonically richer tones to be produced, adding the 5th, 12th, 17th and 19th intervals respectively.

There are also two mixture drawbars associated with the pedals (with fundamentals of 16- and eight-foot voices), a percussive tone effect for the upper (Swell) manual, and a chorus/vibrato generator, switchable to each keyboard independently. Some models also included a spring-type reverb unit.

The core components in all of the tonewheel organ models mentioned above were identical — the massive tonewheel generator with its 91 active tonewheels and its two motors, the vibrato/chorus 'scanner', the arrangement of key contacts, drawbars and transformer mixing components, the percussion circuitry, and of course, the valve preamplification.

In fact, the tonewheel generator, key contacts, drawbars and transformer mixing arrangements changed very little from the original Model A throughout the entire life of the tonewheel console instruments. There were actually only relatively minor incremental changes to the other aspects of the machine too — and so the essential sound remained almost identical throughout the tonewheel Hammond's long life of over 40 years.

The most significant difference between a B3 and its earliest predecessors was the introduction of the 'scanner' and associated delay line to generate the vibrato and chorus effects. The original Model B (launched in 1936) was fitted with a complete second tonewheel generator set using slightly sharp and flat tonewheels to enable the creation of a realistic chorus effect (with a massive increase in weight too)! In the early 1940s, the first scanner and delay line system was introduced to the model line-up, producing the V variant (such as the Model BV), and this design was improved to enable the effect to be applied to each keyboard independently in 1949, with the release of the Model B2.

In 1954 the 'percussion circuit' was added to the upper keyboard, allowing the second or third harmonic to be given a percussive envelope while the remaining tones were routed through the vibrato/chorus system, and this model — the B3 (and its siblings) — was never bettered... until now!

For more on the Hammond and its history, check out Rod Spark's article on the subject in *SOS* October '97, or head for www.sound-on-sound.com/sos/1997_articles/oct97/hammond.html.

Photo: Hugh Robjohns

The pull-out Information Centre is a powerful addition to the new B3, allowing you to edit many of the parameters relating to sound generation.

The New B3 has 96 digital tonewheel generators whereas the original B3 (and its siblings) had 91 active tonewheels (although there were actually 96 physical wheels in the generator, the unused ones being there to maintain the mechanical balance). However, if you do the sums, you'll find that there are actually 109 different tones required to produce nine harmonics on a 61-note keyboard. The original Hammonds overcame this problem by cheating — partial octaves were repeated at the top end of the keyboard, the repeating starting on different notes and octaves depending on the drawbar. On a stock B3 the 2-foot drawbar repeats only the top 'C' from the octave below, while the 1-foot drawbar repeats an octave and a half starting from the 'G', and the intermediate drawbars start repeating somewhere in between. This arrangement is usually referred to as 'fold-down' and it is an essential element of the subtle but characteristic tonal quality of the B3 and its siblings.

However, some of the later Hammond organs (like the H100 and X77) were fitted with a full complement of 96 active tonewheels, adding an extra half-octave of notes at the top end which allowed the amount of fold-down to be reduced as well to enable the introduction of seventh, ninth, 10th and 12th harmonics (usually paired as 7/9 and 10/12 combinations on two new drawbars). These organs had a subtly different, slightly brighter tonal character.

The bottom octave of the lowest drawbar was also repeated from the octave above on the original B3 and its siblings, to avoid producing too muddy a sound (the lowest tonewheel frequencies are normally only available on the pedals). However, all the earlier and some later models retained the full range of the 16-foot voice on the manuals and many B3s were hot-rodded to enable bass lines to be played with the true 16-foot pitch notes in the bottom octave of the lower keyboard.

So, returning from this small diversion, the New B3 has all 96 frequency generators available, and the fold-down configuration can be adjusted to set precisely where the top and bottom notes are repeated. The default mode is to replicate the stock B3, naturally, but it is simple to recreate the tonality of other Hammond organs, or indeed to create a new fold-down configuration.

Information Centre Facilities

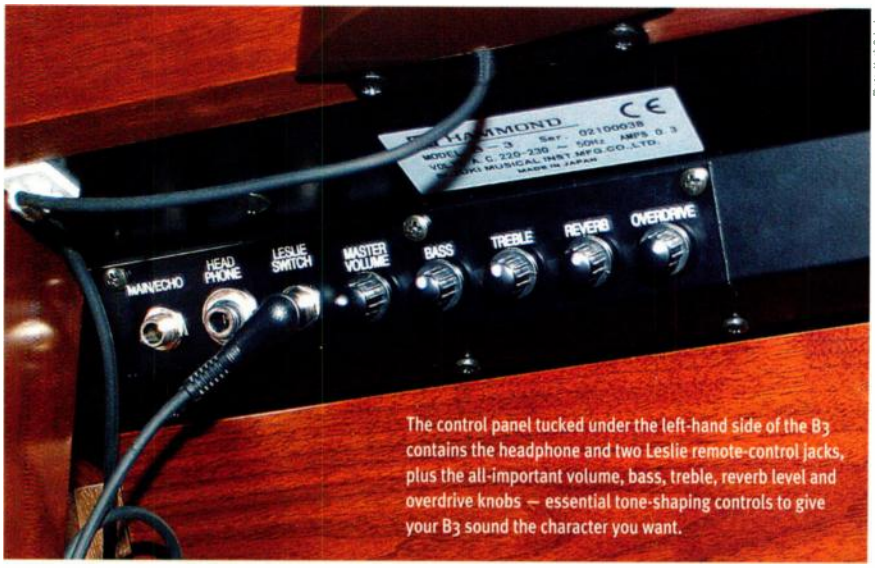
Hidden in a pull-out drawer under the right-hand side of the lower keyboard is the 'Information Centre.' This comprises a 2 x 20 LCD screen with nine buttons to navigate the half dozen or so menus, select the various options, and adjust the available parameters. Everything that was inherently adjustable or which varied with different B3 models can be configured here, along with many hot-rodding parameters to suit pretty much any personal preference spanning the full range of B3 characteristics. So, the tonewheel set can be changed (independently on manuals and pedals) to give different tonalities, and the percussion volume, sustain, keyboard level, and 1-foot cancel mode can also be adjusted.

On a stock B3 the keyboard drawbar level is reduced by about 3dB when the percussion is switched to its Normal (loud) mode in order to preserve the overall volume, but many players prefer to disable this function and that can be replicated on the New B3. Similarly, the 1-foot drawbar is normally disabled when Percussion is switched on (the 1-foot key contacts are used to trigger the sustain circuit) but many players rewire the organ to use a less important drawbar contact so that the 1-foot pitch remains available when percussion is used. Again, this feature can be replicated through the Information Centre menus.

A rather more unusual feature is that the vibrato rate can also be changed, the expression pedal minimum volume can be adjusted, and the volume versus position curve altered. The side switch on the expression pedal can also be programmed either to toggle the Leslie speed, force the Leslie to run fast while pressed, or to act as a sustain damper pedal (probably most useful for controlling external MIDI sound

generators). Standard Liturgic, Jazz, or Theatre voice banks can be allocated to the preset keys (or user-programmed variations), and there are facilities to store or recall preset voices to and from a Type I compact ▶

Photo: Hugh Robjohns



The control panel tucked under the left-hand side of the B3 contains the headphone and two Leslie remote-control jacks, plus the all-important volume, bass, treble, reverb level and overdrive knobs — essential tone-shaping controls to give your B3 sound the character you want.

Flash Card.

The latter can only be accessed by removing the rear panel of the case, which seems unnecessarily awkward. However, since all settings and custom configurations, including the MIDI setups, can be stored in both the internal user memories and the CF card, a touring Hammond player could carry his or her own preferences to instantly reconfigure another New B3 anywhere in the world.

You can also select a digital reverb type, the overall tuning pitch, key-transposition modes, and the MIDI configuration, and the organ's output tonality can be optimised for solid-state or valve Leslies. Given the attention to detail in other aspects of this replica, it's a shame that while the built-in digital reverb facility includes programs for two halls, three rooms, two churches and a plate, there is no spring-reverb setting — a popular choice of effect with real Hammonds.

The pedalboard on the original B3 was effectively just another keyboard, so it was therefore fully polyphonic and the notes stopped when the pedals were released. Some later instruments used an electronic system for generating pedal tones and introduced a plucked-string effect where the note decays slowly after the pedal is released, but with the side effect that the pedals became monophonic (highest note priority). Needless to say, the New B3 can be configured through the Information Centre for normal or muted (softer) bass pedals, polyphonic or monophonic operation, and pedal sustain can be switched on or off with a user-determined decay time (scaled simply from 1 to 5).

The MIDI facilities of the New B3 are similar to those on the decade-old, MIDI-capable Hammond XB3 and XB5. The keyboards are velocity-sensitive and zone-mappable with octave-switching facilities, and the reverse-key presets and drawbars can be used to send program and channel control data. The pre-production models of the New B3 were also equipped with aftertouch (as on the XB3) but apparently this has been dropped from the production models because it made the key action too long and detracted from the accuracy of the replica — and few players felt it a necessary feature anyway.

Having mentioned MIDI, I should point out that there is only a MIDI Out socket, so while the New B3 can be used to control an external expansion rack, say (allowing you to play a nice Rhodes piano sample from the top half of the lower keyboard, for example), there is no MIDI In or Thru facility. This is because the note generation is controlled by the physical key contacts, and can therefore not be activated remotely.

You Spin Me Right Round — The Leslie 122XB

The Leslie speaker intended to accompany the New B3 is the 122XB — a model that was originally developed for the Hammond-Suzuki XB series of organs. This cabinet is identical to the original Leslie 122 and the internal mechanics are very similar, with a rotating treble horn and a polyurethane foam bass rotor. The speaker weighs about 150lbs (70kg).

It is connected to the organ with the modern 11-pin plug which uses low-voltage switching instead of the high-voltage phantom system used on the original six-pin Leslies, and mains power is provided via an IEC connector directly to the built-in amplifier chassis. Like the original designs, the 122XB uses a single-channel 40W valve amplifier (again similar to the original design and using the same 6550 output tubes), and a passive crossover connected to a pressure driver for the horn and a 15-inch woofer. There are three speed options: slow (choral), fast (tremolo) and off (brake).

The optional Leslie 122XB rotary-speaker cabinet.

At present there is no equivalent to the Leslie 122RV cabinet which had a built-in reverb spring line, separate amplifier and static speaker fitted in the side of the Leslie cabinet. The significance of this is that because reverberation is added internally in the New B3, the reverb is also 'Leslied' along with the rest of the organ sound when using the 122XB rotary speaker.

A straight B3/122RV combination has no reverberation at all, and many players don't use reverberation anyway — but where they do, it is traditionally through a B3/122RV setup. This gives a non-dopplered reverb tail which not only sounds more natural, but also enhances the Leslie effect by providing a constant pitch reference. This is not currently possible with the New B3/122XB combination. It's a small point, perhaps, but one which struck me immediately as a change in the sound character.

Photo: Hugh Robjohns

Inside the 122XB. The treble horns can be seen at the top of the unit, while the rotating polyurethane drum which shapes the sound of the bass speaker is visible at the bottom.

Test Spec

- New B3 firmware version reviewed: v1.02.

Organ Grinding

After playing the New B3 I can say with all confidence that this instrument is virtually indistinguishable from an original machine — certainly judging by sound alone. Naturally, with the same controls placed exactly where they should be it feels very familiar to play too. Although the stock sound is stunningly faithful, it is also easily tweaked to closer replicate the broad range of real Hammond organ tonalities from bright or even screaming, through smooth and mellow, to downright dirty!

The adoption of individual contacts for each drawbar pitch on every note not only produces real key-click (with all the inherent variations each time a note is played), but also causes the different harmonics to be introduced one at a time as a note is pressed slowly. Some players claim this is an important performance aspect of the instrument — a kind of touch sensitivity — and although I'm not personally convinced by the argument, the effect is there anyway! I found the key click initially rather more prominent than I like, personally, but a gentle reduction in the treble control soon brought a balance which was more to my liking. This was using the standard B3 tonewheel set, and when I changed to the Mellow set I discovered an overall sound much closer to my own vintage A100.

From the performance point of view

I found the keys to be a little more 'springy' than my own Hammond, but then the review organ was brand new, whereas mine has been played for 40 years, which is bound to make the keys a little slack! With the waterfall-fronted keys, palm glissandos are as easy and satisfying as on the original, and the drawbars (again a little tighter and 'notchier' than my own instrument) were delightful to 'play'.

The customisation features accessed through the Information Centre expand the flexibility of this instrument considerably, and I particularly found the ability to adjust the volume of the percussion and keyboard useful. The pedal sustain was also nice for some kinds of music, and the facility to change the tonewheel set and fold-down configuration is excellent, allowing the user to change the entire character of the organ from a bright, 1970s-sounding instrument with the late-model high-end filter networks installed, to a more mellow 1960s organ full of crosstalk, beer and smoke — and all at the touch of a few buttons! As they say in the best jazz clubs: "*Niiice...*"!

The adjustable overdrive is fantastic too — this is the one of the most natural valve-overdrive systems I have yet heard, really capturing the traditional effect and allowing everything from full-on Jon Lord rock grunge, to a gentle grit, to an ultra-clean gospel sound. I wish it was that easy to change the amount of distortion on my vintage organ!

The Leslie 122XB is well established and makes all the right kinds of noises — both electrical and mechanical — and the inclusion of the Brake mode which stops the drum and horn completely is a welcome facility. Although I'm not used to using it, the Leslie speed switch on the expression pedal makes it easy to play with both hands while still winding the Leslie up and down. However, this can also lead to some confusion as the half-moon Leslie control switch on the bottom keyboard rail can end up out of sync with the Leslie's actual status, and you have to change the switch position to regain control. I found it less confusing to reconfigure the expression pedal switch so that it had a momentary action which forced the Leslie to fast mode.

Bottom Line

I don't think there can be any doubt that the New B3 is a true replica of an original B3 — both in terms of the look and layout, and the actual sound. I honestly don't believe anyone could tell the new instrument apart from an original if it was set up appropriately. This new organ looks the same, it sounds the same, it plays the same, it has the same quirks and characteristics, but it can also be tweaked and configured to replicate the most common and important hot-rodding techniques, which extend its capabilities just enough to satisfy the widest possible range of B3 lovers.

Although I am ecstatic that Hammond-Suzuki have laboured for so long and so lovingly to recreate the original Hammond tonewheel sound with modern technology, I'm not entirely clear as to the intended market. This instrument is seriously expensive and more than twice the price of a really nice original tonewheel Hammond complete with Leslie. Okay, so B3's are extremely rare in the UK — but they do still turn up — while C3s and A100s are widely available and cost a fraction of the asking price for a New B3 and 122XB.

Having said that, an original instrument is going to be at least 30 years old, so will need regular (and sometimes expensive) maintenance. It is also extremely heavy and difficult to gig with, and the tonality of each instrument is more or less fixed unless you are prepared to give it a major overhaul and makeover. It won't have MIDI either, and although this can be retrofitted, there are serious compromises and it is an extremely expensive process. Compared to all that, the New B3 is substantially lighter, has far more flexible tonality, built-in MIDI facilities, and should prove as reliable as any other modern keyboard.

So to whom will it appeal? The average gigging keyboard player looking for a Hammond-esque sound is already pretty well catered for, and I would have thought that most serious Hammond nuts would prefer to cosset an original instrument since the idiosyncrasies of an old tonewheel Hammond are an inherent, almost organic part of its character, just like a much-loved original Fender guitar.

The Leslie controls on the B3 itself. Care needs to be taken when using these in conjunction with the Leslie speed switch on the expression pedal, though, as the switch here can get out of sync with it.

However, there are many home organists who might like to go down this route, and with a MIDI expansion rack to provide additional voices such as strings, brass, pianos and so on, the New B3 would make a fabulous centrepiece for an impressive home setup — and with a cost not too dissimilar from some of the state-of-the-art home organs currently available.

Apparently Hammond foresee the biggest market area to be Gospel churches, especially in the USA, and professional Hammond players — like Jimmy Smith, for example, who is already using one of the new B3s — will appreciate the convenience and reliability of the new model too. These benefits will appeal equally to recording studios and musical-equipment hire companies as well. The ability to change tonewheel sets to manipulate the sound quality, and to adjust other performance and sonic characteristics of the organ to suit different user requirements will be a major boon to these organisations as well, since it enables them to tailor the organ to the user quickly and easily, rather than having to maintain several different instruments with different characteristics.

Understandably, the production run of Hammond's New B3 will not be huge — the instrument is virtually hand built — but the official word is that it is an ongoing project and they will continue to manufacture the beast as long as orders come in. Judging by the reception the New B3 received at its NAMM and Musikmesse launches I would imagine that much of the technology developed for this fabulous organ will spin off into other keyboards in the near future too — although I'm told that the key-contact mechanism requires a lot of depth below the keyboard which makes a single-manual version — along the lines of the XK2 — rather difficult to engineer.

However, for B3 *aficionados* everywhere, the New B3 is an immensely important product, and full marks to Suzuki for investing the time and funds to create a replica which is, finally, truly indistinguishable from the original in every important way. I think we will be seeing a lot of this instrument in the months and years to come. SOS

Thanks to John Henrys (+44 (0)20 7609 9181) for the use of the B3 pictured in this article.

information

- **S** New B3, £14,995;Leslie 122XB, £2495. Prices include VAT.
- **T** Hammond-Suzuki Europe (UK Sales) +44 (0)802 157586.
- **F** +44 (0)1582 768296.
- **E** info.uk@hammondsuzuki.com
- **W** www.hammondsuzuki.com

The Sound Guy *SFX Machine RT*

Formats: PC & Mac OS 9 VST, Mac OS X VST & AU

SFX Machine has been around for several years now, and in its original incarnation as a Mac-only Premiere-format plug-in, won wide acclaim for its extremely flexible and frequently outlandish sound-mangling capabilities. However, Adobe's Premiere plug-in format, while perfectly usable, was never the most popular choice among musicians, partly due to the limited range of host applications supporting it, and partly because it doesn't offer the same convenient real-time operation as, for instance, Steinberg's VST format.

With *SFX Machine RT*, The Sound Guy Inc (sound designer and software developer Earl Vickers, with the assistance of Marc Poirier) has addressed both of these concerns by porting the core *SFX Machine* code to two different real-time plug-in formats: the aforementioned VST (for Mac OS 9, OS X and Windows) and Apple's new Audio Unit standard (for OS X).

SFX Machine RT appears superficially very similar to its predecessor: the user interface is divided into two halves, with an extensive list of presets to choose from on the left, and a row of faders to adjust the parameters of the active preset on the right. Not only can you adjust fader values in the usual way, but the range of each fader may also be altered by entering the desired Min and Max values in the fields above and below. Choosing very high Max settings for one or more faders in a preset allows you to transform it radically — very often to the point where it becomes something altogether different. Your warped patches may be saved either in the standard AU/VST preset file format, or (in some plug-in hosts) simply by saving a song or project file. It's also possible to import presets saved with the older Premiere version of *SFX Machine*.

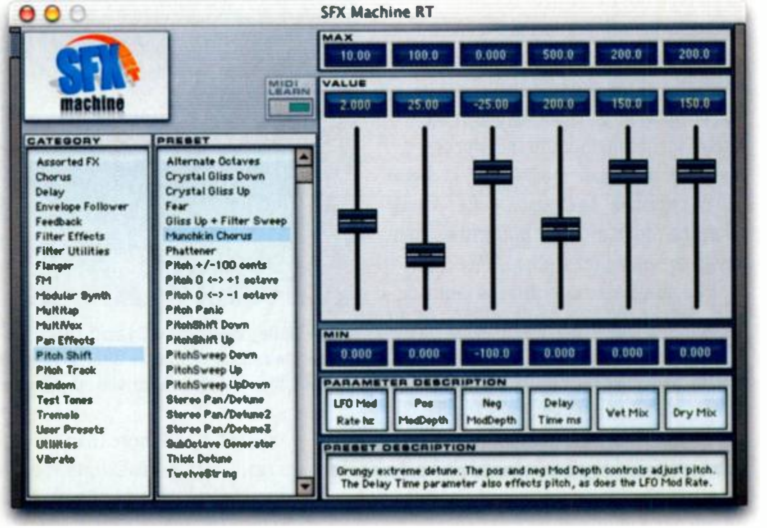
While presets can be modified and saved in this way, *SFX Machine RT* does not currently allow you to build entirely new patches from scratch via the slightly intimidating modular interface its predecessor provided. Some users may not even notice this absence, but others will mourn it — and especially the much-loved random patch generator. The Sound Guy's developers are working to implement a patching window in a later real-time version of *SFX Machine*, however, so all is not lost. In the interim, Mac users who purchase *SFX Machine RT* are entitled to download the old Premiere version as well, which they can use to make new presets which can then be imported.

SFX Machine RT offers literally hundreds of built-in presets, in no fewer than 21 different categories. These range from the conventional (chorus, flanging), via the more unusual (megaphone and vinyl noise simulations), to the utterly unique and startling ('UFO Descending' for instance, or the superb pitch-tracking Theremin emulator). Fans of the original *SFX Machine* will be pleased to discover most of their old favourites included, along with some new ones, all now working in real time as promised.

The sheer range and quality of effects offered by *SFX Machine RT* is staggering. Starting with a simple drum loop sample, I began auditioning the presets one by one. Half an hour later, I was still going, and my humble drum loop had undergone dozens of surprising and exciting transformations, many of which rendered it completely unrecognisable. As a creative sound design tool *SFX Machine RT* could be extremely valuable, as it consistently produces unexpected, interesting and usable noises and textures. There are few if any effects that can't be coaxed out of this plug-in — and there are one or two stranger ones that you probably won't find anywhere else.

New to *SFX Machine RT* is the option to map MIDI Controllers to the parameter faders in a patch (provided you're using a host application that allows you to send MIDI data to plug-ins). An ingenious MIDI Learn function allows you to assign any MIDI CC to a parameter by simply Control-clicking and sending a message from your preferred controller. In this way you can take advantage of all your sequencer's MIDI editing capabilities, and program any number of precise fades or sudden tweaks during the course of a song. This is great fun, and it's easy to come up with results that are quite striking.

Unsurprisingly, all this clever real-time processing comes at some computational cost, and while I certainly wouldn't call *SFX Machine RT* inefficient, I would caution that some more complex patches will place a fairly hefty load on your CPU. Newer Macs and PCs should be able to rise to the challenge, but my old G3/400 test machine really began to show its age at times.



SFX Machine RT works with any VST plug-in host under Mac OS 9 or Windows, and any Audio Unit host under OS X. The VST version for OS X works in any host that supports so-called 'Mach O' (ie. OS X-native, not Carbon) plug-ins. *Cubase SX*, *Metro* and recent versions of *Peak* and *Spark* should be fine. Other applications may not. You can easily find out if your preferred host works by simply downloading the demo version and giving it a try. For an asking price of just $99 (about £62), *SFX Machine RT* is excellent value for money. There's really nothing else quite like it on the market, and its originality of design is reflected in the wide range of sounds it's capable of producing. If you're interested in doing something a little out of the ordinary, this could be just the tool to do it with. *Paul Sellars*

£ $99.

W www.sfxmachine.com

Plug-in News

Applied Acoustic Systems have updated their popular *Lounge Lizard* electric piano emulator to version 2. This implements OS X and RTAS support, while new features include a preset browser similar to *Tassman*'s and the ability to sync effects to host tempo. It should be available as you read this. Meanwhile, the *Tassman* modular synth has also been ported to RTAS.

W www.applied-acoustics.com

Wave Arts are also entering into the OS X fray, having ported their three major plug-ins to both VST and MAS formats. *TrackPlug*, *MasterVerb* and *WaveSurround* are believed to be the first third-party plug-ins to support *Digital Performer*'s proprietary MAS format under OS X.

W www.wavearts.com

MXL Valve Mics

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W www.yamaha-music.co.uk
W www.yamaha.co.jp/english

Recording with digital equipment can produce very clean and accurate results, but the flattering, imperfect sound of analogue gear is often missed. Consequently, as digital equipment has become more common in studios, the market for classic analogue front-end preamps and valve microphones has grown greatly. Up until recently, valve condenser microphones were relatively expensive when compared to dynamic mics, but today there are many high-quality condenser microphones on the market at reasonable prices. Marshall Electronics were one of the first companies to start producing affordable valve-driven condenser mics, and *SOS* reviewed their MXL V77 tube condenser microphone back in January 2001. This month, MXL's UK distributors, Yamaha-Kemble, are giving away seven MXL mics to *SOS* readers, including the aforementioned V77 as first prize.

The MXL V77 has an RRP of £699 which includes a shockmount, power supply, wind screen and 15-foot cable. The mic is designed to provide the characteristically warm sound of popular mics from the '50s and '60s. MXL have used a low-noise valve, a transformer-free circuit and a large one-inch diaphragm so that the V77 benefits from the best of both old and new technology.

The mic's diaphragm is a mere 3 microns thick, giving it an extremely low mass so that it is able to respond quickly to the steep transients of percussive sounds, for example. Its size-to-weight ratio also helps the mic to deliver its 20Hz to 20kHz frequency range. The mic's polar pattern is a fixed cardioid type making it suited to most studio recording applications. Maximum SPL handling is quoted at 122dB.

Our second MXL prize, the V69, is another large-diaphragm tube condenser microphone with a fixed cardioid polar pattern. Its RRP of £329 also includes a shockmount, power supply, wind screen and cabling. Like the V77, the V69 has a large 25mm diaphragm and a frequency response ranging from 20Hz to 20kHz. It is capable of handling maximum SPLs of 140dB while suffering just 0.5 percent total harmonic distortion. The mic's specification boast an extremely low noise level and a wide dynamic range, while its sound is intended to mimic vintage valve models. The mic also has vintage-style cosmetics exemplified by its 24 carat gold-plated grill and brass enclosure.

To each of the next five winners, Yamaha are offering an MXL 990 worth £89 including VAT. The MXL 990 is also a condenser microphone, but it has an extremely competitive price and is, therefore, aimed at musicians on a tight budget. Unlike the V77 and V69, the 990 doesn't use a valve in its circuit, although it does use a good-quality FET preamp and runs on 48V phantom power. Its diaphragm measures 20mm in diameter, is 6 microns thick and provides a fixed cardioid polar pattern. The mic's frequency range is from 30Hz to 20kHz and its maximum SPL handling is measured at 130dB. The 990 comes with case, shockmount and mic clip.

To win yourself one of these excellent studio microphones, you just need to fill in the entry form on this page and post it to the address on the coupon. (Alternatively, you may enter via the electronic form on the SOS web site.) Please make sure you answer all the questions and complete the tie-breaker. We also require your full address and **daytime** telephone number. Your entry must arrive with us by the closing date of **29th August 2003**. SOS

questions

How thick is the diaphragm of the MXL V77?

- a. 6 microns ☐
- b. 3 microns ☐
- c. 2 microns ☐
- d. 13 microns ☐

What sort of mics are the MXL V77 and MXL V69?

- a. Large-diaphragm ☐
- b. Small-diaphragm ☐
- c. Platinum-diaphragm ☐
- d. Long-diaphragm ☐

Which polar pattern do all the mics in this competition share?

- a. Figure of eight ☐
- b. Diamond ☐
- c. Triangle ☐
- d. Cardioid ☐

MXL tie-breaker

The MXL mics are designed to provide the characteristics of vintage mics. Which modern recordings do you think could do with a bit of vintage tone and why? Answers in 30 words or less please.

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Daytime tel. no: ....

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

Post your completed entry to: **MXL microphones competition** July 2003, *Sound On Sound*, Media House, Trafalgar Way, Bar Hill, Cambridge CB3 8SQ, England.

WRH

the small print

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

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Apple 17-inch Powerbook

Laptop Computer

With the largest screen ever featured on a portable computer, Apple's flagship laptop would be tempting even if it didn't offer lots of musician-friendly features such as versatile connectivity and quiet operation...

Photos courtesy of Apple

Mike Watkinson

When Steve Jobs announced that 2003 would be the "year of the notebook" at January's Mac Expo, he showcased two new products to underline his point. The 12-inch Powerbook, as reviewed in May's Apple Notes, is a great solution for those who demand power and portability, but it was the 17-inch Powerbook that drew the loudest cheers, proving that size still matters in this age of increasing miniaturisation. It's taken a while for production models to become available, but now they're finally here. So what do they offer the musician?

New Dimensions

The much-hyped screen is the same unit as fitted to the 17-inch iMac, and is currently the largest available on any portable computer, stretching the term 'notebook' somewhat! The physical dimensions of the Powerbook (at 392 x 259mm) are 18mm deeper and 51mm wider than the 15-inch model, making it quite an armful to carry around the office or studio. However, the thickness remains fixed at 26mm, and at 3.1kg, its weight is at the lower end of the scale for 'super' notebooks such as the Sony GRV600, the previous holder of the largest-screen-on-a-notebook crown.

When you first open the Powerbook the screen does not initially seem as impressive as you might think. The styling is subtle with the alloy finish helping to smooth sharp edges, and there is a very narrow screen surround. Even the hinge is specially designed to minimise the unit's height with the screen open. For those who travel by air on a regular basis this could be a crucial design point: competitors such as Dell's Latitude D800 open a full 40mm higher, despite having a smaller screen, which could be crucial if the seat in front of you is reclining!

As soon as you boot up, however, you become aware that this is something special; and watching a DVD in full widescreen, with speakers far enough apart to give an impression of stereo field, on a machine that can slip into a briefcase could become a landmark in computing. If that is not enough to impress your mates,

> **SOUND ON SOUND**
>
> **Apple** 17-inch Powerbook
> **£2599**
>
> **pros**
> - The screen.
> - Wide range of connectivity, including two separate Firewire ports.
> - High performance.
> - Quiet.
>
> **cons**
> - Possibly a little large to be used as a notebook while in transit.
> - Expensive.
>
> **summary**
> Laptop computers don't come bigger or better than this.

then how about the keyboard backlight, which automatically fades in as the ambient light dims...

On The Inside

Apple's claim that this is a replacement for desktop computers is supported by the fact that the 1GHz processor is the same as that found in the bottom-of-the-range Power Mac. Similarly, the processor is supported by 256k Level 2 cache and 1MB of DDR Level 3 cache. The motherboard clock speed matches the faster Power Macs, moreover, at 167MHz, as does the memory specification. This model is supplied with 512MB of PC2700 RAM running at 333MHz, and with both memory slots full the total memory supported is 1GB.

The optical drive on the Powerbook is a slot-loading Superdrive, but be aware that not all Superdrives are the same: this one is a DVD-R (and CD-R/W) drive, but not a DVD-R/W as in the Power Macs. Graphics are provided by an NVIDIA GeForce 4 440 Go with AGP 4x support and 64MB of DDR SDRAM. This will support a 1440 x 900 pixel resolution on the built-in display and up to 2048 x 1536 pixels on an external display, both at millions of colours. Equipped as it is with S-Video and DVI Video sockets, this machine will be an object of desire for those who wish to cut a dash when making presentations.

"I would personally feel a little ostentatious opening the thing on the 7.35 to St. Pancras..."

The Outside World

Communication with the outside world is possible both with and without wires. Bluetooth and Airport Extreme are fitted as standard: Bluetooth can allow you to use an appropriately enabled mobile phone as a modem, for example, while Airport Extreme is the Apple's name for 802.11g, a so-far unratified development of the standard 802.11b 'WiFi' protocol called Airport by Apple but also adopted by many other manufacturers and popping up in offices, cafes and airport lounges all over the place. Apple have taken a small gamble with the adoption of this new protocol since there is a potential competitor, imaginatively called 802.11a, which delivers 54Mbits/s compared to 802.11b's 11Mbits/s, but this does not have 802.11g's advantage of being backwardly compatible with the earlier protocol.

If you are still using wires the Powerbook has its sockets arranged down both sides, as opposed to on the back, as with the 15-inch model. While the Titanium iBook's rear panel is an elegant solution once everything is plugged in, access is inconvenient, which can be a problem on a computer whose main role is to be portable, so I welcome this change. Along the left-hand edge are the sockets for power, modem, one USB, Type II PC card slot, audio line in and headphone sockets. Along the right-hand edge are sockets for the second USB, Firewire 400 and 800, Gigabit Ethernet, S-Video and DVI video sockets. The inclusion of two Firewire sockets is a welcome addition, especially since they are on separate busses, which should improve performance and stability for those using Firewire hard drives and audio interfaces. The Firewire 800 socket is also backwards compatible with Firewire 400 using the appropriate cable.

Table 1: Plug-in Counts

The table shows a range of Apple computers and the number of mono *PlatinumVerbs* available at two sample rates, before *Logic* reported a 'processor overload' warning.

	44.1kHz	96kHz
Power Mac 1.42GHz Dual	39	17
Power Mac 1.25GHz Dual	35	
Powerbook 17-inch 1GHz	28	12
Power Mac 800MHz Dual	23	
Powerbook 12-inch 867MHz	22	7
Powerbook 667MHz	16	
iBook 800MHz	14	5
iMac 800MHz	13	
iBook 600MHz	10	

Table 2: Track Counts

The table shows the number of continuous 24-bit 96kHz tracks playable before *Logic* reported a 'disk too slow' warning.

Lacie d2 800/Power Mac 1.42GHZ Dual	48
Lacie d2 800/Powerbook 17-inch 1GHz	39
Power Mac 1.42GHz Dual internal drive	38
Powerbook 17-inch 1GHZ internal drive	27
Powerbook 12-inch 867MHz internal drive	23
iBook 800MHz internal drive	18

Performance

So how does Apple's claim that the 17-inch Powerbook is a desktop replacement work out in terms of performance? Like all 2003 models apart from the lower two in the eMac range, it boots only into OS X, in this case Mac OS 10.2.6. I chose to carry out tests with Emagic *Logic* 6.1 and a MOTU 896 Firewire audio interface, for the sake of comparison with previous test results. In all tests, *Logic* was set with I/O buffer size at 512, Process Buffer Range set to Large and Larger Disk Buffer On.

Comparing the number of *PlatinumVerb*s available shows the Powerbook performing exactly as you might predict from its processor speed. Dual processors in the two Power Macs allow them to rise above the trend, but not as much as you might expect. The performance of the Powerbook's Fujitsu internal hard drive is very respectable considering it is a portable drive spinning at 4200rpm, and although beaten by the Seagate Barracuda ATA V system drive in the Power Mac, it is not in any way disgraced. Testing with the Lacie d2 Firewire 800 drive gave better results in both computers, and it is interesting to note that Firewire 800 on a Power Mac gives a different level of performance to its equivalent on the Powerbook.

Battery life is claimed by Apple to be 4.5 hours, which seems reasonable. I ran the battery down from full charge in 2 hours 43 minutes by playing a CD on loop — an activity with high power consumption — so expect something in between for general audio use.

Conclusions

The figures above show that the Powerbook is performing at levels that were previously the domain of its bulky stablemates, and this was a surprise for me, since up to now it has been fairly easy to differentiate between desktop machines and notebooks. The former you kept in the studio, the latter you took on the road, but suffered all the inherent compromises. Now the distinction is less easy to draw, as Apple's latest Powerbook performs well enough to be considered as the only computer you might need. I would personally feel a little ostentatious opening the thing on the 7.35 to St. Pancras, but the display is large enough for serious editing and the backlit keys are actually really useful in dim studio lighting. It makes hardly any noise — the hard drive whine is slightly louder than on an iBook, but fan noise went unnoticed, the larger enclosure dissipating heat far more efficiently than the 12-inch version, which develops noticeable hot spots. With the right choice of peripherals it will give performance in Power Mac territory and it looks a million dollars... SOS

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Recording A Jazz Band On Location

We follow a location recording project from start to finish, showing you how to plan for and conduct the session, as well as how to avoid common pitfalls.

Hugh Robjohns

I was recently asked to do a recording for a student jazz band from Cheltenham College, and it serves as a good example of how to conduct a complete location recording project. The band, called Jig, play a variety of mainly popular/MOR music, and the aim was to produce a CD album which could be sold to students and their parents for fund-raising purposes. The line-up included drums, electric bass, electric guitar, piano/keyboard, three saxes, two trumpets and up to four vocalists, all led by a music teacher called Gill Mew.

Recording Philosophies

I do not operate a conventional recording studio at home, because I always record on location, although I obviously post-produce the recorded material in what is probably best described as an editing/mastering suite in part of my home. Consequently, I consider it my role while recording to capture the best possible performances of the musicians, with the highest possible technical quality. However, I often also have to don a producer's hat, choosing a suitable venue to record in and then deciding on the most pragmatic way to achieve a sound character and style appropriate to the project — in such cases, it's crucially important to discuss with the client what their expectations are.

In some circumstances a multitrack approach is required, with the ability subsequently to replace/add parts, apply pitch-correction, adjust timing discrepancies, and so on. However, in this particular project, the concept was to capture the excitement of a live performance (albeit without an audience). Another important consideration was the fact that the musicians were capable students, not professionals. It would be unrealistic and unconstructive to expect perfect performances, and although it is technically possible to create near-perfection using a multitrack technique and a lot of post-production time and resources, this would have been unrepresentative of the true ability of the musicians involved which, while not perfect, was still very impressive given their age and experience.

Knowing that your every note is being captured can be quite a stressful experience to those not familiar with recording. The band often perform live in concerts to their fellow students and on school tours, and we wanted them to feel comfortable during the recording, so the approach we decided to adopt was to record the band straight to stereo and more or less 'as live' in a large auditorium at the school. The goal was to produce an album which recreated the impression of sitting in front of the band during a great performance, and for the students to enjoy the experience of making a recording they could be proud of.

The GML mic preamp at the top of the rack was used for the vocal mics, whereas the processor below it was used to control and decode the Soundfield mic above the drum kit. At the bottom of the rack is a Lexicon PCM90 reverb processor which was used for vocal reverb. A DK Audio MSD600M meter sits atop the rack.

Planning The Project

I like to have some idea of what I'm getting into before rigging the equipment, so although I was already broadly familiar with Jig's style and the quality of Cheltenham College's musicians, I arranged to sit in on a rehearsal to familiarise myself with their repertoire and performance. Unfortunately school holidays contrived to make the only practical date for this audition the Friday night before the recording, which was scheduled for the Sunday. That didn't leave much time for detailed planning, but, as this was a relatively small project, the time scale worked well enough.

With everyone squeezed into a modest rehearsal room (I wish I had taken some ear plugs with me!), the band rehearsed each number and I was able to take notes on what the different tracks were, how they started and ended, when the solos were and who took them, what the line up changes were, and so on. For example, there were various combinations of alto, tenor and baritone saxes on different songs, and lead vocals were provided by three different singers. These notes then formed the basis of my detailed technical planning, as well as acting as an *aide memoir* during the recording.

After the rehearsal I spent a few minutes discussing with the conductor the order in which we would record the tracks. When there are brass players involved you have to give serious consideration to the order in which tracks are recorded, as their lips can tire easily — especially with amateur musicians. In general, I have found it best to record the hardest and most challenging tracks for the brass just before lunch. That way the musicians are warmed up well from the morning's recordings, but their lips have not become too fatigued yet.

I spent part of Saturday planning the detail of the rig and programming the sound desk — essential work to minimise rigging time on the Sunday, which would be short enough anyway! Most of my recording involves choral and small classical works, and I normally use a Mackie 1402VLZpro analogue desk, which is more than adequate if used carefully, and very portable. However, this job was going to need more resources than that desk could provide.

When I recorded Jig three years ago I used a Yamaha 03D digital mixer, but I sold that a while ago in anticipation of acquiring a DM1000 or 01V96 (I have yet to decide which). Unfortunately, Yamaha had not released either of those desks to market in time for this recording, but the company was able to help me out by generously loaning me an 02R96. I reviewed this desk a few months ago, and I was delighted to be able to use it on a project like this. However, I was concerned about its physical size, since I had to transport all the equipment down to Cheltenham in the back of my estate car, so it was going to be a tight squeeze!

Doing The Paperwork

I started my technical planning by working out a rigging list, deciding how I was going to mic up the various sources, and then allocating mics, mic stands, multicore cable circuits and desk channels. In this case I was only dealing with around twenty external sources, but taking the time to plan and allocate everything carefully — and plot it all on paper — always pays dividends in the long run. It avoids running into last-minute problems like not having enough stands or cables of a particular type, and helps to focus the mind on using the available resources to best effect. Also, by duplicating the rigging lists, several people can work on different aspects of the rig in the certain knowledge that everything is being done to a common plan.

My rigging list is a simple table with seven columns. The first four apply to the recording floor, and the last three to the control room. The first column lists each source on separate rows (kick drum, bass guitar, and so on), the next allocates the appropriate mic stand, and the third the microphone or DI box. The fourth column lists the multicore channel allocation for each source. When recording on location with tight deadlines it is always a good idea to build in a degree of redundancy if you can. In this case, I was able to leave one channel on the multicore spare so that, if something failed, or if I needed to add an extra mic for something, I could accommodate it simply and quickly.

The remaining columns in the list apply to the control-room end of the signal path, allocating each multicore channel to a desk input and, because I was using a digital desk with an input router, which physical inputs are allocated to which desk channels. The last column is for comments relating to each source. In this case I also included the desk's four internal stereo effects as well as my own Lexicon PCM90 as sources on the list, because I needed to allocate inputs and desk channels for these as well. The 02R96 has 24 physical channel faders, with the familiar layering system to access the additional channels. As I was intending to record this session as a live stereo mix, rather than using multitrack, I required immediate access to all channels, rather than having to continually swap fader pages to access the effects returns, for example.

Setting Up The Desk

With just eighteen sources from the studio floor and some intelligent handling of stereo

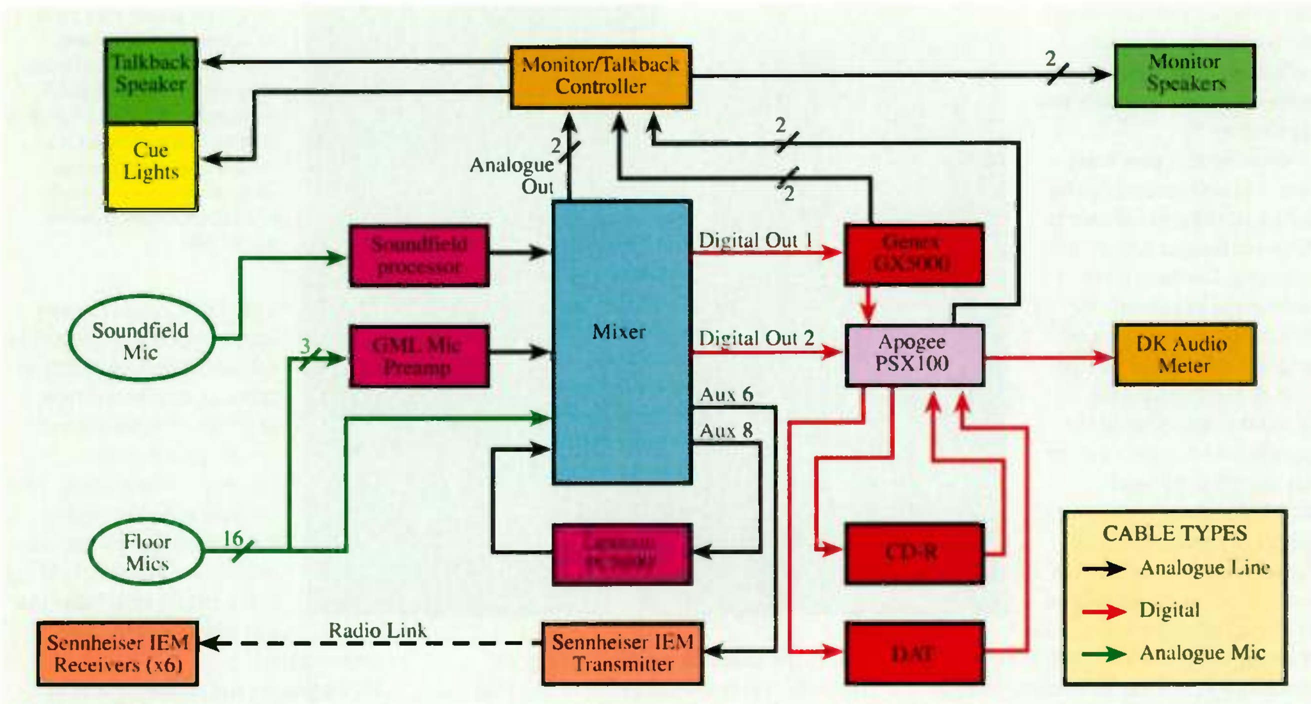

Figure 1. A schematic representation of the rig used for the location recording.

sources, I was able to configure the desk to accommodate all 18 input channels and the five effects returns on a single fader layer, although it took some careful planning and an hour or so of programming the 02R96. I used the desk's vertical pairing facility to provide single-fader stereo channels for the stereo drum overheads, stereo piano mics, and all the stereo effects returns. The downside of this approach is that it makes the input signal routing more complex, which is why the rigging list is so important. For example, the stereo piano mics were allocated to multicore channels six and seven, desk inputs seven and eight, and desk faders/channels seven and thirty one! Clearly, there is a great deal of scope for confusion unless everything is carefully documented. To further reduce confusion on the day, I also attached scribble tape above the faders and below the analogue input gain controls, and marked them up with the corresponding channels. Marking the input gain controls was essential, since they did not always correspond to the fader channels with the same numbers.

I mentioned re-allocating the 02R96's four internal stereo effects processor outputs just now. This is because I wanted to be able to adjust the effects return levels immediately, sometimes riding them along with the source channels, so I reallocated these four returns to faders 20 to 24. I also allocated the Aux sends so that the first four fed the four internal effects processors and aux eight fed the Lexicon. (I only chose aux eight because my Lexicon input cable wasn't very long and the aux eight output was closest to the edge of the desk!) I then configured the internal effects processors to provide a set of generic effects based on the Hall reverb, Plate reverb, Ambience, and Delay algorithms respectively — these could be applied quickly and easily to the various sources as appropriate for each track. All these settings were stored in one of the desk's snapshot memories and, with the desk scribble strips already marked up, all I had to do when rigging was plug everything in as documented on the list and turn it all on, keeping my fingers crossed at the same time!

The final stage of the rigging list was to draw out a block diagram of the equipment and how it would all hang together, documenting all the specific cables required to hook everything together, the mains leads and distribution boards, digital cables, clock leads, and so on. This helps to make sure that everything gets loaded into the car, both on the way there and on the way back. All the paper work might sound over the top, but I keep copies of everything on my computer, so in reality it takes very little time to recall a file from a similar previous project and modify it accordingly, and it provides a lot of confidence that everything has been taken care of.

Preparing The Ground

The recording session was slated to start promptly at 11 o'clock on the Sunday, but it was obvious that it would be a challenge to load, travel, unload, rig and test everything by that time without getting up at the crack of dawn... and I'm not good with mornings at the best of times! So, to ease the load a little, my 'delightful assistant' on this occasion Paul Hedges (a senior trainer at the BBC's training and development facility in Evesham) and I went down to pre-rig the cables and set up the recording floor on the Saturday afternoon. The recording venue was a large rectangular auditorium with a raked seating area and a high stage. It was reverberant, but not excessively or unpleasantly so, and the acoustics could be controlled to a degree by closing curtains around the sides and rear of the hall, as well as on the high and deep stage.

An adjoining room at the back of the hall served as a control room, being far enough away from the performance area to enable good monitoring isolation. This room was long and narrow, with bare plastered walls, a high ceiling and a hard floor, so was also rather more reverberant than would have been ideal. We decided the best solution was to work across the short axis of the room, keeping the monitor speakers fairly close, and to hang duvets on the walls to reduce reflections as much as possible.

The cable run from control room to studio floor was around 45 metres in total, including running the cables carefully around the walls to avoid having to cross passageways (and thus reduce the risk of anyone tripping over a cable). We installed three cables: a 16-way multicore, a dedicated cable for the Soundfield mic, and a three-circuit multicore cable used for my bespoke communications system (providing


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talkback, cue lights, and a closed-circuit telephone, although the latter was not needed on this occasion).

A Genex GX8500 was used as the master recorder, with a DAT machine providing backup. The Apogee PSX100 converter provided digital routing to the DAT and CD-R machines, as well as acting as an A-D for playback.

Having rigged these cables we then set up the mic stands in the approximate positions for each performer, and rigged the cables between the multicores and stands. We set up a pair of heavy-duty mic stands around the drum position, one to support the Soundfield mic I intended to use for the stereo overheads, and the other to enable a duvet to be suspended between the two stands to provide some isolation around the mic in case spill became a problem, although I didn't actually make use of this provision in the end. We also set up a large wooden baffle (a piece of scenery borrowed from the stage) to provide some additional screening for the upright piano — or more accurately, for the microphones placed behind it — since I was concerned that these mics could pick up a lot of spill.

The positioning of the musicians was designed to maintain good sight lines between them and the conductor, while also maximising separation for the microphones. The drums were positioned close to the stage and in such a way that the (effective) overhead mics presented their least sensitive axes towards the brass section and guitar amps. The vocalists were positioned on the stage behind and above the brass and drums, with the curtains closed behind them to help prevent reflections getting into the front of the vocal mics. The small bass, guitar and keyboard amps were set to as low a volume as was practicable, again to minimise the spill reaching other mics.

This whole pre-rig took about an hour, including some time trying to locate suitable baffle materials — but this effectively saved us an hour of frantic work on Sunday morning, so was well worth while.

And On The Seventh Day...

Come the Sunday, we arrived at the hall at 9:30, my car creaking with the strain of transporting all the control-room equipment. We quickly unloaded everything and Paul started rigging the microphones, talkback speaker and cue lights on the recording floor, while I started setting up the control-room equipment.

Having placed the loan 02R96 desk on a table in the centre of the room, I set up the recording equipment on the left-hand side, and the outboard rack and odds and ends on the right. The larger of the two recorder racks contained a Genex GX8500 hard disk recorder, an Apogee PSX100 converter, an HHB CDR850 CD burner and a Panasonic SV3800 DAT machine. I generally use the DAT as a backup recorder, and the CD burner to provide the client with a disc of the raw recording to listen to after the session. This not only gives a great feel-good factor, but it also means the client can make preliminary editing decisions in their own time, and reduces the need to sit through endless playbacks at the end of the session when all you really want to do is tear the gear down and get home! When I'm working with the Mackie analogue desk, the Apogee acts as my main A-D for the digital recorders, but in this case I was using it more as a digital input signal router and monitoring D-A.

The basic system wiring can be seen in Figure 1, with the Genex receiving a 24-bit 44.1kHz digital output from the desk. A second digital output was connected to the Apogee which redithered the signal to 16 bits and distributed it to the CD-R and DAT. The digital outputs from the Genex, CD-R and DAT were all routed back to the Apogee where they could be selected to feed the D-A converter, the analogue outputs of which were hooked up to my custom-built monitoring unit. An auxiliary output from the Apogee passed whatever was selected to the D-A input across to my DK Audio MSD600M metering system. This meter was vital, as the desk was not supplied with the optional meterbridge. For those not familiar with this meter, it provides a phase meter, bar-graph level metering (with various scales and ballistics), and a 'goniometer' which provides a lot of real-time information about the spatial nature of the stereo soundstage. The unit also incorporates third-octave and FFT spectrum analysers and a test tone generator —

The Soundfield mic was set up in place of drum overheads, while the snare drum was close-miked using a Neumann KM185 hypercardioid mic.

really useful and informative tools that I couldn't work without!

The outboard rack positioned to the right of the desk contained a four-channel GML mic preamp which I used for the vocal mics, the Soundfield mic processor, and the Lexicon PCM90 which I used mainly for the vocal reverbs. The MSD600M meter was positioned on top of the rack, and the monitoring unit was positioned to the right-hand side. This monitoring/talkback unit was custom made for me by Audix Broadcast a few years ago and is based on the standard monitoring controls found on most BBC-specified consoles. You might think that makes me an anorak wearer — and I often am — but the design includes a lot of important facilities that most console monitoring sections leave out. A row of four expensive but very ergonomic lever-key switches provide input selection (desk output plus two external machines), cut-left/cut-right (helpful when trying to isolate odd noises or faults), mono-to-left/mono-to-centre (for accurate mono-compatibility checking and to set the balance control), and polarity reverse/dim controls. With all the switches in their centre positions you have normal stereo monitoring of the desk output. There is also a balance control and a volume control, both with centre detents at the default settings — the latter defining a reference monitoring level to help maintain consistent mixes.

The unit also contains red and green cue light switches, and a built-in talkback system arranged so that operating the talkback switch also dims the loudspeakers to prevent howlrounds and to make the talkback easier to understand. There is also room in the box to incorporate remote transport controls for the Genex recorder, which will be my next project!

With the control room up and running, I played a CD-R of various test signals and familiar music to make sure the monitoring and metering were all working correctly. I then routed this source through the desk to check the output was getting back to all the recorders correctly, and to test the reverb send to the Lexicon. At least with digital signals you don't have to worry about left-right reversals and incorrect gain structures anymore! Finally, I faded up each channel on the desk and roughly adjusted the input gains for each mic, making sure all the sources were present and clean.

It would be standard practice at this stage to 'scratch' each mic in turn, making sure each source is what it is supposed to be and that all the cables are working properly. However, because all the equipment (bar the console and bass drum mic) was my own, and we had rigged everything according to my beloved list, I was confident that this wasn't necessary — if there was a problem we'd find out soon enough when we started to build up the first balance.

The bass player and guitarist were set up close to the drummer to provide good communication between the players. Bass guitar was recorded through an active DI to allow heavy compression without spill becoming a problem, while both DI and miking were experimented with for the guitar. The bass, guitar and keyboard amps were all kept as low in volume as possible to minimise spill problems.

Mic Selections

I had recorded Jig once before, and had used a Soundfield mic to provide a stereo pickup for the entire brass section, which worked well. However, this approach relies on the section having a good ensemble balance, and I was concerned that this might not be the case on this occasion — especially since the saxophonists changed instruments for different tracks. Instead, I decided to use separate mics for each brass instrument. In turn, this decision influenced the allocation of the Soundfield mic for the drum overheads, principally because this mic uses its own cable, which released two multicore channels for other purposes. The other advantage of using the Soundfield in this role is that the stereo width and the effective angle of its 'virtual mics' can be adjusted in the control room, making it relatively easy to optimise the settings to minimise the amount of spill it captures.

My choice of kick drum mic was dictated partly by the arrival of an Audio Technica AT2500 for review. I had seen this mic at various trade shows and was intrigued with its combination of separate moving coil and electret capsules in one body. The output from each capsule is presented on separate XLRs enabling the engineer to balance the weight of the dynamic element with the snap of the electret. It seemed to be a very flexible approach and I was keen to see how it worked in practice. The other thing in its favour is that both capsules exhibit a fairly flat frequency response. A lot of bespoke

Reliable Multicore Systems

My multicore cables are slightly unusual in that I don't use end boxes at all — I have found them generally unreliable and very difficult to repair on location. Instead, my cables are broken out into tails, terminated with XLRs. On the smaller multicores (I have two 25-metre eight-way cables) I used colour-coded strain relief collars to identify the channels, but with only ten colours available that option was not suitable for the 16-way cable. I subsequently discovered that Canford Audio offer an engraving service which costs little more than the coloured collars, so I asked them to supply me with engraved XLR connectors numbered 1-16, which look very neat and are easy to read.

The usual problem with multicore tails is that they tangle themselves into a huge knot whenever you look away! My cheap solution to this annoying (and damaging) problem is to use spiral wrap, with just a couple of inches of each tail emerging after a couple of turns of the wrap, spaced to correspond with the typical layout of XLR sockets on most sound desks. Thus, plugging one end to a desk is simplicity itself, and the other end is just as easy to plug up to separate cables to the appropriate mics.

This arrangement has several advantages, including no heavy box on the cable, and no cramped multi-pin plugs to damage. I have suffered all manner of problems in the past with more conventional stage box/multi-pin plug arrangements — almost all of which were impossible to repair on location, and often the damage affected more than one channel too. In contrast, I have yet to suffer a failure with my tails system, but should the worst happen the likelihood is that any failure would most likely be restricted to a single channel. I am also confident that repairing an XLR on location would be quick and easy, even without a soldering iron if necessary — it's amazing what can be done with a piece of Blu-Tak!


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kick drum mics have enormous frequency response peaks to create an 'instant' rock-oriented kick drum sound. This approach can save a lot of work if that is the sound you want, but is rather restrictive if you need something more natural. The only fundamental problem with the AT2500, of course, is that it requires two multicore and desk channels instead of just one...

I allocated a Neumann KM185 for the snare drum, which is a small-diaphragm hypercardioid condenser mic. I find this works quite well in the role, and the deep response nulls at 120 degrees can be used to good effect in rejecting the hi-hats by simply angling the mic appropriately.

I used a Canford Audio active DI for the bass guitar (powered via 48V phantom). I usually find it necessary to apply quite a lot of compression to the bass, which can pull up a lot of spill if you mike the bass amp — DI'ing gives a far cleaner signal. I left my options open on how to deal with the electric guitar, allocating an EMO passive DI box to interface with the amp output, as well as a CAD M179 condenser mic on a banquet stand in front of the cabinet. I planned to use one or the other (but not both because I was short on multicore channels) deciding after a chat with the guitarist on Sunday morning — it turned out that there were effects coming from his amp which were best picked up using a mike. The keyboard player used either the upright piano or a Roland XP30 on different tracks, so I allocated another pair of DI boxes for the keyboard, and a pair of Sennheiser MKH40 cardioid capacitor mics for the piano. I planned to swap the cables between mics and DI boxes for the different numbers. The Roland keyboard was being used to simulate a jazz organ sound, but the simulation lacked realism, which was a shame because the keyboard parts featured heavily in the two tracks it was used on — one a classic JTQ track. Fortunately, Paul Hedges came to the rescue by bringing along his son's Oberheim OB3 organ module, which we hooked up to the MIDI output from the Roland keyboard. After a couple of minutes tuition on how to use the drawbars and Leslie speed switch, the keyboard player was well away, and the recording certainly benefited from this far more realistic Hammond sound.

For the brass section I used Microtech Gefell M930 mics for the two trumpets (these mics have bags of headroom and a nice, large-diaphragm sound quality), a pair of AKG C414s for two of the saxophones, and a Neumann TLM103 for the third sax player, who also played baritone on one track.

The vocals were miked up with two Audio Technica AT4040s and another Neumann TLM103, all three being cardioid mics supported in shockmounts and fitted with pop shields — the shockmounts were important because of the wooden stage floor. When all four vocalists sang at the same time, the two backing singers shared one mic, and the two main lead vocalists used the others. I have found that the TLM103 often suits male vocalists slightly better than females, and the AT4040 is often the reverse, so they complement each other well.

Whereas the instrumentalists all made plenty of noise and could hear each other without problems, the vocalists didn't and couldn't! Foldback over floor monitors would have exacerbated the spill problems, so we really needed to provide a cue mix on headphones. This particular problem was solved with the loan of a Sennheiser wireless monitoring system courtesy of the BBC. The transmitter was located in the control room and fed from Aux 6 on the desk, providing a straightforward post-fade mix of the vocal mics plus the Lexicon reverb. Each vocalist clipped a receiver to their belt and wore a pair of Sony closed-backed headphones — usually keeping one side behind an ear to be able to hear the rest of the band - and could adjust their own listening volume. We had six

A piece of stage scenery was used as an impromptu screen to increase separation for the piano mics. However, this made the sound quite boxy, so some corrective EQ was required at the console.

Riding The Mix

When mixing live sound (such as for live broadcast or front-of-house PA) if you make a mistake it's too late — the moment has passed and you have to concentrate on getting the rest as good as you can. However, when making recordings things are slightly different, even if recording 'as live', as we were here. The reason is that, in all probability, you will be editing different takes together to form the best composite that you can. However, that will only work if the mixes remain substantially the same for each take.

That's not actually as hard as it sounds, but it does mean that you have, at the very least, to get the rhythm section well balanced from the first take, since generally their contribution is constant throughout a track. Lead instruments and vocals may come and go, and you usually have to ride the faders accordingly, but mistakes here are easier to correct with editing, provided that the underlying rhythm elements are near identical in all takes.

When it comes to pulling up solos and lead lines, life is much easier if you can see the musicians, because you can observe their body language and anticipate the entry. However, in this particular case we were mixing in a room without sight lines at all. I've been thinking about acquiring a simple closed-circuit TV system to try to overcome this — perhaps a cheap security camera system — but I have yet to do so. Fortunately Jig's music was all very familiar, so by and large we were able to anticipate where each solo would come in from memory, and the notes I made during the rehearsal session told us who would be taking each solo. Most times we guessed right first time, and on the few that we messed up we were able to do a retake.

Where instruments played solo lines, we changed the balance as necessary, pulling the appropriate fader up maybe 5-6dB, and returning it to its normal mix position afterwards. Similarly, we often pushed the vocal mics down by around 4-5dB when the singers weren't singing. This is an important point, because we were compressing the vocal mics, and with no input signal the compressor's make-up gain would effectively 'suck up' more spill. By pushing the vocal mic faders down by a similar amount to the typical amount of gain reduction when the vocalists were performing, the level of spill remained more or less constant. However, other than when a particular source wasn't being used at all in the recording, we didn't ever close any faders. In a relatively live environment such as the one we were recording in, closing a mic completely would change the ambience and that would make editing difficult again, as well as being a distraction during the track.

Although some tracks would require fades in the final CD, we recorded everything at a constant level to leave options open for the editing stage, and we always recorded about ten seconds of silent ambience before and after each take. The reason for this was just in case we decided to run the quiet hall acoustic between tracks on the finished CD instead of digital silence. When a recording is made in a lively environment like this one, the disparity between the track ambience and 'digital black' can be a little distracting sometimes.

receivers, so the conductor and guitarist wore systems too.

Take One

The band took their positions and started to warm up, playing through a number which was moderately challenging and involved the entire band, while Paul and I started to work on getting a basic balance. This is always the hardest part, as you are working in unfamiliar monitoring conditions, and have a lot of raw sources to deal with, sorting out gains, equalisation, dynamics and effects, with the clock ticking relentlessly.

After a couple of run-throughs we were about 80 percent of the way there with the mix, but there was still some fine-tuning to do. We were compressing the bass guitar quite heavily, along with more modest compression on the vocals and piano, the latter also requiring some EQ to help compensate for the slightly boxy effect of that baffle board enclosing the mics. At this stage you can either carry on trying to perfect the mix and risk boring the musicians, or start recording to get the session under way, knowing that the first take won't be the best from an engineering point of view. Experience has taught me that the latter approach is the best one, and so we went for a first take at 11:30.

This first take certainly wasn't the best any of us could achieve, but might provide useful material for editing and at least helped the band to overcome the nervousness of being recorded. The second take was a lot better, particularly from the mixing perspective, but was perhaps not the best that the band could achieve. However, rather than slog through it again we decided to move on and return to it later.

Individual mics were used for each of the members of the brass section: Microtech Gefell M930s on the trumpets, AKG C414s on the top two saxophones, and a Neumann TLM103 for the third sax player, who doubled on baritone sax.

The next track came together very well — it's amazing how that red light really focuses everyone's attention! — and we had everything we needed in two takes. By now we were heading towards lunchtime so we decided to have a stab at the most challenging track for the brass. The first take was pretty good, but there were a couple of sections where things could have been tighter, so we did a few retakes to improve those sections. Finally, we went for another complete take which worked very well indeed. At this stage we called everyone into the control room to hear the playbacks and, enthused with the progress so far, we broke for lunch. However, we had spent two hours recording just three tracks, the first of which we knew we had to do again... so we had our work cut out to record another seven tracks (plus the retake) in the afternoon!

Rejuvenated by a meal and an hour's break, we reconvened and launched straight into the remaining tracks, getting most of them in just two takes, although some took three or four attempts. We even had time to go back and have second stabs at a couple of earlier tracks and were rewarded with even better performances. We turned the red light off for the last time just before 17:00, and an excited band convened in the control room to listen to some playbacks. While running through the various takes I was also derigging the unused control room equipment while Paul broke down the recording floor. We had everything derigged and ready to load into the car an hour later, and within two hours we were enjoying a cold beer!

However, there was still work to be done before the project was complete, so next month I'll be discussing how we went about editing, compiling, and mastering the CD, as well as producing the artwork and sending the whole lot off to the pressing plant. SOS

Mixing Or Remixing?

During my audio career I have recorded on just about every format and with pretty much every recognised technique. I have recorded all manner of performances straight down to mono and stereo, as well as spending hours multitracking bands in state-of-the-art studios, with even longer spent remixing the tapes in mono, stereo or surround formats. My experiences have repeatedly lead me to the conclusion that I prefer to record a performance rather than to construct one, and I think the end results are usually the better for this approach. Consequently, I generally record straight to stereo whenever possible, rather than engaging in multitrack recording/remixing. Apart from anything else, this technique is much more exciting and challenging for the engineer, requires slightly less equipment (and a lot less post-production time), and delivers near-finished results immediately.

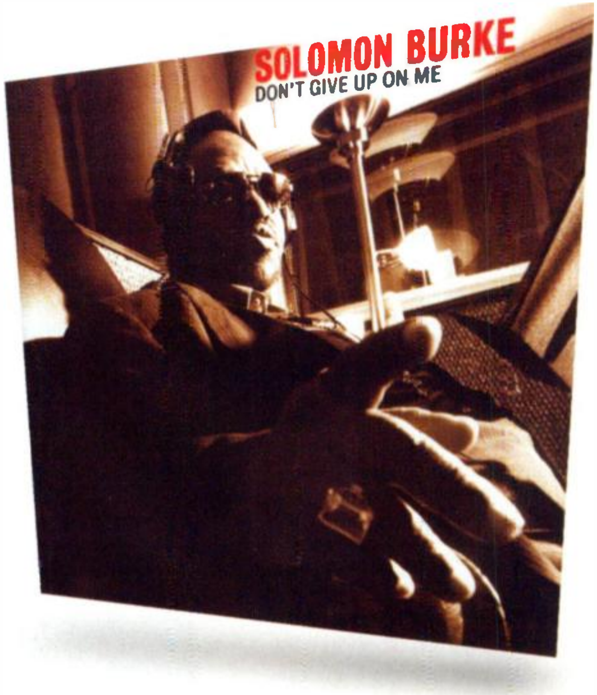
Obviously, there is a time and a place for multitracking, and clearly certain genres of music have to be recorded using multitrack techniques. I'm not dismissive of this technique at all when used appropriately, but I do feel that it is often misused. The vast majority of the work I do now involves recording performances given by talented musicians — and I am trying to capture that performance. Unless the recording is extremely complicated or unpredictable I find there is rarely any need to multitrack the sources, and while it may be possible to produce a slightly more precise remix, in my experience it rarely sounds as exciting or natural as the monitor mix recorded during the performance.

Multitracking also brings procrastination, putting off engineering recording or production decisions because you can fix it in the mix. While this flexibility can be extremely useful in certain situations, it can also cultivate sloppy engineering at the recording stage, and encourage endless fiddling with the minutiae of individual tracks, consuming huge amounts of time often with little material benefit to the final product. Having spent much of my career in the BBC, mixing live is a familiar challenge to me and it is the way I generally prefer to work. Obviously, it doesn't always work out right — I may be late fading up a solo, or a musician may split a note — so I record multiple takes and subsequently edit the final track together using the best elements from each take.

There is a practical issue here about knowing how many takes to record, and really it comes down to experience. You need an awareness of both the technical quality of the mix (whether you can improve upon it as an engineer) and also the quality and accuracy of the musical performance and whether the musicians can do better. It also requires an appreciation of how different takes may be edited together. Another crucial point to bear in mind is that the editing is of mixed tracks, and this places certain constraints on mixing and on what can and cannot be changed between takes. For example, changing the fader levels between takes for the lead instruments or vocals will usually be fine, but if the level of the bass guitar changes significantly then trying to edit two takes together may prove impossible.

S Husky Hoskulds

Recording Solomon Burke's *Don't Give Up On Me*



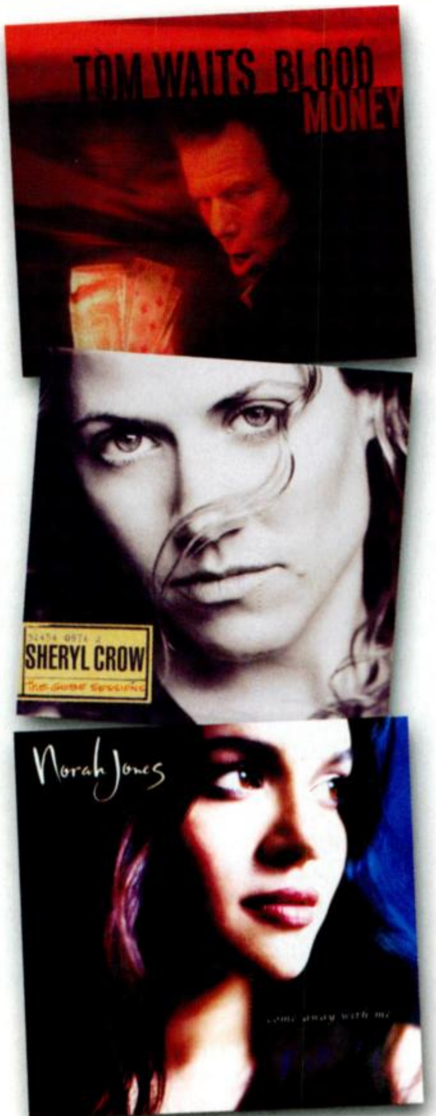
The success of soul legend Solomon Burke's comeback album is a testament to the virtues of keeping things simple in the studio.

Mike Senior

It would be easy to look at the charts and come away with the impression that IT skills are more important to the making of a successful record than musicianship or recording nous. However, bucking this trend at the 2003 Grammy awards was veteran soul singer Solomon Burke's remarkable album *Don't Give Up On Me*, a set of songs with a warm and organic sound miles removed from the over-produced fizziness of many modern CDs. And what will no doubt be particularly edifying to the traditionalists amongst us is that it also took only a week to record and mix in its entirety.

The man responsible for this feat of engineering was S Husky Hoskulds, who cut his teeth working as staff engineer at Hollywood's legendary Sound Factory studios, and whose studio credits include albums by Sheryl Crow, Tom Waits, Norah Jones, Fiona Apple, Aimee Mann, Turin Brakes, The Wallflowers, Vanessa Paradis and Joe Henry. I was keen to know how he and producer Joe Henry managed to get so much done in such a short time. "It was an incredible amount of work," laughs Husky. "We went in and did a marathon. We decided that we'd have four days to record and two days to mix. Looking back at it, I'm amazed that I even made it through, but on the other hand there was something really liberating about doing that. There can't be a lot of chasing your tail, and if you decide that before you start, certain things become really straightforward. There are certain things that you just don't have to worry about. I think that the fact that records are not made like this today is kind of a

S Husky Hoskulds at the API desk in the Sound Factory.



Hoskulds' work can also be heard on some of the tracks from Norah Jones' best-selling *Come Away With Me*, while other credits include Sheryl Crow and Tom Waits.

drag, because it's a lot of fun not labouring over everything. I worked with Tchad Blake and Mitchell Froom at the Sound Factory for a long time, and they both have the attitude that there's no point in spending five months making a record. If you have an accurate representation of the song, whether there's more or less delay on the vocal isn't ultimately going to matter.

"There seem to be two schools of making records. On the one hand, there's the school which hires people to do what they're really good at doing, and then lets them do that to make the record. On the other hand, there's the school of production where you're more in charge of the whole musical side of things. It's a much more laborious situation and it's four to six months out of your life. Both of these methods can work, although I'd say that making a record in six days is certainly more fun than making a record in six months. If you mix things in a couple of hours, some things are going to be left poking out. It isn't going to be a completely close shave, but that's the character, and I think it's unfortunate that more records aren't made like that. You can get a unique sound if you don't labour over things too much.

"Normally when I work on records like that, I like to come in the day before and set up for a few hours. But for this one, I had to finish a record in New York, then I flew back home to Iceland for a couple of days, and then got back to the Sound Factory at three o'clock in the morning of the Monday that we started, so there was almost zero preparation time. Fortunately, I know the Sound Factory very well — it's easy to do those kinds of records in a studio like that where every scenario you come up against has already happened at some point. I know that I can put the upright bass over here and it'll sound fine, even though it's quite close to the drums, that I can put the upright piano over there, and so on.

"We had drums, upright bass and upright piano in the same room. Then we had the acoustic guitar player off to the side, with the Leslie for the Hammond B3 behind a single door, and Solomon was in his own room. It's a fairly small studio, so Solomon had sight lines to the drummer and the bass player. Having everybody in that proximity makes it feel less like being in a 'recording studio' and more like a bunch of people sitting around together — they can tap each other on the shoulder and communicate with each other."

In The Room

Looking at the album's sleeve notes, the vocal mic looks like a Neumann U87. "I think it's an 87," he agrees, "but it may also be a 67. It was going through a Hardy M1 preamp and into the ADL 1000 compressor — a modern reissue of the LA2A, made by Antony DiMaria Labs. We tried a couple of different mics, but singers like Solomon just start singing and you say, 'Oh, that's excellent.' For some people you have to be really picky with microphones, but with others the least obvious choice just works perfectly. It's a strange thing."

Husky's own web site (www.eightbitaudio.com) includes links to some of the other interviews he's done in the past, and I happened upon a section in one of them where he described the drum mic setup he had been using on a Wallflowers project: Electrovoice RE20 on the kick, Shure SM57 on the snare, Neumann KM84 on the hat, and a Calrec stereo mic for the overheads. Was a similar approach used for *Don't Give Up On Me?* "That was pretty much the setup, more or less — it's kind of the regular setup. Then I'll maybe have a ribbon mic which might be filtered or compressed more. I've had good luck with a ribbon mic sitting behind the drummer at ear level, maybe a couple of feet back from the drummer's head, in the corner of the room. A lot of times the sound is just that ribbon mic and a touch of the kick drum mic. It's amazing how drums sound great if you don't stick a mic right up next to them. To hear a rack tom hit with a Coles ribbon mic on it about four feet up is great — that, to me, is the sound.

"I try to get the drum sound going in the room rather than relying on processing on the way to tape. First of all, Jay Bellerose [*the drummer on the Solomon Burke sessions*] has a pretty amazing collection of broken drums and miscellaneous bits of metal, and I've also got a pretty good collection of snare drums and things like that. Part of the lo-fi feel comes from what he's hitting, but part of it also comes from experimenting with mics and maybe filtering things a bit. I like to mix things around a little bit, so I usually have one fader open for miscellaneous ▶

Real Reverb

Digital reverb processors might seem inseparable from the mixing process to many people, but S Husky Hoskulds is one of an increasing number of engineers who are relying on artificial ambience less and less. "I use delays much more than reverbs as effects. The Watkins delay is my favourite vocal delay — it's all over the Solomon Burke record, and has probably been on half of the vocals I've done since I got it three years ago. It's such an amazing-sounding box! I also use the Line 6 Echo Pro, which is just brilliant. I couldn't care less how close it sounds to the original units, because it's just a delay with 20 different sounds for a few hundred bucks.

"The way I position sounds in the mix has a lot to do with delays, and also a lot to do with the fact that I like to use the rooms while I'm mixing. I'll usually have two small PAs going and usually a couple of speakers as well, so I often have all three rooms at the Sound Factory miked up. All that stuff will come back into the mix through a small submixer at the side of the main console so I have a choice of different sounds: the Calrec in the main room, or a couple of close mics, or a couple of stranger mics on another PA in another room. It's amazing to see how people react to me using the rooms like this. People will ask me how I get my room sound, and the simple answer is: by using the room! There's a lot more character to real rooms, and they don't have that flag on them that shouts 'Reverb!'. Room sound has dimension to it, because a real room has real dimensions."

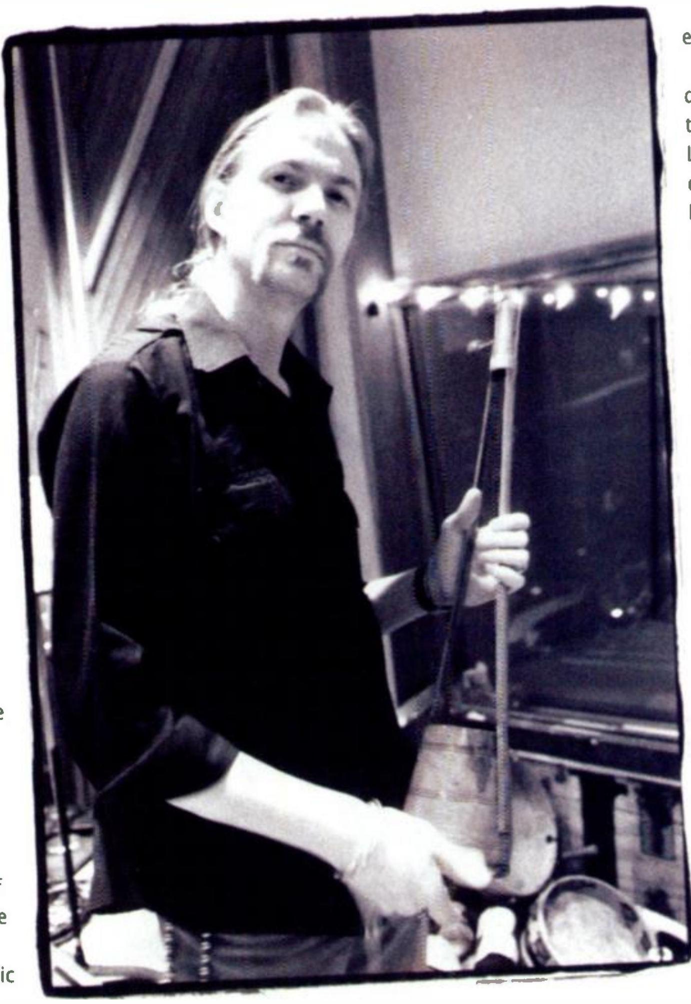
experimentation — an old gramophone horn with a mic in the end of it, a toy mic with a spring inside. I sometimes call it the $100 channel."

Making A Commitment

Although he likes to experiment, Hoskulds' approach to recording is economical. "I often only record drums to three tracks: kick on one track, snare on one track and then a mono overall drum track — if I'm using the Calrec, a pair for it. If I have 10 faders to choose from on the way to tape I'll try to commit to the sound as we're recording. One of the reasons I can mix a song in two hours is because there's nothing that really needs to be discovered in the mix — it's all there on tape. I make notes of what things are working together as I record, and what the panning is, so I don't have to struggle to get things to fit together in the mix.

"Drums and bass will usually get EQ'ed pretty heavily on the way to tape. With electric guitars I'll usually try to stick with what's coming out of the amp — it's usually straight to tape through a compressor — and with acoustic guitars I'll try to move the mic about a bit if I can, because that certainly helps. I don't often EQ vocals on the way to tape. I'll try to pick the right mic and compressor and try to steer the singer closer or further away. I'm not afraid to EQ things after the fact, though, either. You get as far as you can with mic placement and everything else, but then there's still room to EQ.

"Although I set up each track separately by ear, I guess if you looked across the console you'd probably see a theme of 'more bass, less treble'. The mastering guy — Greg Calby usually for my stuff, although Doug Sax did the Solomon Burke record — probably says 'here we go again', pulls out 3dB at 400Hz and adds a couple of decibels from 6kHz upwards because there's nothing there, but it all works out somehow."



One of the aspects of the sound of *Don't Give Up On Me* which I found most interesting was the flattering, larger-than-life compressed sound of the individual instruments and the mix. As I'd suspected, Husky has a secret weapon... "I have an old black-face Al Smart compressor," he reveals. "It was the first one he made, and I think it's called the C1. It's always on the main stereo buss. I usually set it to a ratio of 2:1, and use one of the slowest attack settings. There's just something about the way it sounds — I can't even track without it, because I'm always working towards the mix. It also seems to sound better to me than the silver-faced one, for some reason, even though it's cheaper!

"I'll also usually buss all the drums through a compressor at the mix — same with the bass. Lately I've been doing more extreme stuff, such as on the latest Joe Henry record, where I have a couple of compressors on busses, and then also a couple of compressors in side-chains, and most of the mix will get sent through those. On the Solomon Burke record, I think the buss would have been fairly lightly compressed, although individual channels would have got pretty squashed. I have four Dbx 163s, the '70s ones with the slider and the LED meters, and if I'm in the mood I'll put those on all the guitars or the drums."

Beyond Pro Tools

Despite an obvious fondness for traditional recording techniques and analogue equipment, S Husky Hoskulds is keen to harness the creative power of computers. "In the middle of last year I decided to get a bit more into the whole computer scene — beyond Pro Tools, that is. I don't really do much in the way of editing or fixing — what really interests me is the processing capabilities of computers. For me, the world of VST plug-ins and instruments goes far beyond what's out there for Pro Tools, in terms of price and availability, so for a while I was using Emagic *Logic Platinum* with the Digi 001 hardware, which I'd had since it came out, although I now use the MOTU 24I/O interface. Lately I've even started setting it up so that I have plug-ins going as I'm mixing. I'll take things off the sync heads on the two-inch, run them through a few plug-ins, delay them as far as they need to go to put them in sync again, and then bring them up on the console. I also have the Nord Micromodular, which I've often used in the same way, although I use it less and less now that the VST things are involved. This stuff is so much more interesting than lining snare drums up to a grid or tuning vocals, which I refuse to do. Part of the reason I got the Digi 001 originally was because you couldn't really do multitrack-style editing on it — sorry, can't do it! However, I've got the numbers of guys that will, and in the meantime I'll be over here distorting vocals and stuff..." SOS

Spilling Time

"I like to use spill to my advantage," says Hoskulds. "Unless you're recording in your garage, the sound in the room is usually pretty interesting. I'll go and move things around quite a bit to make the spill work. For example, if I'm recording drums, piano, and acoustic guitar, I'll often run out while people are getting sounds and move the piano mic further away to match the ambience in the guitar mic, and then pan them left and right. That way the piano and guitar mics actually include half of the drum sound. A lot of the organ stuff that you hear on *Don't Give Up On Me* is well set back in the mix, and I got that sound by using the spill from the acoustic guitar mic in the next room. With the mics panned in a certain way you get a kind of 'stretch' across the stereo field. There's no reverb on the track except for what was going on in the room. I'll always try to print the talkback mics as well, and sometimes they'll get used."

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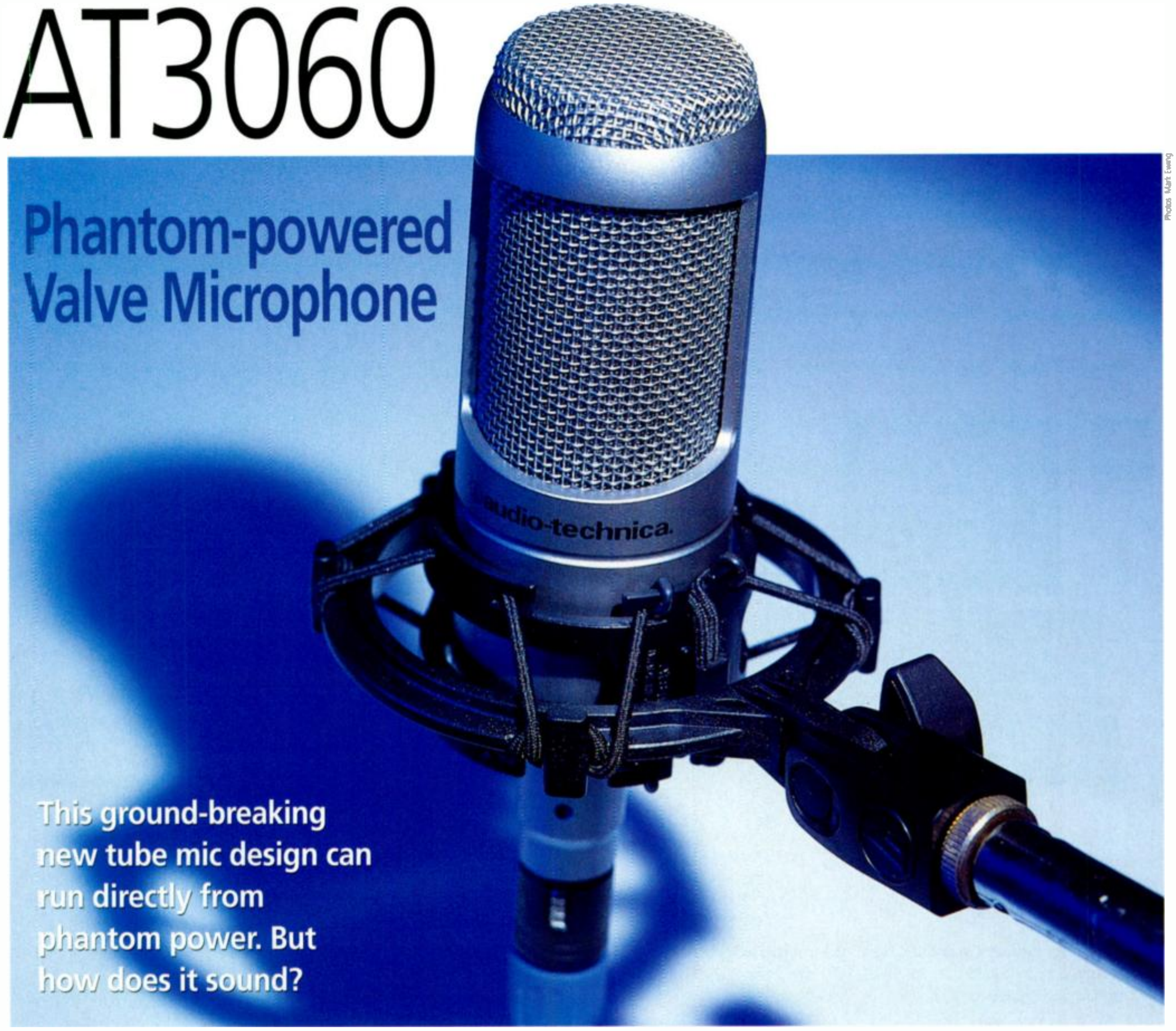
Audio Technica AT3060



Phantom-powered Valve Microphone

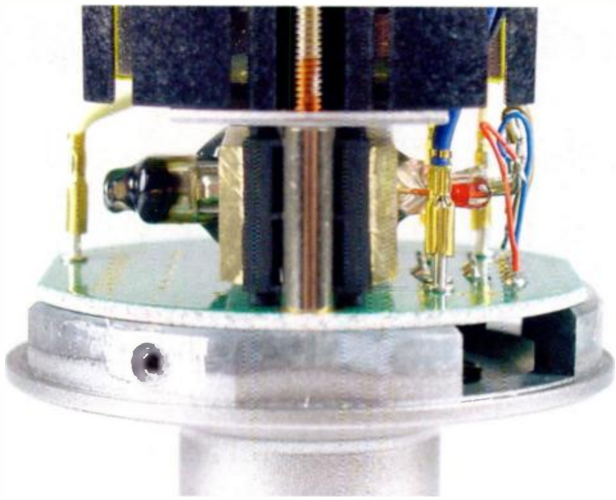
This ground-breaking new tube mic design can run directly from phantom power. But how does it sound?

Paul White

From the outside, the Audio Technica AT3060 looks like a conventional large-diaphragm capacitor microphone, but inside it's a somewhat different story. This is the first tube mic I've encountered that doesn't need a separate power supply — it runs from regular 48V phantom power and draws only 3mA of current. This seems impossible when you consider that conventional tubes require high plate voltages and high heater currents, but the AT3060 uses no conventional tube. It doesn't even use a miniature military-grade Nuvistor tube. Instead it uses a miniature low-current tube developed for use in hearing aids, the aim being to achieve tube warmth at a lower UK price point and without the need for bulky power supplies.

The AT3060 is built around a large-diameter back-electret cardioid capsule not physically dissimilar to that used in other Audio Technica mics such as the AT4033, though its frequency response curve seems to have been deliberately shaped to produce a specific 'vintage' tonal character. While the frequency response is wide, there's a presence hump around 5kHz and the upper response starts to drop away gradually above 10kHz, which makes the official -3dB response 50Hz to 16kHz. At 20kHz, the response is down by around 10dB. Other aspects of the mic's spec are more traditional, with a respectable 25.1mV sensitivity at 1Pa and a maximum SPL handling of 134dB. Electrical noise is reasonably low at 17dBSPL, yielding a dynamic range of 117dB and a signal-to-noise ratio of 77dB at 1Pa (at 1kHz), where 1Pa equates to an SPL of 94dB.

Supplied with a soft pouch and a good-quality shockmount, the AT3060 is well engineered and stylish, measuring 170.5 x 52mm diameter and weighing 540g, making it substantial but not so heavy that a typical mic stand will sag under its weight. It connects via a regular three-pin balanced XLR mic cable, but, as with any tube device, it needs around ten minutes to warm up and stabilise after the 48V phantom power has been applied.



The miniature valve inside the AT3060 provides a useful degree of tube warmth even though it's only run from the normal phantom power supply.

Studio Test

Although the AT3060 has a tiny valve, the sound is anything but tiny. I was initially concerned that a miniature tube might sound very different to the tubes used in mainstream tube microphones, but I needn't have worried. There's nothing overtly overblown about the valve sound, but the mic does manage to sound warm, with plenty of density in the lower mid-range where voices often need support. Furthermore, the frequency response tailoring isn't at the expense of the high end, which is clear and well extended without being strident or artificially sizzly. In fact the whole sound offers a good combination of smoothness and clarity.

In addition to checking the AT3060 with vocals, I also used it to record a violin track with the mic set up behind and slightly above the player and found that this setup delivered a silky sound quality, with focus and clarity but not too much edge. The AT3060 also turned in a good result with acoustic guitar, which suggests it would make a good all-rounder as well as a very nice vocal mic.

Overall I feel the AT3060 delivers the essential flavour of what most people think of as the tube microphone sound, but without pushing the tube sound in your face. Although some spectral tweaking has been done to give the mic a specific character, this is subjectively quite subtle and, when used with vocals, the overriding impression is of a well-balanced, natural sound rather than anything too hyped. If you're after an attractively priced tube mic and fancy a change from all the 'me too!' Chinese models currently doing the rounds (very nice though some of them sound), then the AT3060 is an obvious contender, and the fact that it runs from regular phantom power rather than a separate PSU is a bonus. SOS

SOUND ON SOUND

Audio Technica AT3060 £400

pros

- Subjectively neutral sound helped by subtle flattery.
- Needs no external PSU, because it runs on regular phantom power.
- Attractively priced.
- Well built and stylishly designed.

cons

- As with any mic that has been designed to have a specific sound, it may suit some voices better than others.

summary

The AT3060 is a really nice little mic that has just the right amount of tube character to make it sound classy. It's also the first tube mic I've come across that works from normal phantom power.

information

£ £399.96 including VAT.
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Synth Secrets

Articulation & Bowed-string Synthesis

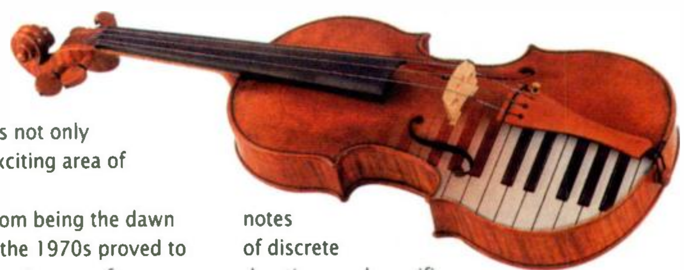
Gordon Reid

The skilful articulation of a synthesized string patch can improve it no end, even one created using very basic building blocks, as we saw at the end of last month. But we can take this approach much further...

Nowadays, when people talk about synthesizers and synthesis, they invariably talk about keyboard instruments. They may ask each other what type of synths they're using, but this will usually be a question about the nature of the sound generator... is it analogue, PCM- or sample-based, FM, additive, 'virtual' analogue, physically modelled, or a host of lesser variants. Alternatively, the question may refer to whether the instrument is monophonic or polyphonic, or it may be an enquiry about the manufacturer or model. But, with the exception of a handful of modules designed for use with wind controllers or guitar pitch-to-MIDI converters, synthesizers are almost always keyboards or modules controlled via CV and Gate signals or MIDI from a keyboard. Consequently, the vast majority of synths play discrete notes, with exactly 12 of them per octave, each tuned in accordance with the strict set of frequency relationships that comprise 'equal temperament'. Of course, you can often control electronic instruments using computers and/or sequencers, and some newer models offer control over aspects of the sound using innovations such as Roland's D-Beam, but the musical philosophy and the temperament of the notes remain the same.

The Tyranny Of Discrete Notes

It wasn't always like this. You only have to travel back 30 years or so to find bands such as Roxy Music and Hawkwind using synths to create new sounds that were part of the music, but were in no way played chromatically. You could argue that this was making a virtue of necessity — the EMS VCS3 that both bands used was the least likely to stay in tune of all the early synths — but this would be a simplistic argument. Brian Eno and Del Detmar could just as easily have bought a Moog or ARP and played widdly melodies as most of their contemporaries were doing. But, for a couple of years, a-melodic synthesis was not only acceptable, it was an exciting area of musical exploration.



Unfortunately, far from being the dawn of a new age of music, the 1970s proved to be the end of this fascinating era of synthesis. Despite numerous turns in the cycle of musical fashion, from electro-pop to industrial electronica, from New Age to dance, the music of the past 30 years has been firmly rooted in the 12-note chromatic, equal-tempered scale fully explored by Johann Sebastian Bach in the 18th century. Sure, *Sound On Sound*'s readers generally make more use of rhythm instruments than did the Baroque composers of Bach's era, and — for the most part — we make less use of melody and counterpoint, but 200 year-old central European music scholars would probably find no difficulty comprehending the form of today's popular music. Like my mother, they would undoubtedly hate it — but they would understand it.

This is stranger than it might seem. Whereas pianos and organs can only play discrete notes locked to a chosen scale, many traditional instruments are less constrained. All non-fretted string instruments allow you to play any pitch within their ranges, and many brass and woodwind instruments allow you to slide between the semitones defined by their holes or valves. And then there's the trombone, but I'm not sure that we should mention this in polite company.

So how did Western music become so firmly locked into forms of music limited by notes of discrete durations and specific pitches, all of which conform to the well-tempered scale? It didn't happen in India, nor in Bali... countries whose wonderful music sounds so 'wrong' to most Western ears. In all likelihood, it's the result of exposure to one musical form from birth, resulting in our stunted appreciation of what constitutes a note, what constitutes a musical interval, and what frequencies are acceptable in a melody. All of which brings us back to the synthesizer — an instrument that is in principle so flexible that it can emulate all the musical forms known, and be the inspiration for quite a few new ones, were it not for the limited imaginations of those who use them, myself included.

Now, you could point out that many modern synths offer alternative temperaments such as Pure Major, Pure Minor, Just, and Werckmeister III, but these are different ways of tuning the 12 keys that comprise each octave. The underlying philosophy remains unchanged.

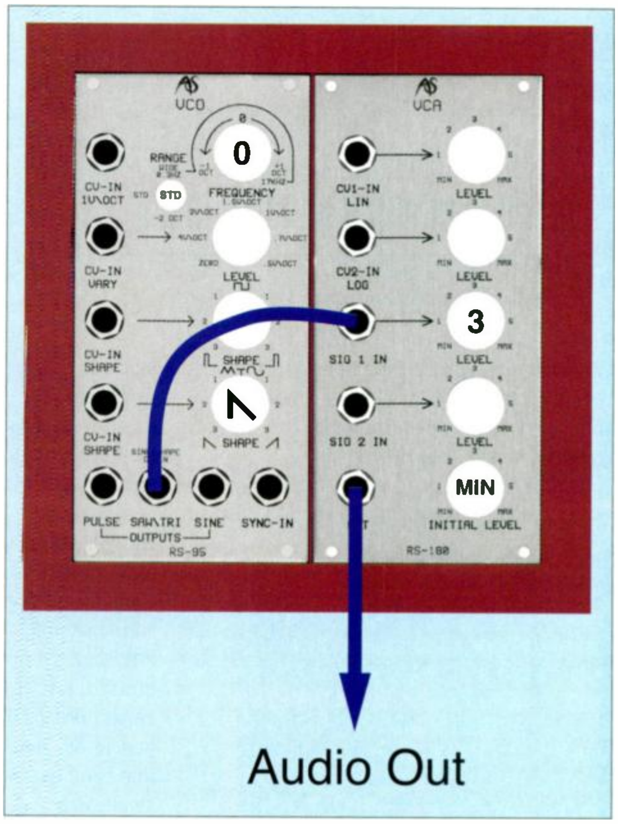
To find an instrument that breaks this mould, you might turn (as did I) to the experimental synthesizers developed during the first six decades of the 20th century. Some of these eschewed keyboards in favour of other mechanical systems for determining pitch, tone, duration and loudness but, if you inspect them closely, you'll find that many used paper tape with punched holes to play notes according to... conventional scales and temperament. In other words, a set of electronic sensors replaced the musician's fingers, but the musical philosophy was again unchanged.

Perhaps the only well known exceptions to the tyranny of the 12-note octave were the instruments made by Don Buchla in the 1960s and 1970s. His 'System 100' incorporated a sub-divided touch pad that made no concessions to the standard keyboard geometry, and allowed you to tune each division independently. Nevertheless, this *still* forced the player to think in terms of discrete notes with fixed divisions of pitch, with pitch-bend or slew to generate frequencies that lay between the notes.

As far as I am aware, there were — and remain — only two electronic instruments that challenge the domination of sub-divided pitch and discrete articulation of the notes. Both were developed in the early part of the last century, and both have found favour within all musical genres including classical music, experimental, jazz and *avant-garde* music, plus pop and rock. They are the Theremin and the Ondes Martenot.

Of the two, the Theremin is by far the better known, and over the past few years there have been many models produced. These range from small, basic, single-antenna boxes that cost a few pounds, to the popular 'Etherwave', through to Bob Moog's expensive recreations of the original, floor-standing instruments.

Figure 1: A very simple patch.



Nevertheless, we're not going to discuss the Theremin this month. Instead, we're going to concentrate on the Ondes Martenot, and show how an obscure musical controller invented three-quarters of a century ago might improve many aspects of your synthesis technique in 2003, particularly with regard to our bowed-string patches.

Another Way To Play Synths

The Ondes Martenot, unveiled by Maurice Martenot in 1928, is a fascinating instrument, and if you're not familiar with it, it's really worth investigating. One of the best Internet-based resources for obscure 20th-century electronic instruments is www.obsolete.com/120_years/, and you'll find a page devoted to this unique device at www.obsolete.com/120_years/machines/martenot/ (see left). Those unfamiliar with it might well ask what is so different about it — it is, after all, another keyboard-based instrument. But the Martenot has *two* pitch-control mechanisms. The first of these is its conventional keyboard, but we'll ignore this from now on. The second is a little ring attached to a wire that runs along the front of the instrument. This makes the Ondes Martenot almost unique among electronic instruments because, if you slip your finger through the ring and move it to the right, the pitch of the instrument rises without quantised steps; if you move it to the left, the pitch falls without quantised steps. The keyboard then becomes no more than a reference, letting you see where you are *vis-à-vis* the conventional scale, but you are no longer constrained by its discrete divisions: you can move the ring to any position you choose. This, of course, is the electronic equivalent of a fretless stringed instrument.

OK... so the wire and ring let you control the pitch, but how do you get a sound out of an Ondes Martenot? Or, more pertinently, having done so, how do you shut the thing up? The secret lies in a second control found to the left of the keyboard/ring/wire assembly; a large, wooden button that controls the loudness of the sound. When this is 'out' (ie. when you're not pressing it) the Ondes Martenot produces no sound. As you press the button in, the sound becomes louder until, when the button is fully depressed, the sound is at its loudest.

With careful use of the two controls, you can pitch and articulate notes in ways that are impossible on a conventional synth. Sure, we got close last month by using a joystick to control the loudness, but it's not quite the same, believe me.

All of which brings us neatly back to the string sounds we were discussing in the last couple of instalments of Synth Secrets, and the point at which I left you last month.

More On String Synthesis

Let's start by considering the patch in Figure 1 above. This must rank among the simplest of all possible patches, with two modules — in this case, a voltage-controlled oscillator and a voltage-controlled amplifier — connected by a single cord. At this point, there's no keyboard attached, so there seems to be no way to determine the pitch of the note, nor to determine its start and end times. However, if you turn the amplifier's Initial Level knob away from its

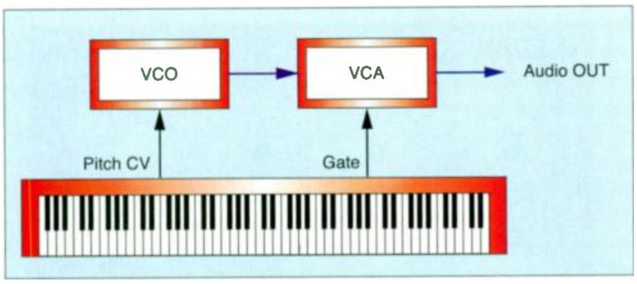


Figure 2: Connecting a keyboard to the modules in Figure 1.

minimum, you will obtain a note whose tone is determined by the oscillator's Shape knob, and whose pitch is determined by the Frequency knob above it.

On the most basic level, these three knobs provide everything you need to create musical performances. If you could twist them quickly enough and accurately enough, you could make precise changes in pitch and tone, and articulate the resulting sounds as you pleased, just as if you were using envelope generators, pitch controllers, modulation generators, and all the other bits and pieces that comprise an integrated analogue synthesizer.

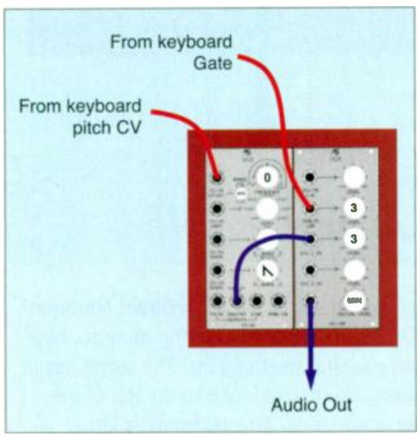


Figure 3: Adding patch cables to control the pitch of the note and when it sounds.

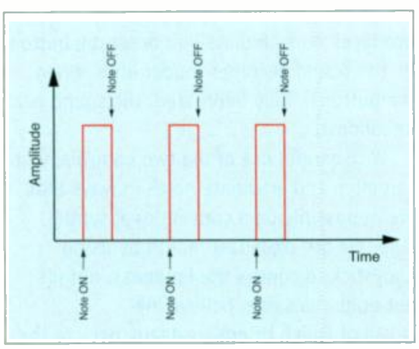


Figure 4(a): Using the Gate itself to control the gain of a VCA, resulting in 'organ-like' notes.

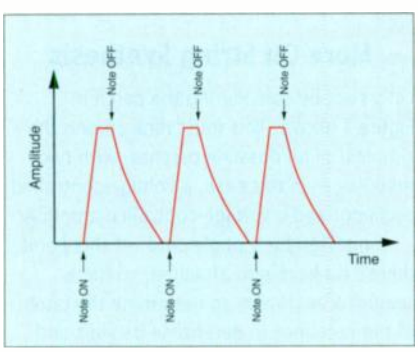


Figure 4(b): Using a contour generator to control the gain of a VCA.

Unfortunately, this is not simple. In fact, it's all but impossible, so we add a controlling mechanism to do most of the work for us. And, as discussed above, this mechanism is almost invariably a keyboard; perhaps with modulation and pitch-bend wheels that help us to inject some humanity into the performance, but a keyboard nonetheless. So we end up with the architecture shown in Figure 2 (top left), and the connections shown in Figure 3 (left). Note that I have connected the keyboard Gate directly to control the amplifier. This is an acceptable practice because synthesizers with +5V Gates usually have +5V or +10V CV inputs on their VCAs, so you will do no damage. Sure, you lose the shaping capabilities you obtain when you use the Gate to trigger a contour generator and then use the resulting envelope to control the VCA, but the square 'organ-like' notes produced by the Gate itself are acceptable because they are, well... like playing an organ (see Figures 4(a) and 4(b) left).

If we play the patch in Figures 2 and 3, we obtain notes with sawtooth waveforms and the predicted organ-like articulation. It's perfectly useable, and no doubt would have graced many 1970s prog-rock recordings, but despite being based on the appropriate waveform, it sounds nothing like a bowed, stringed instrument.

So let's replace the keyboard with a version of the Ondes Martenot that has been developed specifically for use as a controller of analogue synths. There is only one such beastie in production — the French Connection from Analogue Systems, shown below — and this offers the ring/wire controller and the amplitude button of the original instrument, as well as an X/Y joystick controller (for more on the French Connection, as well as much more on the Ondes Martenot on which it is based, take a look at the review back in *SOS* February 2002, or head for www.sospubs.co.uk/sos/feb02/articles/frenchconnection.asp).

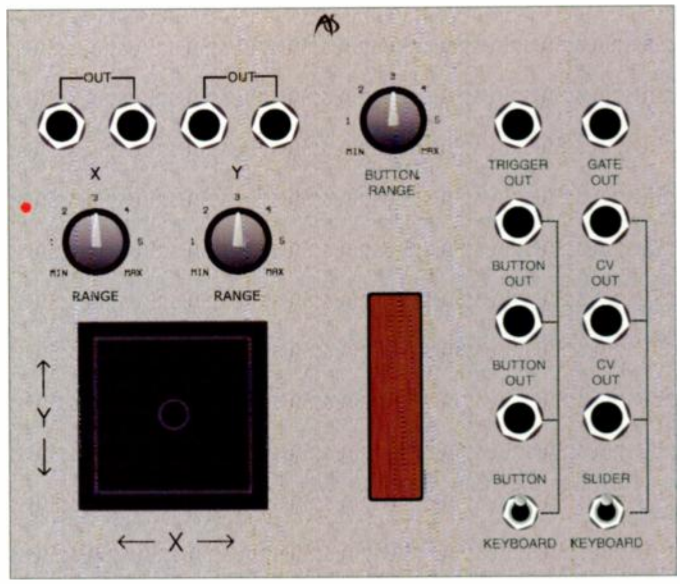


Figure 5: The control panel of the French Connection.

Figure 5 (above) shows the French Connection's control panel. As you can see, there are two switches to the lower right of this, and if we were to flip both of these to 'Keyboard' and connect CV and Gate cables to the appropriate sockets, there's nothing stopping us from using the instrument as a conventional CV/Gate keyboard. I've shown the resulting patch in Figure 6, and

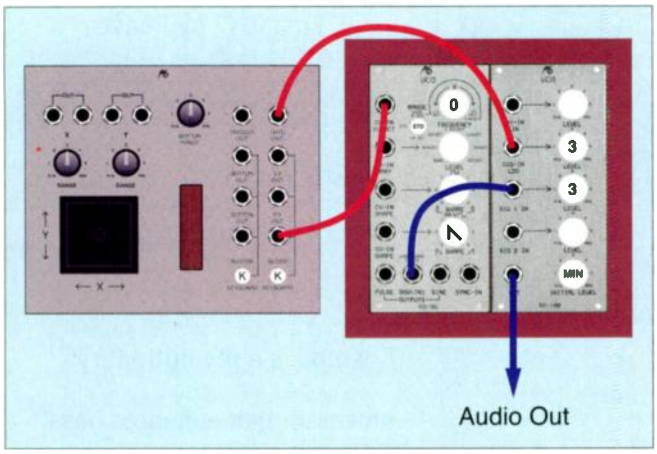


Figure 6: Controlling the notes using the French Connection's keyboard.

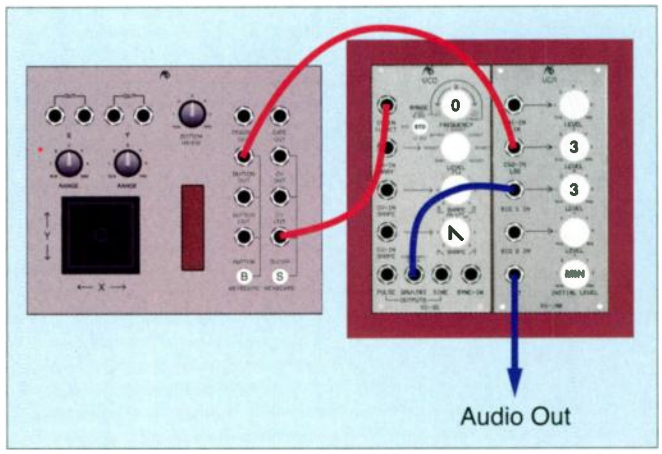


Figure 7: Controlling the patch using the French Connection's wire and button.

you can see that this is identical to that shown in Figure 3, so it will come as no surprise to you that it produces the organ-like sound already discussed.

However, if we now flip the switches to Button and Slider, playing the keyboard produces no sound whatsoever (see Figure 7 above). We must re-patch the cord in the Gate Out and insert it into one of the Button Outs. If we then press the button, we produce a variable voltage that controls the VCA Gain, and therefore the loudness of the sound. Likewise, playing the keyboard no longer controls the pitch. That duty is now undertaken by the wire, with the position of the ring determining the pitch at any given moment.

If we now play, the result is remarkable. With a little practice, the performance is no longer that of a soulless single-oscillator, unmodulated sawtooth buzz. You can add vibrato by wiggling your 'ring' finger from side to side, controlling both the speed and depth in a way that feels and sounds completely natural. Glide is merely a matter of pressing the button as you move to the next note. Moreover, judicious use of variable pressure on the button allows you to articulate notes in human ways, making each note unique in a fashion that is not possible when triggering envelope generators.

So, how does it sound? As you probably have guessed, the unquantised nature of the pitch controller, and the fluid way in which you can articulate sounds, means that the Ondes Martenot lends itself to imitations of unfretted string instruments. With just an oscillator and an amplifier, higher pitches sound remarkably like a violin, while lower pitches sound much like a contrabass or cello, although probably not one that the late, great Jacqueline du Pré would have cherished unduly.

The modern French Connection lacks the amazing resonant sound reproduction system of the original Ondes Martenot, but it nevertheless allows you to articulate and add expression to the almost limitless range of sounds available from any synthesizer that provides a pitch CV input and a VCA Gain input. Unfortunately, this precludes most common analogue (and digital) instruments, but if you own a modular or semi-modular synth, you're in business.

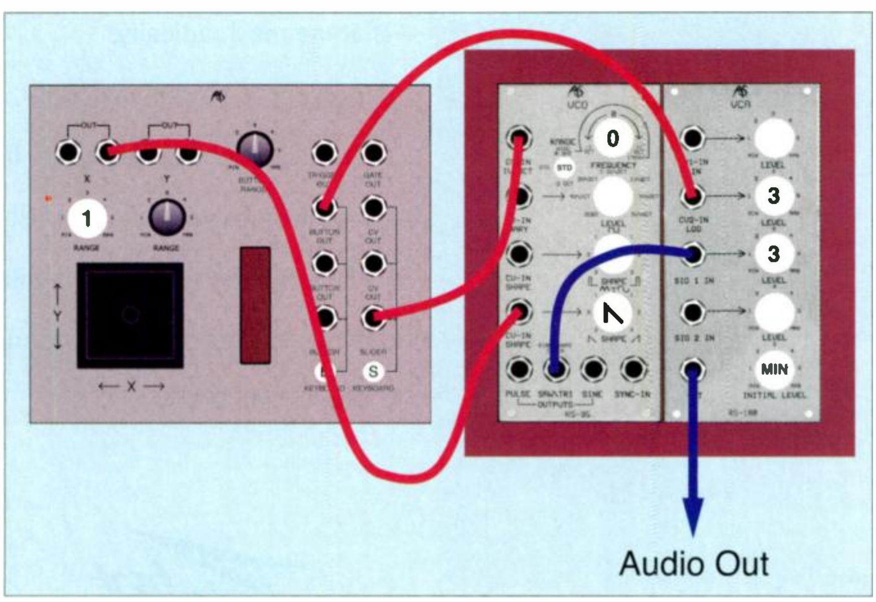


Figure 8: Controlling the harmonic content of the sound.

Of course, we need not stop here, and there's nothing preventing us from using the 'feel' of the French Connection with more complex patches. Figure 8 (below) shows a small extension to Figure 7 in which we control the waveform using the 'X' direction of the joystick. We achieve this by connecting a cable from one of the 'X' outputs to the sawtooth CV-In Shape input on the oscillator, so that left-to-right movements of the joystick change the wave from a sawtooth, to a triangle, to a ramp wave, and back again.

Playing this patch is surprisingly easy... use the inside of your left index finger to move the joystick while you use your thumb to press the button. Now you can articulate the note *and* determine its harmonic content with one hand, while playing the pitch with the other. It sounds simple, and it is, but this is something that you will find almost impossible on most synths. What's more, it's hugely expressive, because you can reduce the amplitude or even eliminate harmonics by moving the wave from a sawtooth towards a triangle as you reduce the overall loudness of the sound. This relationship between loudness and high-frequency content is — as we have discussed before in Synth Secrets — very much the behaviour of blown, bowed, strummed and struck instruments, and we're recreating it without a filter anywhere to be seen. Neat, huh?

But hang on a moment... Haven't I spent the last couple of months telling you that you need low-pass and high-pass filters, formant filters, modulation generators, mixers, joysticks, reverb, and loads of other gubbins to create even the barest likeness of

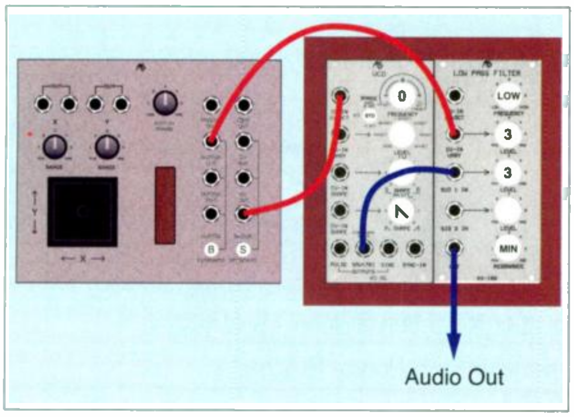


Figure 9: A simple but impressive (and, with the correct articulation controllers, *expressive*) brass patch.

a string instrument? Well, yes, I have. However, the ability to *play* the synth in the manner of a stringed instrument overcomes so many of the limitations imposed by a CV/Gate keyboard that we no longer need many of these to create recognisable imitations and performances. And they're *better* performances, believe me.

Adding Articulation To Other Sounds

Once you've got the hang of shaping notes and pitching them using the Ondes Martenot architecture, there's no reason to confine yourself to imitations of violins and cellos. You can create stunning imitations of instruments such as flutes and, in particular, vocal 'formant' sounds (both of which will be the subject of a future Synth Secrets). As for generating new sounds and effects, the freedom afforded by the ring and button opens up completely new areas of sound design and creation.

But perhaps the most fun (for today, at least) are the brass sounds that you can conjure effortlessly by replacing the VCA in Figure 7 with a low-pass VCF (see Figure 9 above).

As you can see, we're still using just two modules, and the patching is identical, but the button — instead of controlling the loudness — is now controlling the cutoff frequency of the filter, and therefore the tone of the sound. And, because the initial cutoff frequency is set to Low, all the harmonics are filtered out until you press the button, which means that silence reigns between notes. Consequently, the filter is not only shaping the tone of the sound, it's also differentiating one note from the next. This is incredibly elegant!

If you now play a note, articulate it with the button, and add suitable vibrato using the ring/wire, the result is magic, especially when played through a good digital reverb. You can play distinct notes, imitate swell, recreate the mis-pitching of certain notes that invariably occurs when playing real brass instruments, slide notes to imitate trombones... and if you detune the oscillator by a couple of octaves, you'll obtain the most realistic tubas you've ever heard from an analogue synth.

So there we have this month's Synth Secret; two modules and a more appropriate method of controlling them can be far more expressive and create more realistic bowed string and brass sounds than any number of modules and facilities controlled by a less suitable device. It's an important lesson, but because of the ubiquity of the keyboard synthesizer, it's not one that many people have had the opportunity to learn. SOS

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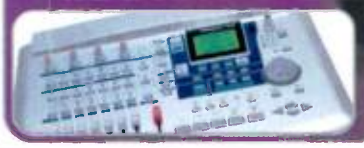

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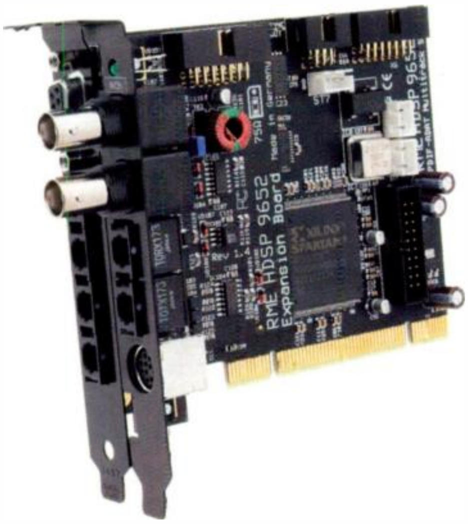
RME Hammerfall DSP 9652



Soundcard For Mac & PC

RME have added MIDI I/O and extensive DSP facilities to their well-regarded Hammerfall digital soundcard. Will the new Hammerfall DSP keep its place as a benchmark for audio interface design?

Mark Wherry

RME are one the most respected soundcard manufacturers in the professional audio market, developing a range of products that also encompasses audio converters and, more recently, mic preamps, designed for use primarily with native-based audio workstations. The Hammerfall DSP 9652 (HDSP 9652) is the successor to RME's original Hammerfall PCI card, first reviewed in *SOS* September 1999, which has become a popular choice for those wanting to integrate DAWs with digital mixers or stand-alone A-D/D-A converters. Indeed, Steinberg sold a rebranded Hammerfall card as the Nuendo 9652 card, and the new HDSP 9652 is card is also available from Steinberg under the Nuendo moniker.

Hammerfall DSP technology was first introduced in RME's Hammerfall DSP Cardbus and PCI card products, with the associated Multiface and Digiface breakout boxes, which *SOS* reviewed as Steinberg's Nuendo Audiolink 96 system in September 2002. However, with the success of the original Hammerfall card, it was only a matter of time before RME would build the Hammerfall DSP technology back into the company's self-contained digital I/O Hammerfall PCI card, and the result — with the addition of some other nice touches — is the HDSP 9652 card.

As an existing Hammerfall user and a fan of RME's technology in general, I was particularly interested to see how RME could possibly have improved on what was already considered by many professionals to be a perfect soundcard.

SOUND ON SOUND

RME Hammerfall DSP £399

pros

- One of the most feature-rich soundcards available for digital I/O.
- Multi-client ASIO drivers.
- Advanced monitoring capabilities with Total Mix.
- Indispensible analysis tools.

cons

- A headphone output, as on other Hammerfall DSP products, would have been a nice touch.
- Mac support is lagging behind the developments for Windows.

summary

The existing RME Hammerfall card was close to perfection, but the new Hammerfall DSP still manages to improve on it!

Feature Perfect

As in the original Hammerfall, the 9652 product number refers to the fact that the card supports sampling rates up to 96kHz and audio inputs and outputs for up to 52 channels — 26 inputs and 26 outputs, from three ADAT I/O pairs and co-axial S/PDIF I/O. Each ADAT channel supports 48kHz/24-bit operation, and S/MUX mode is also available to pair ADAT channels in order to provide 12 96kHz/24-bit channels instead. Like its predecessor, the HDSP 9652 also implements what RME refer to as 'ASIO zero CPU load' technology, meaning that you can have 52-channel operation without placing any burden on the host processor.

An ADAT 9-pin sync connection is also available for sample-accurate transfers (with a suitable ASIO 2-compliant application, such as *Cubase*), and BNC word clock I/O connections are still provided. On the subject of clocking, the DSP 9652 card excels at keeping you aware of the clock status of your audio and sync ports via the Sync Check area of the control panel window, and thanks to RME's Intelligent Clock Control technology, the card can automatically sync to the port supplying the most stable clock signal — very handy.

Co-axial S/PDIF I/O is supplied on the same breakout cable as the ADAT 9-pin connection, and you can assign the first pair of ADAT ports to transmit or receive S/PDIF via optical connections instead. As existing Hammerfall users know, it's also possible to set this so the first ADAT output transmits S/PDIF and the first ADAT input still receives ADAT data, or *vice versa*, which is rather neat.

The original Hammerfall was purely an audio card, but the HDSP 9652 adds two pairs of MIDI ports for 32 MIDI input and output channels via another supplied breakout cable. This is a perfect touch since there are few situations where audio workstations don't require MIDI ports — even if you never use MIDI instruments, hardware control surfaces such as Mackie Control still connect over MIDI.

Although the HDSP 9652 uses one PCI slot in your computer, many of the connections are supplied on a separate expansion card that needs to be fitted in an additional backplate, in the place of another PCI card. And if you find the I/O options of the DSP 9652 not to your liking, a range of analogue and TDIF expansion boards are available separately (see The Hammerfall DSP Family box for more information), and you can fit a

Test Spec

- Asus A7S333 motherboard with Athlon XP1800 processor, 1GB DDR (PC2100) memory and an ATi Radeon 7500 64MB dual-head graphics card, running Windows XP Professional.

second HDSP 9652 card to the same system to double the number of available inputs and outputs. The only extra I/O-related feature I'd perhaps like to see on the HDSP 9652 card is a headphone output, which is present on every other Hammerfall DSP product (see the Family box), even if it just mirrored a digital output pair for test and monitoring situations.

Who's Gonna Drive Your Card?

The HDSP 9652 is compatible with pretty much every major operating system: for Windows users there are multi-client MME drivers, ASIO 2 and GSIF (for *Gigastudio*) drivers on Windows 98/ME, and full multi-client MME, ASIO and GSIF drivers for Windows 2000/XP users. The multi-client ASIO drivers in this latter set, which RME added during the course of the review, deserves a special mention since they make it possible for two ASIO-compatible applications to be loaded and use the same soundcard and drivers simultaneously, so long as both applications are using different channels. This makes it possible to use *Cubase* and *Wavelab* at the same time via ASIO, or even *Cubase* and *Nuendo*, for example; and this new multi-client support will have engineers salivating when they see it's now possible to run RME's highly regarded *Digicheck* analysis software alongside an ASIO application, as described in the *Digicheck* box below.

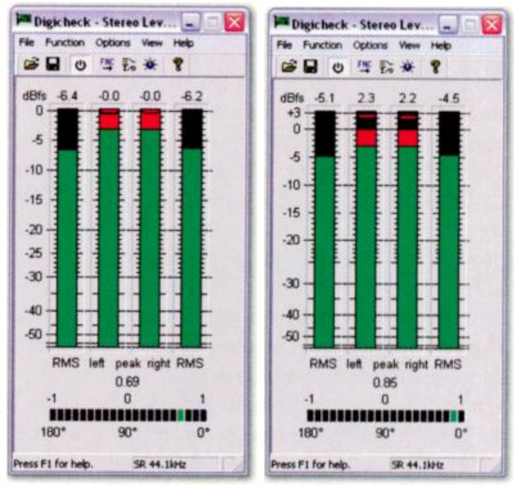
Mac users are catered for as well, with ASIO 2 and OMS drivers for Mac OS 9, and Core Audio support under Mac OS X. However, Mac users won't benefit from any multi-client functionality, and, unfortunately, there are no Core MIDI drivers for OS X users to make use of the MIDI ports at present.

Hammer To Fall

RME have always been pretty good at supporting their products with driver updates, and the HDSP 9652 card is no exception. During the review, for example, I needed to update the drivers to try out some of the latest features, and in order to use the newer drivers, it was also necessary to update the firmware on the HDSP 9652 card itself. However, I got nothing but errors when I tried to use the firmware update program downloaded from RME's web site, which worried me that the firmware update process might have damaged the card, or that it was faulty in the first place. Fortunately, the HDSP 9652 card features what RME describe as Secure BIOS Technology, which prevents the card from being damaged if there's a problem when updating the firmware. This works by providing a second, read-only BIOS on the card that's automatically activated if the firmware upgrade fails, enabling you to check your system and redo the flash update.

After emailing RME's Matthias Carstens, my problem was quickly identified. Apparently, the update procedure for the FPGAs (Field-Programmable Gate Arrays) from Xilinx that RME use on the Hammerfall DSP card is incompatible with certain motherboards, and the only solution if the user encounters a problem when updating the firmware is to put the card in another computer. RME are working to improve the situation, and it should be stressed that it's not the HDSP 9652 card itself that's incompatible with certain motherboards, only the firmware update process. Once the firmware has been upgraded, you can move the card back to the first computer and it should work fine.

The motherboard I was using that caused



Oversampling Mode in *Digicheck* provides peak metering that takes into account the extra harmonics generated by clipped digital signals. Although it peaks at 0dBFS, Madonna's 'American Life' thus generates a higher reading when OVS is switched on (right).

Digicheck

For some time, RME have supplied an often-overlooked secret weapon with their soundcards called *Digicheck*. This is a Windows-only utility based on RME's own DAM1 'Digital Audio Monitor', which provides real-time analysis tools for audio engineers. One of the reasons *Digicheck* has been overlooked in the past is because, until recently, it only worked as a stand-alone utility and couldn't be used alongside an ASIO host application. But with the release of *Digicheck* 4 (which was available as a pre-release at the time of writing) and the new ASIO multi-client drivers, you can now use *Digicheck* to monitor the inputs or outputs alongside an ASIO host application.

***Digicheck* has many different functions, including Stereo Level Meter, indicating the output of a given stereo pair with both RMS and peak readouts, Multi-Channel Level Meter, showing the levels of eight channels at once, and Global Level Meter, displaying all 26 channels simultaneously. There's also a Spectral Analyser offering 10, 15, 20 or 30 bands, a Vector Audio Scope, with a Goniometer, correlation and peak and RMS displays, and Totalyser, which combines both the Spectral Analyser and Vector Audio Scope in one function. And finally, *Digicheck* provides a Channel Status Display to show all the Channel Status data included in a digital signal, and Bit Statistics and Noise provides information about the current state of each bit in the digital data stream, and offers DC- and A-weighted analysis. To call *Digicheck* comprehensive is an understatement.**

New in *Digicheck* 4 is Oversampling Mode (OVS), which enables the measurement of signals higher than 0dBFS. This sounds implausible, since levels higher than 0dBFS are actually impossible in the digital domain, but as Matthias Carstens explained to me, it's possible to generate signals that violate the rules of digital audio. The example he gave, which was demonstrated at this year's Frankfurt Musikmesse, concerned the use of digital clipping to get a higher volume on audio CDs: since the majority of tools available don't include advanced filtering when applying hard or brick-wall limiting, heavy digital compression or clipping, the signal won't be bandwidth-limited and will include harmonics beyond the Nyquist frequency. The resulting waveform will be clipped and now represents what Matthias described as a 'violating' waveform.

When OVS is enabled in *Digicheck* you can see the level the original material had before it was clipped, so I decided to check this out with the 'American Life' track from Madonna's recent album of the same name as a good example of a modern CD release. When OVS was disabled, the peak meters showed the signal peaking at 0dBFS, and with OVS enabled, the peak meters showed the signal peaking at over 2dBFS. Since OVS doesn't change the display of valid digital data, Matthias recommends leaving OVS activated, except when checking final masters for overs when preparing a Red Book-compatible CD.

***Digicheck* is, quite simply, a brilliant tool that is now utterly indispensable. Thanks to the multi-client drivers, it's perfect for use alongside applications like *Cubase SX* and *Nuendo*, which don't include the same detailed real-time analysis tools as standard. Although *Digicheck* can be used with other RME products than the Hammerfall DSP family, the maths required for the signal analysis is carried out on by the FPGA on the soundcard, reducing the CPU load on the host computer when you run *Digicheck* on a Hammerfall DSP system.**

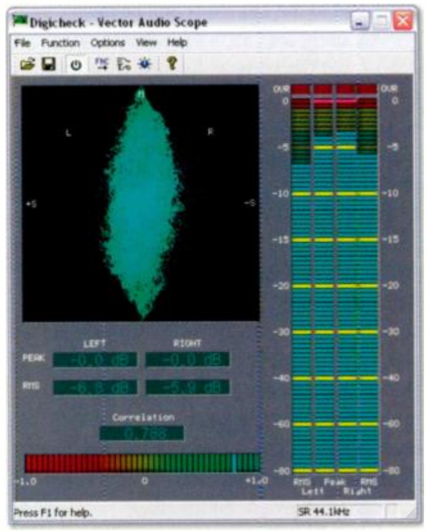
▶ the problems was an Asus AV7M266, a Socket A motherboard featuring an AMD 761 chipset. I tried putting the card in a similar Athlon-based system with an Asus A7S333 motherboard, which has a SiS 745 chipset, and this time the firmware update process worked perfectly. I shouldn't imagine this will be a problem when using the card in a Power Mac, but if you have an incompatible Windows-based system and no additional machine to perform the firmware upgrade, you can send the card to your friendly RME distributor to do it for you.

DSP Without DSP

The DSP appendage to the Hammerfall name refers to the card's ability to carry out certain DSP functions such setting up monitor mixes. Oddly, however, it turns out that the HDSP 9652 doesn't actually have any DSP chips on board; instead, RME provide hardware DSP functions using FPGAs, as mentioned earlier.

A Field Programmable Gate Array (FPGA) can be thought of as a reprogrammable processor, where the behaviour of the logic elements can be configured by the designer at any time, even via a downloadable update from the user's computer. So instead of using an off-the-shelf DSP chip or having custom chips manufactured, RME have used an FPGA to provide the specific behaviour required for the DSP functionality the company wants to implement. This means that later, rather than being limited by the fixed behaviour of a DSP chip, RME can simply update the FPGA if its design needs to be changed to accommodate additional functionality.

One of the big questions asked by new users of audio workstations is whether they need an additional mixing console, and while my personal response is always an emphatic 'no', there are certain tasks, such as setting up monitoring, that are often made easier by adding a digital mixer when recording groups of musicians in a studio. However, to



Digicheck's Vector Audio Scope.

overcome the need for an additional mixer for monitoring duties, the Hammerfall DSP series of products all offer Total Mix, which provides complete control over the input, playback and output channels of your workstation.

The second version of Steinberg's ASIO driver API technology introduced a technology called ASIO Direct Monitoring, which allowed the soundcard to route an incoming signal directly to an output, rather than having it pass through the ASIO host application, such as *Cubase*, and incur a latency that would make it difficult for the musician to play along with the rest of the song. Total Mix basically takes the concept of ASIO Direct Monitoring many steps further by letting the user decide which inputs and playback channels are mixed to which outputs, and all of this happens independently of your ASIO host application, without affecting the way it's set up to record the incoming signals or play back the outgoing signals.

As a simple example of how Total Mix can be useful, say you're recording a guitarist in your live room and you want to give them a different monitor mix to the one you're hearing in the control room. The guitarist's headphones can be connected to a different output pair to the control room, and using Total Mix you can route the input from the guitarist to both monitor mixes on the different output pairs simultaneously. Now, say the guitarist wants to hear more of himself than the playback (for example!), and the engineer wants to hear a balanced level of both the playback and the guitarist — no problem with Total Mix. Simply turn the playback level down on the playback channels routed to the guitarist, and turn the guitarist's input down on the engineer's monitor mix. To finish, the guitarist's mix might be overloading, so you can simply turn the output channels down on that monitor mix.

It's a simple example, I know, and one that's really quite easy when a separate mixer is used with your audio workstation — but that's the point: Total Mix facilitates this type of situation without the user needing a hardware mixer. And as I mentioned before, Total Mix works independently of your ASIO host, so the guitar would still come in on the same input channels at unity gain from your software's perspective, regardless of the routing and levels in Total Mix.

To give you some idea of Total Mix's power, any input and playback channel can be routed to any number of output channels, with each routing having its own independent level setting. This means, to quote the figures on RME's web site, with the 26 input and 26 playback channels being mixed to the 26 output channels, the mixer must be able to do the sums for (26+26) x 26 = 1352 channels simultaneously, which is rather impressive.

Conclusion

As I mentioned at the start of this review, I've been a fan of RME's products and technology for several years now — the attention to detail in every aspect of development is exceptional. Having experienced nothing but rock-solid performance from any audio workstation I've used with an RME soundcard, these products are usually what I recommend if no additional features are required along the lines of Creamware's SCOPE cards, for example. Trust isn't always something you have can have with a product or manufacturer, especially in the technology market, but I trust RME and the HDSP 9652 maintains their tradition of great, yet affordable technology. SOS

The Hammerfall DSP Family

If the all-digital nature of the HDSP 9652 card doesn't suit you, RME offer the EXB range of expansion boards, including the AEB4I, AEB8I, AEB4O and AEB8O, which offer four or eight analogue inputs or outputs, and the TEB TDIF board. The HDSP 9652 supports two EXB boards, which connect to the main card via a ribbon cable and require an additional backplate in your computer, with any combination supported, including two of the same type. It's worth emphasising that these expansions don't offer additional channels, though, just alternative ways of getting audio into and out of the soundcard from the ADAT connections.

If the HDSP 9652 still doesn't sound like the card for you, RME's HDSP 9632 might be more appropriate where Total Mix is still desirable, but a smaller number of channels is required. It features a single pair of ADAT I/O ports, co-axial S/PDIF, a balanced stereo analogue input and output, a single pair of MIDI I/O connections, and a headphone output for monitoring that mirrors the other analogue output. Alternatively, there's the forthcoming HDSP MADI, which will be Windows-only initially, offering a pair of MADI connections for 64 inputs and 64 outputs via optical or co-axial connections, with two pairs of MIDI I/O, word clock I/O, and a stereo analogue output for monitoring. While MADI is usually only found on high-end professional equipment, RME have also recently announced the ADI648 converter box, featuring eight pairs of ADAT I/O to a MADI pair. And, finally, there's the original Hammerfall DSP Multiface and Digiface systems as mentioned at the start of the review.

information

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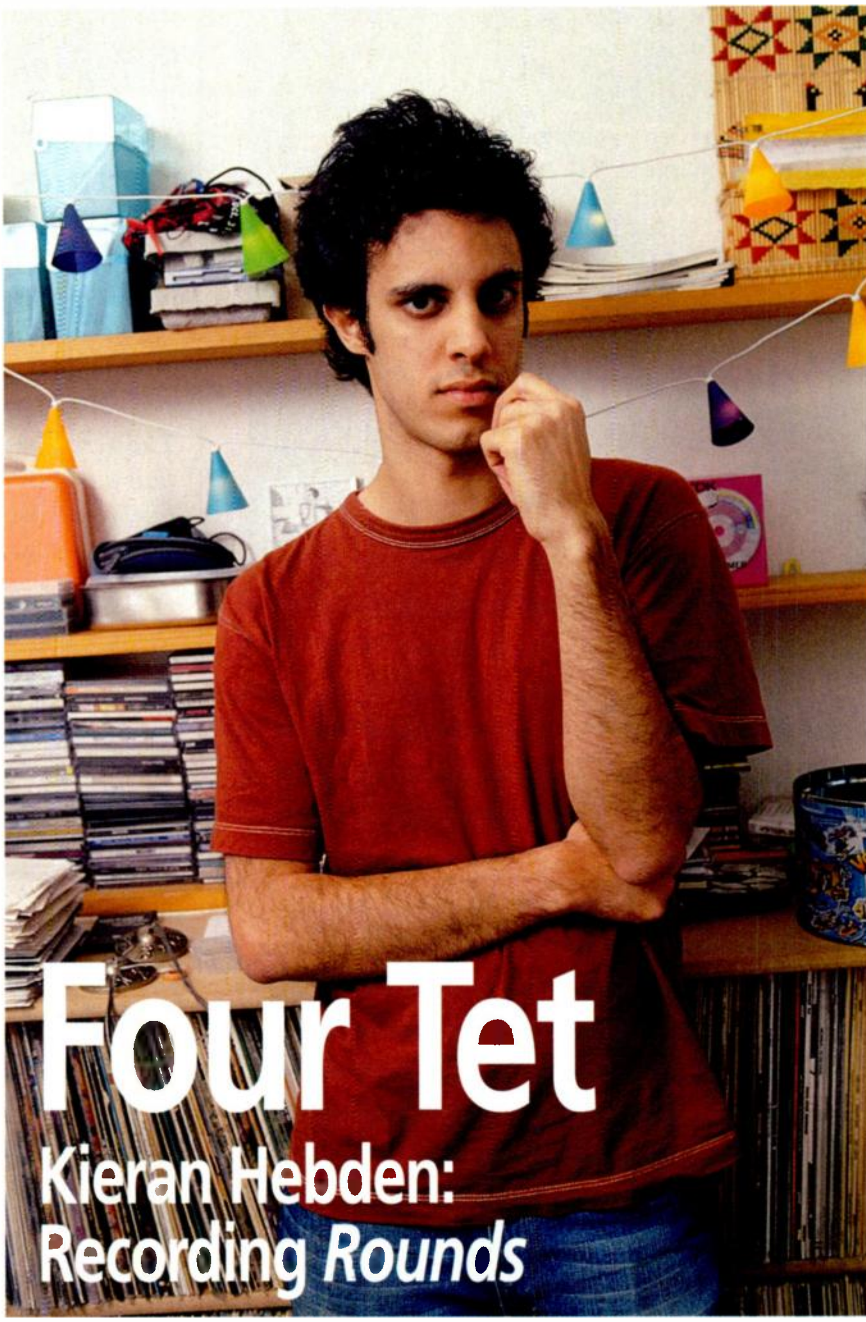
Kieran Hebden of Four Tet is a producer who puts the intelligence into Intelligent Dance Music.

Sam Inglis



Photos: Piers Allardyce

Four Tet

Kieran Hebden: Recording *Rounds*

Two years ago, Keiran Hebden released his second album as Four Tet. *Pause* went on to feature in numerous critics' Top Tens of 2001, and Hebden was even credited with inventing a new musical genre, 'folktronica'. Topping that achievement was always going to be a challenge, but it's one he seems to have met with his new album. Its implausible but beautiful blend of fragile acoustic fragments, brutal beats and glitchy electronica has already garnered a rich crop of five-star reviews, and *Rounds* looks set to be every bit as influential as its predecessor.

Everyday Life

Kieran Hebden is the living proof that you don't need esoteric equipment to make quality music. The music room in his north London flat contains just one piece of specialist recording gear, and that's a DAT recorder. His tracks are put together using only an unremarkable hi-fi and a Windows PC with a Creative Labs Soundblaster Live soundcard, running a motley selection of software, much of it lagging well behind the cutting edge.

"I don't like studios very much at all," he insists. "I feel really uncomfortable in them, I always get the wrong sound and nothing works out the way I want it. I feel a lot happier in here. I know I can get the sound I want in here, because I know these speakers off by heart, I've heard a million records through them. I also like the way I can get up in the morning, sit in my pyjamas and eat a slice of toast and work. The music becomes part of my everyday life. That's become one of the defining things about the music I'm doing, I don't go and hide away in a studio to get it done, I am able to do it in this really relaxed way where it does tie in with all the other things I'm doing in my life."

Despite the importance he attaches to a relaxed attitude, though, what's most striking about Hebden's approach to making music is its discipline. Unlike many producers, he doesn't seem at all daunted by the endless possibilities that computer-based recording generates. "I get asked in pretty much every interview, 'How do you know when a track's finished?'," he laughs. "I'm like 'That's the easiest thing ever.' I think that's what my talent is, essentially. That's what makes a producer. The whole reason I'm talking to you, and that the record's coming out, is because I know how to work on something and know when it's finished, or know when it's not finished and what it needs, and when it's going to make sense to listeners. That's the bit that's second nature. The more difficult part is coming up with ideas and starting points for tracks that have some real substance behind them, both in terms of emotional content and musical ideas."

A Man Of Ideas

Ideas are central to Keiran Hebden's music-making process, and although improvisation also has an important role in generating his distinctive soundscapes, it's always used in the service of a clearly defined musical goal. Even the 'folktronica' genre is something Hebden has thought about and planned from the top down. "For the very first Four Tet recordings, I was obsessed with free jazz, and the whole spiritual jazz thing from the late '60s and early '70s. There was so much dance music coming out that was claiming to be influenced by jazz music, but it was always really mellow, much more laid-back, and the influences were always fusion influences rather than the really dark, evil sort of jazz that I knew and loved. I wanted to do a contemporary record that was influenced by the evil dark jazz, and the whole first album was all about that. When when the next album came out, I abandoned the whole jazz thing — not because I wasn't into it any more, but because I suddenly started thinking about what was possible with this project and where it could go. I became a lot more ambitious.

"For the second album I was really into all this American R&B stuff, like Timbaland and Rodney Jerkins, who were putting out big chart records with harps and thumb pianos and really weird instrumentation. This made me think 'I need to be loads more ambitious about the sounds I'm using and the instrumentation. All the things that everyone's so familiar with, bass, drums, guitar and piano, that's in the past now, that way of thinking. It can be three harpsichords, a banjo and four drum kits, and that can make sense.' I was thinking about that and I got really interested in British folk music, and started buying loads of records and hearing people like Pentangle and Fotheringay and Fairport Convention. I wasn't crazy about all the music, but I really liked the sort of sounds I was hearing on the records. The thing I was looking for in that music that I couldn't find was a much more heavy rhythmic element, and that's when I thought 'Imagine if you got Kraut-rock and British folk music and fused them together,' and that became a kind of blueprint for the second album.

"For this one, I was happy with a lot of those ideas, but if there was one criticism I had [*of* Pause], it was that it did sound like I'd got a bunch of records and squashed them together at moments. I felt that it was time to try to be a bit more ambitious about what I was doing, and try to move things away from being such a product of their influences, to start thinking about making music that was coming more from me, where people would hear it and they wouldn't be able to know what records I'd listened to or what sort of music it was, even: something that was coming from my complete knowledge of music rather than what I was interested in at the time."

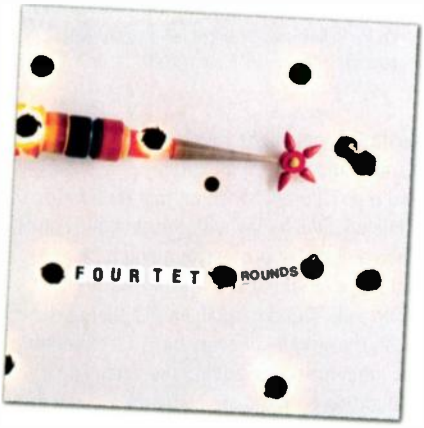
In The Beginning

When he began work on *Rounds* a year before its eventual release, Hebden thus already had a firm idea of what he wanted to achieve and how he wanted it to sound. "One of the things I wanted to do on this album was not try to achieve the sound by endlessly layering everything up a lot, and try to do some tracks that only had two or three sounds and keep that interesting for the whole song without introducing tons of countermelodies and stuff to keep it going.

"Particularly on this record and the previous one, the idea is very much that the computer's the instrument. If I wanted a guitar line or something, I'd never pick up a guitar and write a guitar melody to go on it. I might record some guitar into the computer, then start working on a track, and if I decide

Kieran Hebden: "I thought 'Imagine if you got Kraut-rock and British folk music and fused them together.'"

I need some guitar, I'd go to that recording, break it up into pieces, and then compose the melody using that sound. To get the sound I want and do what I want to do, it's all about using the computer as the instrument, and the most interesting stuff I've done has been all about that kind of idea. Loads of people think there's lots of live playing on my music, but there's nothing at all. It's all from the computer. I think one of the nice things about



that is that you listen closely and it conjures up the image of a musician, but you realise that everything's humanly impossible. It's the little details that make it.

"I think I f**k with things a lot more than people realise, to get them to work the way I want. There's a track with a gamelan on it on the record, for instance, but it's in a different scale to Western music, and to make it work within the track I sampled a loop of gamelan from the record, but I had to break it up into all sorts of pieces to try to get the timing right, and I had to stretch bits to get it to work in the context of the track I was doing. Every single sound's been messed with like that. On *Pause*, for instance, there was no bass of any sort, all the bass was all the other instruments slowed down, and everyone kept saying 'It sounds like this is a band playing.' For me that was a really insane idea, because I knew that everything was at such unrealistic speeds, and so many of the notes are backwards and messed with.

"There's different ideas behind what I'm trying to achieve at the beginning stages of a track. I might have a very clear idea in my mind about something melodic that's got to happen, and I'll put all my energies into making sure that works out right first, and then sort out the other bits. Sometimes I might have quite a clear idea for the whole thing, and sometimes I might just do a track in a couple of hours. It really depends on the feel I'm going for what my starting point is. It changes a lot depending on whether drums are something you start very early with in the track, or whether they're something that comes a lot later."

Through The *Rounds* Windows

Typically, Kieran Hebden has very considered opinions on the process of making music on a computer: "People who make music on computers don't realise how powerful the visual element is. Whether you like it or not, your minds starts to think in terms of patterns, because it's a natural human way to do things, and you start seeing the way drums are lining up on the screen, and it becomes completely instinctive to line them up in a certain way. It's important just to close your eyes and use your ears, and trust what's coming out of the speakers more than anything. So many shitty records come out with terribly programmed drums, where everything's really stagnant and sterile. It's always been a problem with British dance and pop music when they've got a hip-hop beat on something. It's always so sterile, and when you listen to American productions they walk all over them, because there's so much more swing. It's just confidence — they'll just get an MPC and hit the beat out, and it's human at that point, they haven't just hit the quantise button. They trust their own judgement more than the computer's, and I think that's an important thing to try to remember."

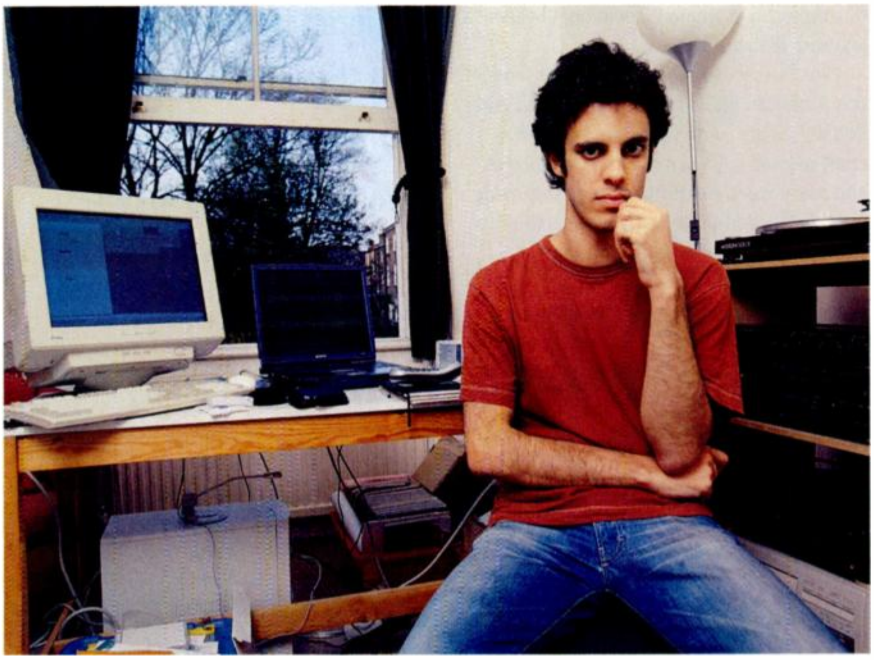
Hebden's main tool for creating and manipulating sounds is Ross Bencina's cult shareware program *Audiomulch*, an idiosyncratic modular virtual studio package for PC platforms which is downloadable from www.audiomulch.com. He also uses Syntrillium's *Cool Edit Pro* to chop up and trim audio, before assembling the results in Cakewalk's *Pro Audio* 9.

"I really like *Audiomulch*, it's one on my favourite bits of software. It's so stripped-down, and you can really abuse it. *Reason* just makes trance music, and you're not really allowed to piss around with it too much and everything's got this horrible reverbed sound, and I kind of hate it — *Audiomulch* is my preferred version of that idea. I find that what I do is just make sounds, get things going in *Audiomulch*, write it all to audio, and then I'll take samples. I'll do a lot of mad improvisation in *Audiomulch* and I'll be writing it to a WAV file the whole time it's going on, and then I'll play it back and find a good bit and grab something from it.

Audiomulch is so quick. If I'm playing a record and I suddenly hear a bit of guitar I want to use, I can try it out with 20 different drum loops in the space of five minutes. Everything I do is rendered straight away.

"In *Audiomulch* you can set the universal tempo for the whole thing and know that everything's going to be running in time, and every little effects unit's got quantise on it, so if it's getting too out-there you can whack all the quantises into effect, and have all the granulators and delays and effects locked in with each other. *Audiomulch* is really good for beats because you can put any sound you want into the drum machine, and because of the universal quantise, you can set the whole machine to 120bpm, get two or three breaks off an album, play them all at once in the right time with each other, and get a drum machine and program in some 808 hits and stuff to get the sound you want, and then you can add a different EQ to every single one of those elements and create a mix.

"To get the right feel of the beat and to get the basic loop working right, I use *Audiomulch* loads, but then I transfer everything over into *Cakewalk*, and then I start chopping it up, and then I start making all the variations in the drums. I usually take a whole bar or so from a record, and then start to add other sounds to it to get the sound I want. If the bass drum's not quite right I might get that and I might add other sounds to that, or if there's a bit I don't like in the drum loop I might chop it out and replace it with something else. It's often the case that my drum loops may be a combination of two or three drum loops from something else."



Kieran Hebden, aka Four Tet, in his music room with desktop and laptop computers (left), and hi-fi (right) used for both sampling and monitoring duties.

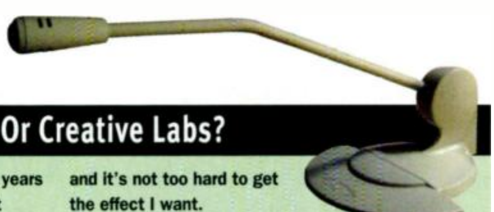
Containing Ideas

Hebden's self-discipline is most apparent in his approach to using sequencers. "*Audiomulch* is really powerful, but everything that comes out is a bit off the hook, so I use sequencers to contain my ideas, in a way. I'll start recording, get all the rough ideas together on a track, and chuck it all into the sequencer, and then I'll probably sit on that for a little while until I'm happy with the general sounds. Then there's the process of going back and refining everything and making sure all the drum rolls sound natural, and all the gaps are right, the lengths of every bit. That's probably the longer process, adding the detail and making sure everything comes in and out at the right moments — the arranging, basically. If I want to change the volume of the sound or anything, I don't use the mixer, I can only turn things up or down 3dB at a time. If I want to add an effect, a lot of the time I just put it in and that's it, there's no going back. I just write everything to WAV files. I mix as I go along, it's not like I make the track and then try to mix it afterwards — each instrument's added and it's put in how I want it to be. You get glitches just by chopping samples in the wrong places. I don't really think about all the separate sounds as putting them into separate contexts when I'm working on the music, I'm not thinking 'Right, now I want to bring some glitchy stuff in, then I want some drums.' Once all the sounds are in the computer, all of it's just sound, really, and they don't become separate moments in the process.

"I use *Cool Edit* for all my sample editing. I do a lot of time-stretching and stuff in *Cool Edit*, and I really like it — it's kind of shit, but good shit. The one thing I've looked to other bits of software for is time-stretching, because every time-stretching device sounds so radically different. Sometimes you can get time-stretching that sounds like it's underwater — sometimes you want bad time-stretching to get that really stuttered effect. I do a lot of time-stretching and editing manually. I don't use any programs like *Recycle*, I chop everything by hand. A lot of the time I don't even quantise, I do it by sight."

Hebden's approach to working with computers is very much based on destructive processing and manual editing, and he tends not to use plug-in effects: "I only use about two plug-ins. I use the TC *Native EQ* and the Waves *C1* compressor quite a lot, and some reverbs, but not much. Everyone's records end up sounding the same, so I really haven't explored plug-ins at all."

When it comes to computer hardware, Hebden's requirements are fairly



Neumann, AKG, Schoeps... Or Creative Labs?

"I have recorded everything for the last three years on this," says Kieran Hebden, indicating what looks like a miniature desk lamp. "It came free with my Creative soundcard. I've recorded electric guitars on this, acoustic guitars, I've recorded vocals, and with this microphone I've done remixes for everyone — Beth Orton, Badly Drawn Boy, all those people. It's a wicked microphone but it's a bit light, so I put it on the desk and clip it under the mouse mat to hold it in place. I have to filter out the rumble from the hard drive, but the acoustic guitar sound from it is stunning. On the last track on the album, 'Slow Jam', all the guitars and everything were done on this microphone. With all the EQs and everything on the computer you can do so much to the sound once it's in there, and it's not too hard to get the effect I want.

"I use it mainly on remixes: it's harder to use samples when you do a remix because of finding stuff that fits the chord sequence. If you've got a guitar line and you want it to continue into the chorus of the track, say, it becomes really hard. Say you're working with samples from a record with some guitar, getting it to sound like a natural chord change without loads of time-stretching, or the guitar suddenly being manically sped up or slowed down, would make it too weird. I like to retain the song when I do a remix, so a lot of the time, the first thing I might do is learn to play the song on guitar and just lay that down into the track straight away."

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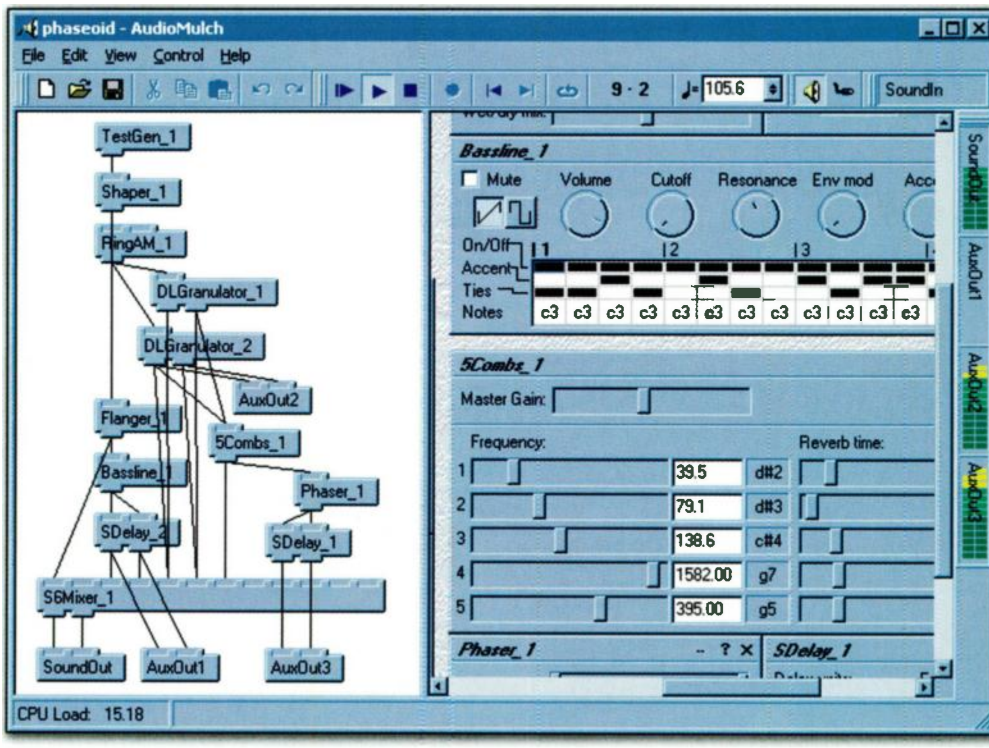
conventional. "I put the computers together myself, and buy in exactly the bits that I want. I need really fast hard drives to do as much audio as possible, and I don't put loads of other crap in them. PCs are much cheaper than Macs and you can build them yourself. You can exploit a PC in so many more exciting ways — you can get a lot more interesting free software, and you get a lot more control over what you're doing. I've got a Soundblaster Live soundcard in the PC, and I'm getting better sound these days because I don't have to use the sound outputs from the computer any more, I just run the optical cable from the computer into the DAT machine, and then always listen through the converters on the DAT machine, which sounds better. I don't bother having any compressors or EQ here, because when I get to the mastering stage, they're always going to have better shit at the mastering place than I can afford. Having said that, on my last album we didn't change anything much, we probably added a very little bit of treble here or there, but otherwise the sound on the record is pretty much identical to the sound of the finished product in here."



Audiomulch is Kieran Hebden's main improvisational tool.

Checks & Balances

An example of the balance between planning and improvisation that goes into a Four Tet track is the centrepiece of *Rounds*, 'Unspoken'. "What happened with 'Unspoken' is that I had the piano and the drums on my laptop, and I was on the Eurostar coming back from France, and I literally listened to that loop for the whole two-and-a-half-hour journey back. I got back and didn't do anything with it for a couple of weeks, I knew I had it sitting there, and then I just sat down for two days and just did the whole thing. For the middle part I had the idea very clearly in my head about what I wanted to happen, with this big grand bit in the middle, and it got really insane with 40 tracks of audio going — there were three drum kits and loads of percussion and all sorts of stuff.

"I laid out the arrangement first, with where the piano was going to go and everything, and than for the bit in the middle there's all these electric pianos, and I probably took sequences of one or two notes from about four or five records with bits of electric piano to make all the parts work. Everything else in I put in as I needed it. It was one of those tracks where it was like 'It doesn't lift quite enough. I'll add another shaker. It's not quite bassy enough. I'm going to put a sine wave underneath it.' I think a lot in those terms when I'm working on something — what does this need at this point? — and I use a lot of really simple devices, things like sine waves and shakers and cymbals, that you need to give the track the right dynamic. To pull off a track like that and make it lift in the middle and make it sound epic enough, it wasn't just one of those things where I was just putting together an improvisation in *Audiomulch*, I had to really think about how I was going to make it happen and listen to it again and again. I wanted the feeling that it would keep rising and rising, and whenever I played it back, if there was ever a point

Live And Dangerous

Unusually for a solo artist specialising in instrumental electronica, Kieran Hebden has developed a reputation for powerful live shows. "Especially with what I'm doing live, I'm really interested in the possibilities of real-time improvisation on computers," he explains. "I was touring a lot last year and that started to influence what I was doing in the music. My live shows are way more glitchy and aggressively mashed-up than the stuff I'm doing on the record. I allow the record to be more subtle, because it's something you want to be able to listen to again and again, whereas the live thing I'm quite happy for everything to be mad and distorted and out of time, because I'm just trying to capture a moment.

"I've got two laptops for live use. They each run different programs, and I've got a Pioneer DJM600 DJ mixer, which has got a sampler on it as well, and effects and things. On one of the computers I run *Audiomulch*. I use that to run a lot of the drum sequences, but the thing about *Audiomulch* is that you can set up any sort of studio situation you want, so I might have a channel running and it's got like a programmed drum sequence from one of my tracks, but it's going into a mixer, and also running through a couple of granulators, and stuff like that, and I've got faders on them so I can bring in all these mad glitchy sounds.

"On the other computer I just run two or three instances of *Cool Edit*. You can play *Cool Edit* a bit like a musical instrument if you put it on a loop setting — you can grab little bits of the file and loop them, so all the melodies and main parts from the album I just have on loops, and I can kind of play them. Stuff can be going in time and I can be manipulating it with one computer, and then I can play all the noises and melodies over the top. No two shows are ever remotely the same, they're all totally different, and sometimes even I'm completely surprised at what ends up coming out. I've got the sampler on the mixer, and if something good's going on, I can take a sample of it and keep it looping and then start doing other variations at the same time. So, effectively I've got three things going on at the same time, all with different possibilities."

where it felt like it was losing its momentum I'd know that I needed to put something else in there again."

Even where something sounds completely spontaneous, it may be carefully planned, like the free-jazz wig-out drumming on 'And They All Look Broken Hearted'. "That random drumming was actually a really laborious *Cakewalk* process that took hours, taking loads of sounds and breaking them up and trying loads of different placings. I wanted to have the feeling that someone was just knocking around a drum kit, and what I really like about that track is that the drums go along in this random way, and it's building up and you expect the drums to kick into a beat, but then another kit comes in, and it's that other kit that suddenly kicks into the beat. The idea was that it was suddenly going to take off, but it wasn't going to be the obvious drum kit — another drum kit would do it."

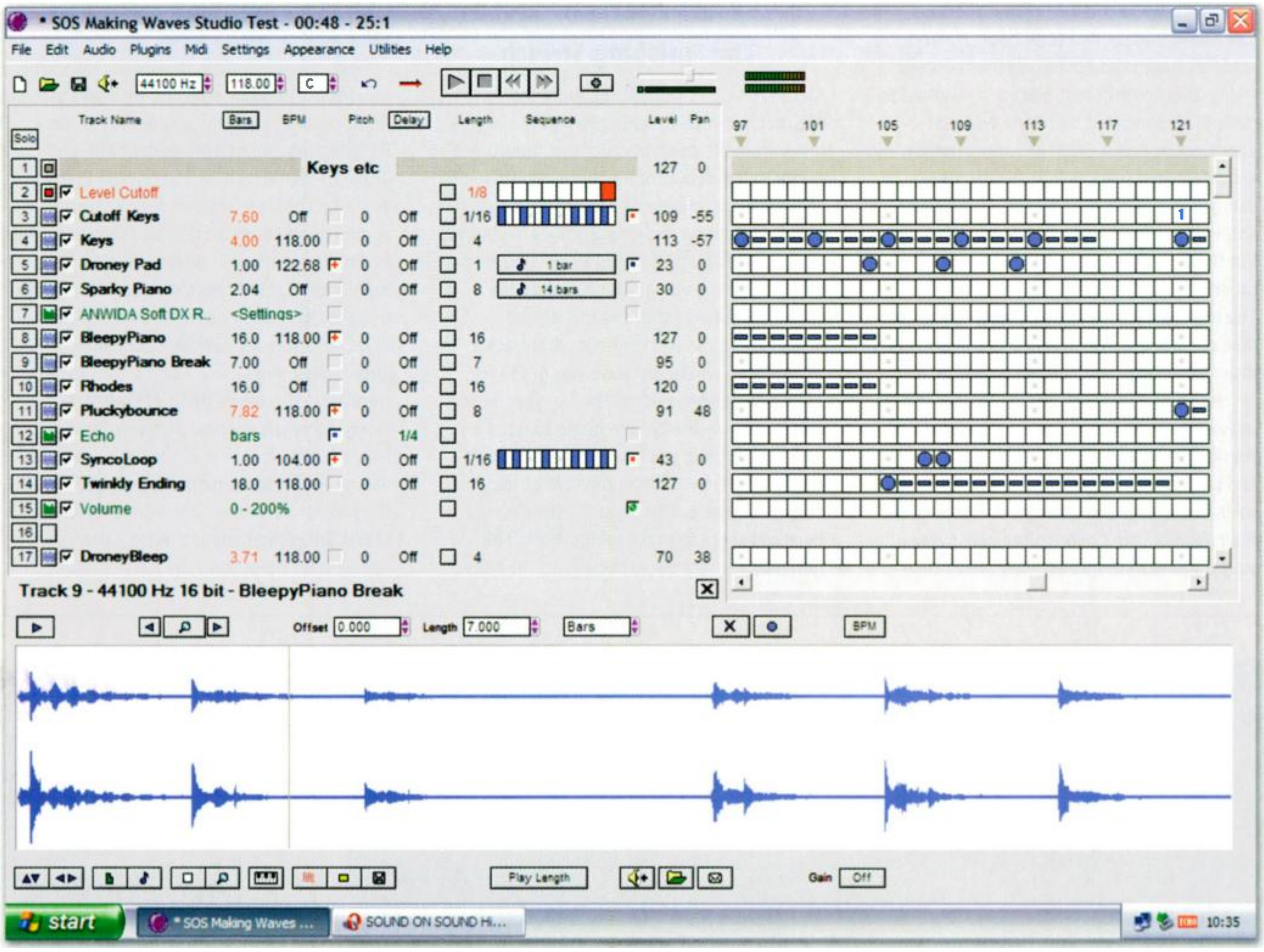
Here and elsewhere, the melodic elements provide a repetitive bed while the detail and musical development comes from the drums and percussion. This deliberate reversal of roles is one of the characteristic features of *Rounds:* "You can't over-complicate every single sound in the track — you have to decide which things are going to ride through the track in a simple way, and which ones the detail is going to be in. Everything I do has some sort of anchor in there that holds it and keeps going, in a way. Even though there's a harp melody that's the main hook, you probably end up listening more to the drums — they become the main focus."

The Finishing Stretch

Kieran Hebden's thoughtful, ideas-led approach to music making is visible at every level, from his conception of the 'folktronica' genre to the details of his drum patterns. It's also apparent in his concern for the neglected art of structuring and sequencing an album to work as an album: "I like to make albums that you play all the way through. Before the album's finished I start organising the sequencing, and working out what track's going to go where, because you get to the point where you're doing the last few tracks and you know that you're going to need a certain energetic track, or it needs another slow track. I'm a bit manipulative at that stage, I suppose. I start using little devices here and there to make things work, like non-musical sounds, especially as the music is instrumental as well. I use a lot of non-musical sounds to try to set the scene. You can just put in the sound of the sea for two minutes before a track starts and it sets the scene, it gets you prepared to listen to the music. The fades and how the tracks end is so important. The album is quite eclectic, and it takes a bit of time to make it work at the same time as being eclectic."

With *Rounds* now in the shops, Hebden is moving on to other projects. He's taken on his first production job for another artist, on a forthcoming album from Beth Orton, and is also in demand as a remixer, while in the live arena he's touring with Prefuse 73 and supporting Radiohead this Summer. Whatever he does next, though, ideas will always be more important to him than gear. "My whole thing is that I don't use a lot of equipment, I just keep it really simple. Good music's about ideas, really, and I don't want to get trapped in the whole thing of constantly worrying about my new software, and learning how to use it all the time. I feel I've mastered the equipment I've got, and I'm at the point now where I don't have to think about the equipment any more. I can concentrate on making the music." SOS

Making Waves Studio



Mike Bryant

Loop-based Music Production Software For Windows

Making Waves have been developing their eponymous looping software to include new features such as VST Instrument support and high sample-rate recording, while retaining its core virtues of simplicity and low hardware requirements.

Loop-based music programs have become enormously popular since Sonic Foundry's *Acid* burst onto the scene back in 1998, and it's not difficult to see why. Automatic pitch and tempo matching is hugely convenient for fans of instant gratification, but it also offers those who prefer more variation in their music a great way of getting a track off the ground, even if you subsequently replace the original parts. Making Waves have been at this game for a while now and have been steadily developing their product range with a more professional feature set, culminating in the top-of-the-range *Making Waves Studio* reviewed here. For information on previous versions of *MW* don't hesitate to take a look at John Walden's earlier *SOS* reviews, available on-line at www.sospubs.co.uk/sos/dec00/articles/makingwaves.asp and www.sospubs.co.uk/sos/aug02/articles/makingwaves.asp.

One of *Making Waves*' distinguishing characteristics is its ability to run on extremely modest hardware: all that's required is a Pentium I with a minimum of 16MB RAM and a stereo soundcard, and all versions of Windows from 95 to XP are supported. Copy protection has changed since version 3, and no longer demands that the installation CD be present in the drive when the program is started. A 21-day authorisation code is provided so users can

Bundled Goodies

Making Waves Studio includes over 5000 samples in WAV format across three CDs, including the *XG Waves* CD available separately for users of the *Audio* and *Pro* versions. I was a little disappointed that a fairly high proportion of the included sounds comprise single notes and hits, as opposed to loops, because I feel that loop manipulation is what the program does best. I also felt that Making Waves would be wise to expand their selection of loops further beyond the dance-type genres, because although the dance market is clearly where they are aiming, it comes across as a little sonically limited on initial inspection. *MW Studio* also bundles a number of freely downloadable plug-in effects and VST Instruments from the likes of MDA, Linplug, ReFX and FXpansion.

get started straight away, and Making Waves now use a challenge-and-response system requiring the software serial number and product code. The company also operate a lifetime free update and technical support policy for registered users.

Standing Waves

I'll just quickly recap the operational aspects of *Making Waves* here, but for a more detailed look check out John Walden's review of version 2 from December 2000. Little has changed regarding the program's basic functionality, and starting it up still reveals the same resplendently grey interface divided into a track pane, arrangement grid, and a browsing window for selecting and auditioning sounds.

There are four basic track types consisting of Single Play, Sample Loop, Percussion and Notes, which determine how an audio file is handled once imported into a project. Single Play, as the name suggests, is meant for one-off hits, effects, and other sounds that do not require time-stretching or pitch manipulation. Conversely, if the Sample Loop button is selected when auditioning audio files, the preview should automatically match the project tempo, making it easy to find musically complementary sounds. Once a set of loops has been added to the project they can be drawn, at a one-bar resolution, into the arrangement grid. Because it's not possible to manipulate samples with greater finesse than in ▶

Sound On Sound

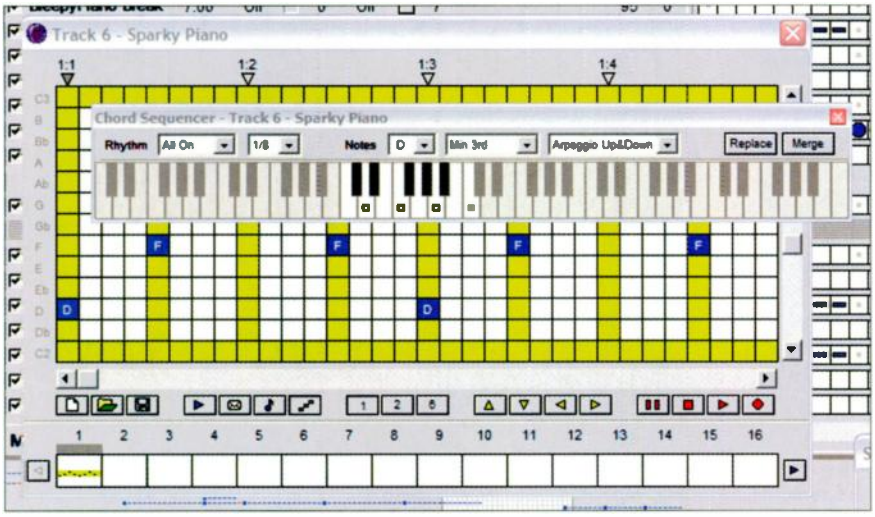
Making Waves Studio v4.7 £160

pros
- Combines good looping functionality with versatile rhythm and note sequencing capabilities.
- Effects automation with VST plug-ins works well and lends itself to creative experimentation.
- Low hardware requirements.

cons
- VST Instrument support limited and flaky as of v4.7.
- Not as easy to use as it could be due to idiosyncratic user interface design.
- Manual could be better (but at least it's not PDF).

summary
Making Waves Studio has an interesting mix of features, some of which stand up well in comparison to some much more expensive programs, although some of its rough edges need smoothing off. Prospective users would well to spend a little time with the demo to ascertain whether or not *Making Waves Studio* suits their way of working better than similarly priced competitors.

MAKING WAVES *STUDIO*



Notes tracks are constructed in *Making Waves* using a piano-roll editor, with the aid of an arpeggiator and a chord generator if required.

single-bar chunks, 'cutoff sequences' can be inserted to rhythmically mute tracks, and it's surprising how intuitive this method becomes after a while.

Moving beyond loop manipulation, *MW* offers some interesting pattern and drum-arranging features to make use of single-shot hits and notes. Switching a track to Percussion mode reveals a miniature pattern sequence strip offering resolutions up to 1/32nd of a bar. As well as being a good way of programming quick drum fills and patterns, I found this feature handy for creating interesting rhythmic variations on loop-based material.

The rather more elaborate Notes mode accesses a more conventional piano-roll editor (see screen shot above), which allows sequences to be recorded in either real or step time, and includes a simple chord generator and arpeggiator. This is strictly non-real-time and simply adds the appropriate chords or arpeggiations to your sequence on clicking either the Replace or Merge buttons. A separate floating window can be used to fine-tune each note's velocity and various other parameters, and although each track can only contain one individual sequence, the 16-bar limitation in previous versions of *Making Waves* no longer exists. There are, however, a few caveats and constraints, which I'll cover in the Gripes And Moans section.

Since version 3 project navigation has been improved, with the addition of nameable markers and Section Tracks (*MW* parlance for groups). Floating palettes are available in the Zoom view, making it easier to jump from one section, bookmark, or tempo change to another. This is a welcome addition, since just about everything in *Making Waves* — every loop variation, effect, parameter change, group, and so on — requires its own track, and even a medium-sized project can soon amass over 100 (though they won't all be playing simultaneously, of course). If you don't make use of bookmarks or sections, getting around a large project can be tiresome, as the main arrange grid has only two size options and is not zoomable.

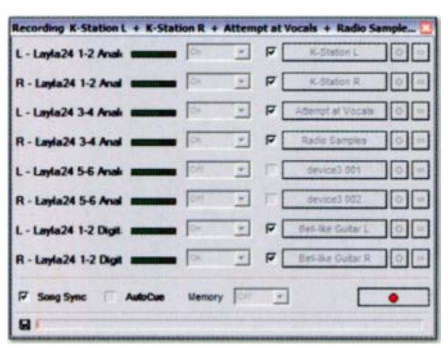
Test Spec

- *Making Waves Studio* 4.7.
- IBM Thinkpad 600E 366MHz laptop with 288MB RAM, running Windows XP Professional.
- Echo Indigo and Layla24 audio interfaces.

Making WAVs

Making Waves Studio is capable of recording eight tracks of audio simultaneously from up to three separate sources. The main new refinement added to the Record dialogue is the option to create a track to house the recorded file(s) with one button click, and it would be good if this featurette could be expanded to also add an event to the arrangement grid at the point recording began. I found recording audio alongside loop-based material to be a painless experience, and although you can't see a waveform overview of your files as a visual aid, everything sync'ed up and worked as expected.

The audio engine has been enhanced since v3 and can now play and record files at formats up to 32-bit, 192kHz, depending on hardware capabilities. I don't expect *Making Waves* will attract many users interested in running their projects at 192kHz or even 96kHz, since the trade-offs tend to outweigh the benefits, but more flexibility never hurt anyone. Those intent on using high sample rates and bit depths would do well to check if their audio hardware supports these modes in non-ASIO applications. I had to enable the Echo-specific 'Purewave' option on my Layla24 before recording would work at 24-bit, and any attempts to use 96kHz resulted in an 'unsupported wave format' message.



Making Waves Studio now supports the recording of multiple tracks simultaneously, at up to 32-bit, 192kHz.

Effective Studio?

The bundled native effects in *Making Waves Studio* are very simple, and most have only a mix level parameter available for adjustment. The quality is so-so: I wouldn't use the compressor, for example, but several others could come in handy. One particular highlight is the *Reverse* plug-in, which managed to make my most pedestrian guitar parts sound something like Ben Harper's solo on Beth Orton's 'Stolen Car'. I used it liberally. The best news is that support for VST effects has now been added in addition to Direct X, and although some third-party plug-ins proved a little temperamental, most seemed to work pretty well.

The effects implementation in *Making Waves Studio* is both simple and powerful, providing you are prepared to put out of your mind those hardware metaphors that predominate in other programs and get into the *MW* way of thinking. In a nutshell, insert-type effects are placed into their own sequence-grid track directly beneath the loop or sequence they are intended to process. Section tracks are used to process several tracks at a time, and 'master'-type effects placed at the very top of the track window. This may sound confusing — and at first it is if you're used to the conventional way of doing this — but I found I could achieve much of what I wanted, albeit with a little less flexibility than on an orthodox send/insert system. I would suggest, however, that *MW* lends itself greatly to creative experimentation in the use of VST plug-ins, due to ease of automating effects in interesting, rhythmic ways. Creating a new VST Parameter track beneath a plug-in allows you to automate levels in either whole-bar,

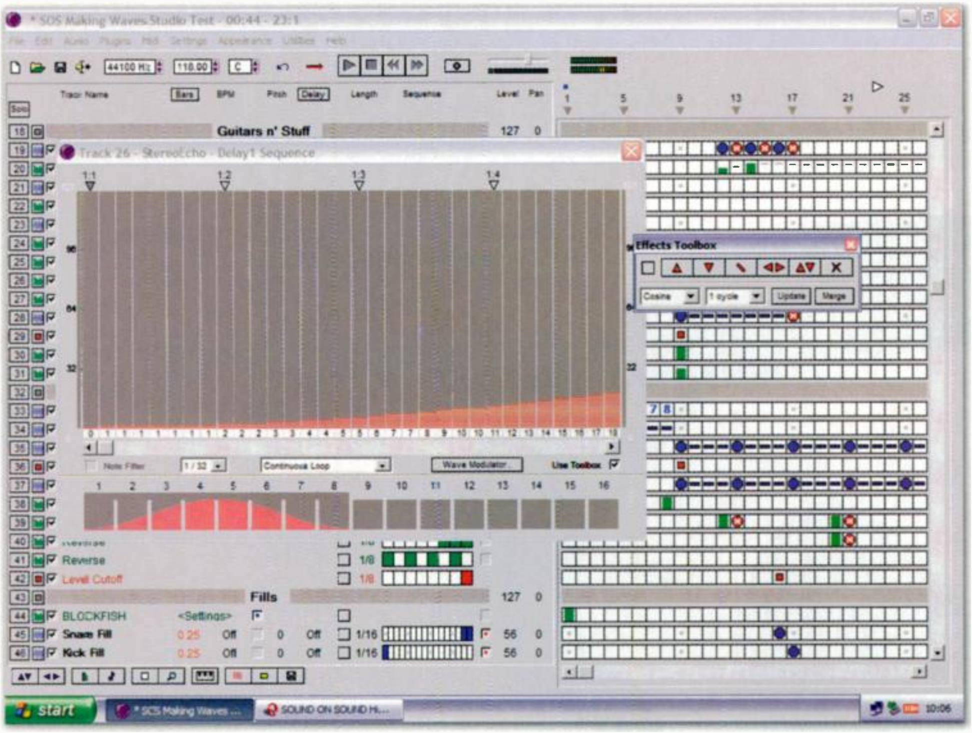
pattern or sequence mode, the latter allowing you to draw parameter automation down to 1/96 bar resolution using a variety of wave-shaping and manipulation tools. My only regrets concern the lack of any preset management system, and the fact that plug-ins' GUIs do not update to reflect automated parameter changes, so you won't get that pleasing sensation of watching dials and faders move all on their own...

Gripes And Moans

I found a few aspects of the program's design rather unpolished and somewhat frustrating. The addition of VST Instruments, for example, has great promise — particularly as they can be automated in the same manner as VST effects — but it doesn't seem to work too well as of version 4.7. Most annoying is the fact that notes played or recorded in the sequence window can currently only be held for the duration of one bar before the sound is abruptly cut off, which makes organs, pads — anything but drums, in fact — somewhat incapacitated. Whilst this limitation remains it's difficult to recommend *MW* as a fully fledged host for VST Instruments, and I also encountered a few stability issues with some third-party VSTis which I hope will be addressed in future updates.

At a more general level, I found recording and editing MIDI sequences something of a chore due to the lack of any visual indication of a note's velocity and length without viewing its properties in the Note Settings panel. Furthermore, as the piano-roll grid is fixed at 1/32nd of the bar, you can only fine-tune note positions by adjusting their offset — not exactly the most intuitive way of editing MIDI. While we're on the subject, I'd better mention that there's no concept of lassoing and dragging notes around, and although you do get accustomed to it after a while, *Making Waves*' method of selecting, cutting, and pasting notes is initially somewhat wearisome. The MIDI side of the program — little changed from version 3 other than the option to sync external equipment via MTC as well as MIDI clock — is still rather hampered in requiring instrument definition files to be available before sounds can be used in a project. There is still no option to sync *Making Waves Studio* to a more full-featured MIDI sequencer.

Unfortunately, the manual — though thankfully made of paper — isn't a great deal of help in casting light on the program's more challenging operational aspects, and is merely duplicated in the on-line help. It's adequate at informing you what a given feature is capable of, but poorer at actually explaining how to go about using it. Another annoyance, noted by John Walden in his review of version 2, is the cryptic nature of the icons that make up much of *Making Waves*' interface, and these have not been made any less obscure in v4.



To use an insert effect in *MW Studio*, you simply add an effects track beneath the track to which the effect is to be applied. If you want to automate the effect, a VST Parameter track can also be added.

Royal Wave?

Making Waves Studio is undoubtedly a powerful product with a lot of good features for the money. In fact, there's nothing with quite the same blend of loop manipulation, pattern creation, audio recording and MIDI sequencing at a comparable price, and it's capable of excellent results on very modest hardware. In an age where the recommended specs for most audio sequencers are approaching the point when they could take on Gary Kasparov and win, it's nice to see something designed for those on a budget, or who are perhaps interested in recycling an old PC. The integration of VST plug-ins with full automation capability is a fantastic addition to version 4, and the enhancements to file-format compatibility and song navigation are definitely welcome.

On the down side, I do feel that the implementation of VST Instruments — one of the most exciting new features — is rather

Latency And The Like

***Making Waves* strikes me as something of a hybrid application with regard to real-time operation: On one hand, a pre-mixing system is employed (similar to that in *Cool Edit Pro*) to ensure the maximum efficacy on slow systems. VSTis and audio samples can be triggered via MIDI in either the main workspace or the sequence editor, but the sounds are pre-mixed and can only be played monophonically. To get around this you must select Recording Preview (for some reason represented by the internationally recognised symbol for 'pause'), which allows real-time triggering with the audio interface's latency by pre-loading the sounds into RAM. Although a little clunky, this works adequately in practice, even given the rather high 30ms latency managed by my Echo Layla24 in *Making Waves*. Overall, I'm inclined to see the manner in which *MW* pre-mixes songs as a mixed blessing, because although it does contribute to the program's very low hardware requirements, it tends to make adjusting parameters by ear rather difficult, as the changes take a bar or so to enact, depending on the song's complexity. This means that one of the principal benefits of low-latency soundcards — a more 'tactile' and responsive mixing environment — does not apply to most aspects of *Making Waves*.**

Another new feature is the ability to host VST Instruments.

half-baked, particularly with regard to the note duration limitation. The other big area ripe for improvement is the general usability of the interface, especially with regard to the more obscure aspects of the sequence editor, which I found to be the biggest obstacle to exploiting the program's considerable potential.

Making Waves Studio is an attractive proposition if you follow the upgrade path from the *Audio* and *Pro* versions, but I think musicians with good PC hardware and the desire to transcend the dance-type genres would do well to check out the demos of some more conventional sequencers before plonking down their cash. Cakewalk's *Sonar* 2, for example, can now be had for under £200 and includes some powerful looping features and a much better MIDI environment. Taking into account discounted shop prices, however, not to mention the cheaper downloadable versions and generous free update policy, *MW Studio* looks like a much better deal and is definitely worthy of consideration. SOS

information

£ **Making Waves Audio £30 (download), £39.99 (boxed); *Audio Pro* £60 (download), £89.99 (boxed); *Studio* £110 (download), £159.99 (boxed); prices include VAT.**

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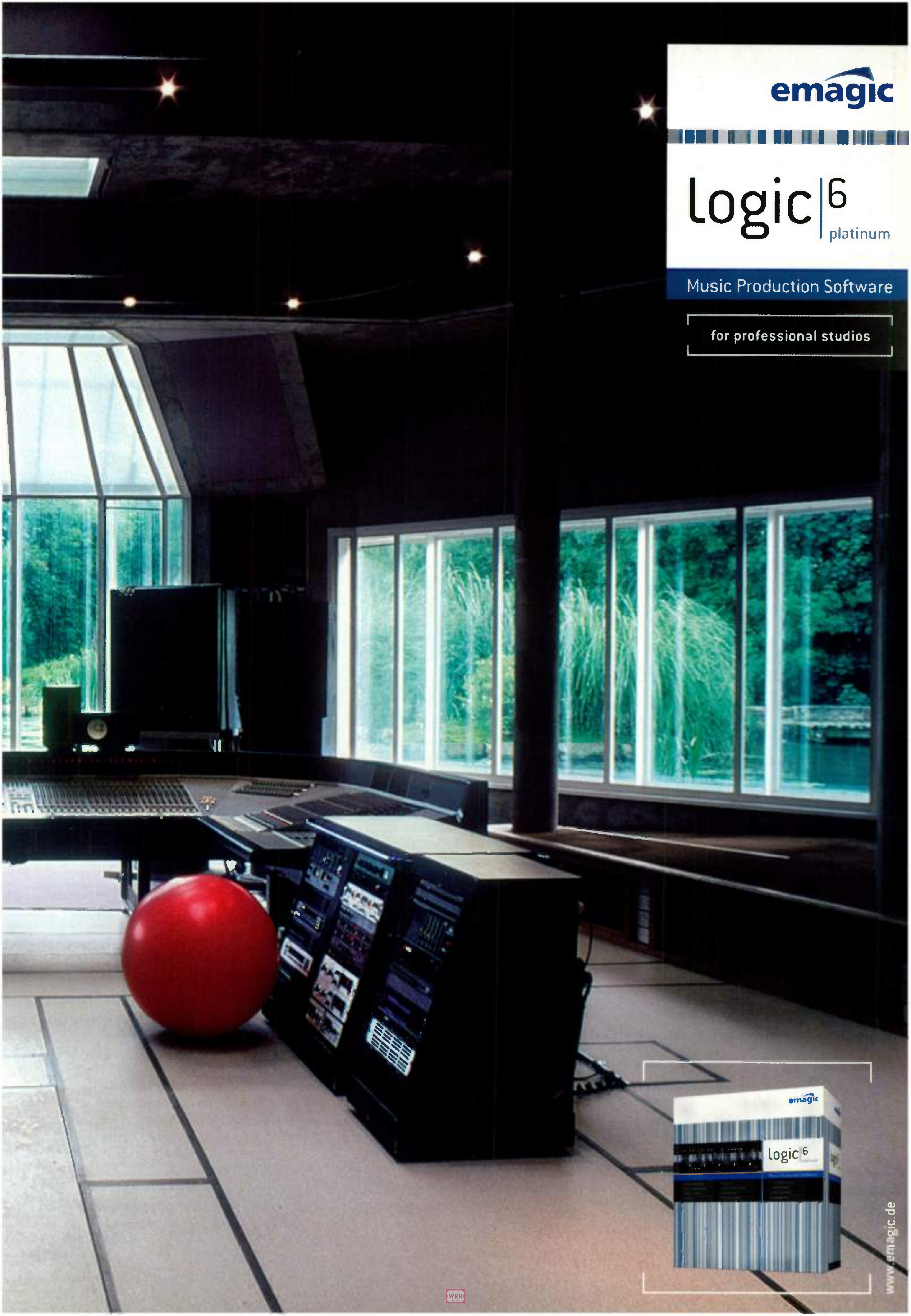
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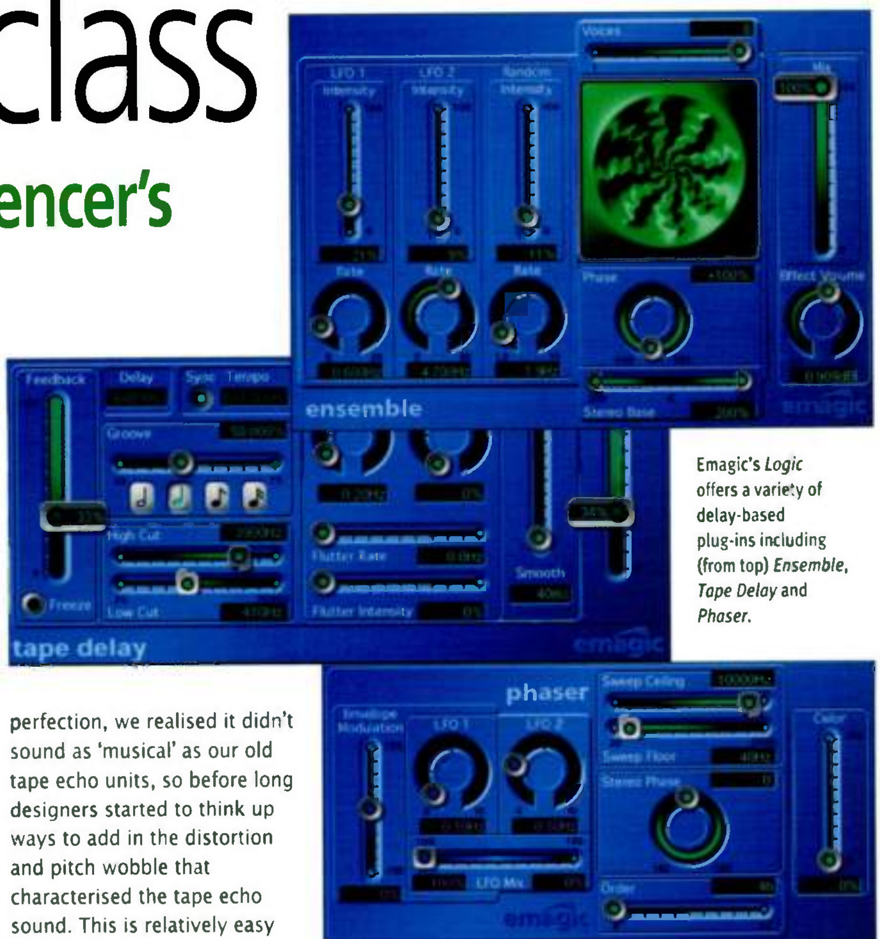
Sequencer Delay Masterclass

Using Your Sequencer's Delay Effects

Although delay can be one of the most simple studio effects, software delay plug-ins often provide a bewildering array of options. So we take a look at what all the sliders and switches do, and provide advice on how best to use them.

Paul White & Martin Walker



Emagic's *Logic* offers a variety of delay-based plug-ins including (from top) *Ensemble, Tape Delay* and *Phaser.*

Today we're exposed to so many sophisticated studio effects, both as hardware and in the form of plug-ins, that digital delay can seem a little tame, but in reality it's still one of the most commonly used effects after reverb. Until the invention of the tape-loop echo box, delay meant nothing to a musician other than a drummer turning up late, but once it had appeared there was no going back, and the effect soon became part of pop culture. Ironically, we all used to complain about tape echo units — they never ran at a steady speed, so the pitch wavered, and the delays weren't as bright or as clean as the original sound, so the repeats became progressively more distorted and distant.

After a brief flirtation with analogue 'bucket brigade' echo devices (which sounded rather dull), the DDL (Digital Delay Line) arrived on the scene, and finally delivered what we'd all been asking for — rock-solid, clean repeats. The trouble was that once we'd got this degree of technical perfection, we realised it didn't sound as 'musical' as our old tape echo units, so before long designers started to think up ways to add in the distortion and pitch wobble that characterised the tape echo sound. This is relatively easy to do in the case of plug-ins, where extra controls don't add to the manufacturing cost. Another thing that plug-ins do well is synchronise to the tempo of the host software, so setting up delays that run exactly in time with the music has never been easier.

The Principles Of Delay

The concept behind a digital echo device is quite simple. A signal is passed through a memory buffer, where it is delayed for a short time and then sent to the output. Mixing the delayed sound with the original signal produces a single repeat following the original sound, but feeding a percentage of the delayed signal back to the input produces a repeating echo effect, where each subsequent echo is a little quieter than the previous one. The feedback gain must be less than unity, however, otherwise the echoes will build up in level rather than decaying, resulting in an uncontrollable howl. A popular dub trick is to keep adjusting the delay feedback level so that it starts to build up, then, just as it threatens to get out of control, the feedback level is reduced again. Plug-in automation makes this very easy to do using plug-ins.

The next level of sophistication is to provide two or more outputs from the delay line at different delay times. A multi-output delay line is said to be multitapped, because the signal is 'tapped' off at different points in the delay buffer. A delay line with three or four taps is better able to emulate the old tape echo units, most of which had three or four playback heads, and by applying feedback (which was usually based on a mix of all the heads), the complexity of the delayed signal becomes far more intense, as the different delay times lead to a more chaotic build-up of repeats.

While tape delays were invariably mono-in/mono-out devices, software delay effects can be made either mono or stereo, and the different tap outputs can be panned to different positions to create a wide, stereo effect from a mono source. The so-called ping-pong delay takes this concept one step further by using two delay lines, one fed to the left output and one to the right. The feedback from the right delay line is returned to the left input and vice versa, so that instead of the echoes always coming from the same pan position, they appear to bounce from side to side as the repeats come alternatively from the left and right outputs. Ping-pong delay is a commonplace commodity in the plug-in world, but is no less effective for that.

In order to emulate the sound of tape delays, some plug-ins allow the feedback signal to be filtered, usually to reduce the high end, but it's not uncommon for both low and high filters to be available. There may also be a way to apply either regular or pseudo-random pitch modulation to the delayed signal to simulate the wow and flutter of an old tape delay. A further nicety not often provided is some way of adding distortion to the feedback signal so that the repeats become progressively duller and more distorted. The reason this sounds more musical is that less well-defined repeats don't conflict as much with the original sound, and as they become duller and less well-defined after several trips around the feedback loop, they appear more distant sounding.

Modulated Delays

In the early days of DDLs, before programmability became commonplace, everyone knew how to set up chorus, flange and phasing effects by modulating short delay times and applying different amounts of feedback, but today these effects all tend to come as separate plug-ins, so some newcomers to recording may be unfamiliar with these basic principles.

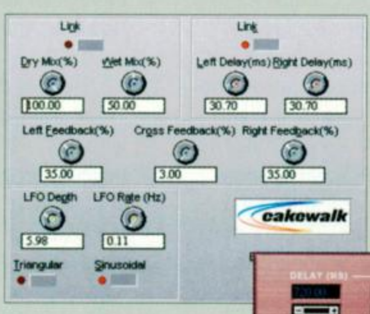
Perhaps the most common modulated delay effect is chorus. Chorus approximates the effect of two or more instruments playing the same part, and is frequently used on guitar and keyboard pad parts. It works using a delay of just a few tens of milliseconds, modulating the delay time at around 3Hz using an LFO (Low Frequency Oscillator). If delay time is modulated, the pitch of the sound is also modulated. The direct and modulated/delayed sounds should be mixed equally so that the listener perceives the differences in timing and pitch as being two slightly different renditions of the same part. Some chorus plug-ins use a multi-tap delay line, so that several pitch-modulated copies of the original sound are combined for a richer layered effect.

By further shortening the delay to just a few milliseconds and using only the delayed sound, modulating the delay time has the effect of producing a pitch vibrato. If the delay time is less than around 15ms, the listener won't usually perceive any delay, so most chorus plug-ins that have a mix control can be used to create vibrato simply by setting them to 100 percent effect. A general rule is that, the longer the delay time, the less modulation depth you need to create the same degree of pitch modulation.

It is a short step from vibrato to phasing, as all you need to do is reduce the modulation rate and add in the dry signal. The concept is similar to chorus, but because the delay time is much shorter, there's no audible delay. Instead, the two signals combine such that some frequencies add while others cancel, producing a comb-filtering effect, and these frequencies ▶

Delay Effects In Cakewalk *Sonar*

Like *Cubase*, *Sonar* has a set of older and rather more basic-looking delay plug-ins for compatibility purposes. *Flanger*, *Chorus*, and *Delay* are identical in appearance, comprising separate Left and Right Delay length controls with an optional Link switch, and Left and Right Feedback controls, along with Cross Feedback for ping-pong effects. An LFO is provided for modulation duties, with Depth and Rate controls and a choice of triangle or sine waveforms, and the plug-ins are completed by Dry and Wet Mix controls with an optional Link switch and an overall Bypass switch.



The only difference between the three, apart from the included presets, are the delay ranges: the *Flanger* ranges from 1-20mS; the *Chorus* from 20-80mS; and the *Delay* from under a millisecond all the way to five seconds. Despite their bland appearance, all three are surprisingly capable. The three feedback controls make the *Delay* very versatile, while the LFO is very useful on longer delays for radical echo pitch-bend effects. The *Flanger* can provide a wide range of through-zero effects, although beware of trying out extreme feedback settings, since they can cause a sharp crack through your speakers followed by silence (mercifully this is a plug-in fault rather than cone damage). The *Chorus* can add anything from subtle thickening to stereo spreading, and all three plug-ins provide a useful set of embedded presets to get you started.

The same three effects are also handled by newer offerings from the DSPFX range, this time with much smarter graphics and automatable controls, although once again the interfaces are surprisingly similar apart from a choice of green livery for the *Fxflange*, blue for *Fxchorus*, and red for *Fxdelay*. Sadly, they also have a similar approach to the EQ that I described in May's EQ masterclass — each 'voice' has an overall gain control, and can be separately enabled, but the remaining controls are shared, requiring you to activate the appropriate voice select button before altering them. While this provides an incredible amount of control, and avoids having 20 to 30 controls on screen, it's convoluted and difficult to use since you can only ever see the current values for one voice at a time.

The *Fxflange* is the simplest, providing two voices, each with its own gain control and on/off switch, along with a select button to access the four extra sliders alongside, controlling Delay, Feedback, Pan, and Mod Freq. Having separate pan controls for each delay lets you adjust stereo spread, and this is a versatile plug-in with plenty of useful bundled presets.

Fxchorus provides Delay, Mod Depth, Pan, and Mod Freq for its four global parameters, and this time there are four voices for some really rich effects. However, this also doubles the frustration during editing, especially since you can't cut and paste settings between the voices. However, there's no denying that *Fxchorus* sounds good, and with 35 presets you may not have program your own sounds.

Fxdelay is once again capable of many versatile and good-sounding effects, but suffers even more from the interface, since there are both Course and Fine delay controls, and there's no read-out of the combined value. The lack of any tempo sync also makes setting up the four voices frustrating, especially as this plug-in is far more likely to require its delay controls adjusting to suit the tempo of each song. However, because the four voices have separate Feedback and Pan controls, they are capable of some clever multitap effects, such as stereo delay sweeps, as long as you have the patience to set them up.
Martin Walker

change as the delay time is modulated. Subjectively, the result is similar to mild flanging, but less dramatic. Phasing is popular for use on guitars and electric pianos, but may be applied to almost any instrument. Adding feedback and increasing the delay time slightly produces flanging, a heavily comb-filtered 'whooshing' sound born in the psychedelic era.

Delay Effects in Emagic *Logic*

Logic's simplest delay plug-in, *Sample Delay*, isn't so much an effect as a tool, as it adds delay in single-sample increments to either or both channels of a stereo signal. The linked mode can sometimes be useful for compensating for a delay elsewhere in the system, whereas the independent left/right mode is handy when you need to get the two halves of a stereo track back into line or where you're combining a mic and a DI signal and need to get the best subjective result when they combine. As *Logic* doesn't support sample-accurate audio track editing and positioning, this plug-in could be used to get around some of the limitations this causes by allowing timing offsets to be adjusted in single-sample increments.

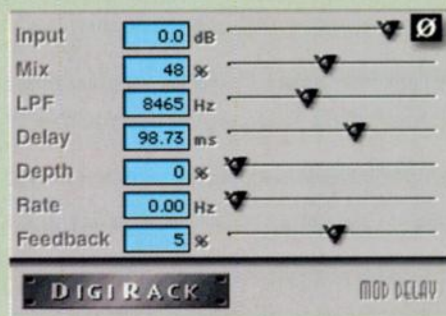
The next delay is *Stereo Delay*, which has independent delay times for the left and right signal paths, as well as two sets of feedback controls, one operating normally and one set feeding into opposite channels for creating ping-pong delays. Sliding high-cut and low-cut filters are available in the feedback path, and delay times can be directly (and independently) sync'ed to one of four note values and then further modified via the Groove slider for tempo-related effects. This latter control has sufficient range to allow triplet delays and other multiples to be set up.

Logic's *Tape Delay* is based on a single repeating echo, again tempo related and with high-cut and low-cut filters, but this time further controls have been added to modulate the pitch of the delayed sound. An LFO-controlled modulation allows gentle chorus effects to be applied to emulate the 'wow' of a worn tape transport, while Flutter Rate and Flutter Intensity simulate the higher-speed pitch fluctuations that would have been caused by worn capstan bearings or worn tape. Used carefully, this comes very close to the original tape echo sound, but the fact that there's only one 'head' means you can't recreate some of the classic guitar effects that were originally produced using three-head and four-head tape delays. However, you can get close to it by putting a *Tape Delay* and a *Stereo Delay* in series.

The final *Logic* delay is the *Modulation Delay*, a single-tap, 'feedback and mix' affair with two LFO modulation oscillators that can be mixed in any combination and set to run at independent rates, where the overall modulation amount is controlled via the Width slider. The delay time is manually adjusted — no tempo sync this time. A further slider is included for volume modulation, so that tremolo can be incorporated into the effect, plus there's a button that switches between conventional modulation and a special mode that is designed to remove the obvious pitch fluctuations that delay modulation normally causes. Additionally, there's a separate Flanger/Chorus stage where one knob morphs between chorus and flange effects while the other sets the stereo phase of the effect, allowing the left/right sweeps to be aligned or delayed relative to each other for more stereo interest. The result is a delay with a chorus/ensemble character that works particularly well on electric guitars,

Delay Effects In Digidesign *Pro Tools*

The Digirack II plug-in package bundled with Digidesign's MIDI + Audio sequencer software doesn't contain individual plug-ins for delay-based effects such as chorus, phasing and flanging. It does, however, include a selection of well-equipped *Mod Delay* plug-ins, which can be used to create all these effects as well as conventional delays. *Short Delay, Slap Delay, Medium Delay* and *Long Delay* used to be supplied in both TDM and native versions, depending on what kind of Pro Tools system you had; from *Pro Tools* v5.3, all run in the native HTDM or RTAS formats, but none of them use a great deal of processor power.



All are available in mono, mono-to-stereo, multi-channel and multi-mono versions, the stereo versions offering separate controls for the left and right delay paths.

In each case, the settings on offer begin with an input polarity switch and an input gain control — if your patch uses a lot of feedback, the latter can help prevent overloads. Following this is a conventional Mix control and a cutoff control for a simple low-pass filter, which allows you to darken the delayed signal to create 'tape echo' effects. After this come the Delay time slider, modulation Depth and LFO Rate controls, and the Feedback setting.

The only difference between each of the *Pro Tools* delay plug-ins is the maximum delay time available, which means you have to choose the right plug-in for the particular delay-related task you have in mind.

For a detailed explanation of what you can achieve with all the different plug-ins, including how to create all the conventional delay-based effects, check out Simon Price's Pro Tools Notes column from *SOS* November 2002. *Sam Inglis*

The *Double Delay* and *Flanger* plug-ins are just two of around ten delay-based processors available within Steinberg's *Cubase SX*.

and that can give mono sources a spacious, stereo feel. If you want to experiment creating modulation effects of your own, this is the plug-in to try first.

Logic also includes a number of modulation effects, though here I'll be concentrating only on its modulated delay effects, which excludes the rather wonderful *Tremolo* and *Spreader* plug-ins. *Logic* offers one really simple *Chorus* effect, with only width, intensity and speed controls, plus the more spectacular *Ensemble*. *Ensemble* is a multitap chorus offering up to eight voices controlled by two LFOs and a random signal generator, where the contribution and modulation rate of each of these three sources can be adjusted using separate sliders. Stereo Phase and Stereo Base controls allow the width of the effect to be adjusted (again by offsetting and panning the left and right modulation sweeps).

By contrast, *Logic*'s *Flanger* is extremely simple, but it sounds no less effective for that. It has a simple four-slider interface with controls for mix (which is normally left set at 50 percent for both chorus and flange effects), Intensity (modulation depth), Speed (modulation rate) and Feedback. That leaves only the *Phaser*, which is a little more elaborate than usual in having two combinable LFOs, plus frequency sliders to set the limits of the sweep range. There is also an Order control to emulate the number of phase-shift stages employed, where more stages produce more notches in the frequency response. Again the familiar Stereo Phase control puts in an appearance.

The *Karlette* plug-in within Steinberg's *Cubase SX* provides a powerful recreation of a multi-head tape-delay unit.

Delay Effects In Steinberg *Cubase*

Three simple delay-based plug-ins were originally bundled with earlier versions of *Cubase VST*, and are still available as an option when installing *Cubase SX* if you need them for compatibility with older songs. All feature the generic 'one slider per parameter' interface, but *SX* users don't seem to get any presets for them. However, if you save the default banks from within *Cubase VST* you can then load them into the *SX* versions.

Stereo Echo provides separate delays for the left and right channels, each with its own feedback control, plus a link control to turn it into a mono delay. *Choirus* occasionally suffered from clicks and distortion, and was later replaced by *Choirus2*, but as far as I can see both are now identical in their SX incarnations, and are capable of a wide range of pleasing chorus and flange effects. Without tempo sync and graphic interfaces all three of these plug-ins are much trickier to set up than their replacements, but in their favour they do use a tiny amount of CPU.

Along with VST Instruments, tempo sync arrived with the VST 2 specification, first seen in 1999 on *Cubase VST* v3.7 on the PC (and v4.1 on the Mac). However, it wasn't until the following year that Steinberg managed to get the Mac and PC platforms 'in sync', with the release of *Cubase VST* v5.0, and this also saw a new 'modernised' graphic look and a bundle of fourteen new plug-ins.

These have once again been updated for *Cubase SX*, and include *Double Delay*, which, as its name suggests, offers two delays, each of up to ten seconds. Each has a pan control, plus optional tempo sync. When tempo sync is activated you get straight, triplet, and dotted options from 1/32 to 1/1

Using Delays

Delays and echoes are useful both on vocals and instruments and, in most cases, setting the delay time to a multiple of the song tempo helps reinforce the rhythm of the song. Tempo sync facilities make this very easy, but be aware that any tempo changes in a song can cause tempo delays to hiccup quite badly, so it may be best in such cases to split the part into two tracks at the tempo change point and use two different delay plug-ins set appropriately to the tempo of each section. You can use the delay sync mode to establish the delay time value needed for each section, then flip to manual mode and enter the time values you noted down. By working in manual mode, you won't get any unpleasant surprises as parts hanging over from the first section spin 'off pitch' as the delay time suddenly switches. Some of Emagic *Logic*'s delays include a smoothing parameter hidden away in the extra controls section (press the button with the binary numbers in it) designed to make the delay time change more progressive when tempo changes are encountered, but in practice, this just makes the transition take longer and doesn't make it sound any less messy.

One very common production trick is to add echo only to certain words or phrases in a vocal part, which is best accomplished by turning up the echo or aux send control just before the phrase occurs and turning it down again immediately afterwards. Using your sequencer's automation capabilities, this can now be done very precisely.

Many rock singers use a short, slapback delay, created by setting a single repeat and then adjusting the delay time so that it follows immediately after the original sound, but not so closely that it blends into it. Artists such as John Lennon and Elvis made extensive use of slapback delay, but it also works well on some guitar parts. Artificial Double Tracking (ADT) is also based on short delays, but with a little added pitch modulation. It isn't as good as tracking the part twice, but it is a useful effect nevertheless. In most cases, some reverb would also be added to delayed vocals.

Chorus creates a lush, full sound, but it also has a habit of making the processed sound sit further back in the mix, which is why it's so often used on synth pad and acoustic guitar parts. I don't like the sound of chorus on vocals, though flanged vocals can work in certain circumstances. Chorus is best suited to string parts, keyboard pads, electric guitar and electric fretless bass where multi-voice chorus plug-ins produce the most complex textural results. In the early days of recording, all kinds of tricks were used to make the chorus effect appear to be in stereo, but with many modulation plug-ins, you get a stereo output automatically, where the phase or width control regulates the stereo spread of the sound. *Paul White*

for the base note value, and then a separate note value multiplier from x1 to x10. It's a bit difficult to get your head around the first few times, but things soon drop into place, especially with the dragable graphic display of pan and multiplier settings.

Mod Delay only provides a single delay, with identical delay time and tempo sync options, but makes up for it by adding control over the rate of pitch modulation for the delay effect. You can use this to set up chorus and flange effects with short delay times (although this is easier using the dedicated plug-ins), but its main use is to add movement to longer delays.

Karlette is essentially a souped-up tape loop emulation with four delay channels. Each 'tape head' has its own set of controls, but a global Sync button lets you switch between a delay of up to 2 seconds, and note values ranging from 1/32 to 1/1, including triplets. Each head has its own Volume, Pan, Feedback, and Damp (low-pass filter) controls, and some quite complex effects are possible, although it's a shame that no 'wow and flutter' options are available.

The *Chorus*, *Flanger*, and *Phaser* are capable of some rich effects. *Chorus* has Delay and Frequency controls that are also displayed in a graphic window where they can be dragged by the mouse, and there are three modulation waveforms available — triangle, ramp, and pulse. In addition, the Stages knob can add up to three delay taps in *Cubase VST* v5, and four in *Cubase SX*, for a really rich multi-layered chorus.

The *Phaser* is somewhat more complex, with tempo sync options for its Rate control, along with an overall Feedback control, plus a handy Stereo Basis knob that controls stereo width. Left at its default 50 percent setting, this retains the width of the original signal (if stereo), while increasing it provides enhanced stereo separation and zero drops it right down to mono. Rate, Feedback, and Stereo Basis can also be set by dragging in the graphic display.

The *Flanger* also provides Stereo Basis and Tempo Sync options, and Delay times from 0-100mS, but is rendered far more versatile by the inclusion of the Shape Sync knob. At its middle setting this provides a triangle waveform, but by moving it to the left or right you can progressively modify its symmetry all the way to ramp down and ramp up respectively. This means you can sync flange sweeps to each note — especially effective with snares and percussion.

Two more delay-based plug-ins are to be found in the SX modulation folder. *Symphonic* combines a stereo enhancer and an auto-panner with a simple delay/depth/rate chorus stage, to provide some rich swirling motion effects. Finally, Paul Kellett's *Rotary* is a Leslie speaker simulation, and so doesn't strictly speaking fall into the delay category. However, since it's bundled with *Cubase* and provides a variety of chorus and flanging effects it's worth a mention, especially since it has loads of controls.

One thing I've found with all the various *Cubase* delays is that they continue after the song stops, which would normally make sense, to capture the echoes after the final note finishes. However, you may have to turn the plug-ins off and then on to permanently silence long feedback delays. SOS

Delay Effects In MOTU *Digital Performer*

Chorus can create effects from the subtle to the downright nasty, and also does duty as a vibrato plug-in. All the expected chorus parameters are present, with delay time modulation being controlled by the Tempo Lock section and the Depth knob. However, *Chorus*'s modulating LFO can't be set to anything other than a sine wave. Although a mono-in/mono-out *Chorus* is available, the stereo-out versions are more flexible, since wet and dry signals can be panned separately. Pan them wide apart for spacious stereo, or both to centre to achieve flanger-like effects. The 'through zero' option allows Chorus to apply delay to the dry signal as well as the wet, so at times the wet signal will actually be heard first. For vibrato, turn the Mix control to 100 percent and then use Tempo Lock and Depth to dial in as much as you need. Bear in mind, though, that greater modulation depth can be applied to more delayed signals, so the Delay parameter has a real bearing.

Delay is ideally suited to producing rhythmic stereo (and surround) ping-pong delays, with feedback paths both within and between its channels. Because of the very real risk of rampaging feedback, *Delay*'s most important control is probably its Panic button, which immediately silences its output and clears all the delay buffers. Extra flexibility is afforded by each channel's multi-mode filters - great for analogue simulations and more creative effects. Feedback can be produced both in phase (positive values) and out of phase (negative values). For more about this great plug-in see *Performer* Notes in *SOS* July 2002.

Flanger has many similarities to Chorus, but crucially does away with the wet and dry pan parameters, and adds that essential Feedback knob. Delay times of below 5ms produce 'classic' flanges, whilst nearer the maximum of 20ms, and with high depth and feedback values, various sci-fi effects can be teased out. The Mix control is quite important too, as it allows delayed signals to be mixed both in phase and out of phase for subtle differences in tone. More sonic variation is provided by the L/R In Phase button, too. When it's not selected, the delay time applied to one channel will be comparatively short whilst the other is long, and vice versa. It's like having separate, opposed flangers on each channel, and it sounds groovy (man).

Phaser is a simple plug-in, with the intensity of the effect set using the Depth knob, and the bandwidth of the resultant notch filters controlled by the Width knob. The Tempo Lock section, as ever, controls the speed of the modulating LFO, and there's an L/R In Phase button here too, for yet more psychedelic entertainment.

By comparison, *Echo* is quite complicated, though in essence what it does is very simple. It's a four-tap design with separate gains for each delay path, and a global feedback path. It's suitable for applying conventional delays using just one tap, or for setting up rhythmic patterns using all four. The global feedback even allows bizarre pseudo-reverb tails, or endlessly ticking echoes.

Finally, though not a dedicated delay plug-in, *Sonic Modulator* has a lot to offer, because it has a flanger-like Delay section preceded by a dedicated vibrato module. What makes it better than *Flanger* or *Chorus* is the open-ended modulation scheme, with multiple multi-pattern LFOs. A versatile filter, crossover and amplifier section just adds to the fun. For more on *Sonic Modulator* see *Performer* Notes in *SOS* October 2002. *Robin Bigwood*

Echo Indigo

Photos: Mike Bryant

PCMCIA Headphone Output For Laptop Computers

Your laptop computer probably already has a headphone output — but does it offer low-latency 24-bit ASIO, GSIF, MME, Direct Sound and Core Audio drivers? Echo's Indigo does.

Mike Bryant

Echo Audio's small but perfectly formed Indigo is a Cardbus headphone output device for portable computers with a PCMCIA slot, which pretty much includes all but some sub-portables and Apple iBooks. The Indigo's feature set is sparse, with no line input, and no digital or MIDI I/O. Unlike more expensive and full-featured Cardbus interfaces such as Digigram's VXpocket, it does not employ flying leads or a breakout box — the Indigo's useful bits are contained within an unobtrusive grey plastic widget adjoined to the credit-card sized adaptor itself, giving access to two 1/8-inch stereo output sockets (one at either side) and a rotary volume control. This means that regardless of where your laptop's Cardbus slot is located, there is a headphone socket pointing towards you. A blue LED shows you the Indigo's ready to go.

You may be wondering what this £100 headphone socket can do that the standard one on your portable cannot. Well, it's capable of playing back 24-bit audio at 96kHz, and Echo claim that it will thus give you superior sound quality for your CDs and DVDs. While this may be true, the big selling point for musicians will be the inclusion of ASIO 2 and GSIF drivers for low-latency operation. On paper, the Indigo would seem perfect for composing on the go with self-contained music software.

Installation

Echo's web site suggests as a minimum requirement a 500MHz Pentium III, though this seems to refer more to the optionally bundled *WinDVD* software then the card itself, and I would venture that for those already content with the performance of their audio software any Pentium II or Apple G3 should suffice ('Lombard' Powerbooks excluded). You will, however, need to be running a relatively recent operating system — Windows ME or higher for PC users and OS X for those on a Mac. Installing the Indigo in Windows XP Professional was straightforward, though perhaps not quite as simple as for a USB device. The driver setup program presents the option of installing 'professional audio support' (yes please!) before prompting you to power down your computer and insert the card. On startup the new device was detected by Windows, which ran the familiar Add New Hardware wizard. After selecting Install Automatically and OK-ing that annoying 'Driver not electronically signed' warning, the blue LED sprang pleasingly into life.

A quick look at Windows' Device Manager revealed the Indigo to be listed as only one device, which was sharing IRQ 9 with the USB controller. Those of us who have wrestled with resource conflicts in the past will reflexively find this cause for concern, but in the event it was not in the least problematic. On installation the Indigo becomes the default output device, so it'll play all of Windows' annoying chimes, bells, and 'tada's. A discreet Indigo control panel is added to your Start menu, containing the buffer size settings for the GSIF drivers and the option to spoof an audio input — a necessity for programs such as *Sonar* that refuse to work without one. This is disabled by default, which had me scratching my head for some time wondering why the Indigo had disappeared from *Sonar*'s audio preferences panel after installing a later driver revision.

In Action

Playing a familiar test song through my Sennheiser HD565 headphones made it clear, even to my less-than-golden ears, that the Indigo could boast significantly better quality than my Thinkpad's built-in Crystal Sound Fusion chip. The biggest factor in this comparatively increased fidelity was probably the headphone amplifier, which was capable of driving my studio headphones with far more 'oomph', yielding a much more

SOUND ON SOUND

Echo Indigo £100

pros
- Good subjective sound quality.
- Compact size and convenient design.
- ASIO 2, GSIF, Direct Sound, MME and OS X Core Audio drivers included.
- Very respectable low-latency performance with ASIO and WDM.

cons
- Windows ASIO drivers not capable of sharing the output with MME or Direct Sound programs.
- No support for Windows 9x or Mac OS 9.

summary
For those wanting to plug in their headphones and mix on the go, the Indigo provides excellent sound quality and compact design.

controlled bass and a crisper mid-range. By far the most startling improvement, however, was the increased definition of the stereo image, which I'll venture has a lot to do with the relatively low jitter of the D-A converter. It made it possible to place instruments far more accurately in the sound stage, and revealed problems with songs I was mixing.

By default the Indigo's ASIO drivers are set with a buffer of 512 samples, which equates to a theoretical latency of around 12ms. Playing software instruments in *Sonar* and the demo of *Cubase SX* felt very snappy indeed, and I only noticed any hint of lag when playing staccato bursts of fast-attack drum sounds — nothing that would bother me in practice. I was also able to go about my business glitch-free and happy at the lower buffer setting of 256 samples (6ms), though on my somewhat antiquated Thinkpad this did slightly increase the processor usage. For those who like to impress their friends, a 128-sample (3ms) buffer is available, though it wasn't practicable on my system. WDM performance in Cakewalk's *Sonar* was particularly impressive: latencies of 2.9ms were usable, albeit with a performance hit. The optimum setting on my PC for a 16-track song with a couple of DX Instruments appeared to be around 10ms, though I've no doubt that those with more powerful systems will be able to significantly improve on this. To put these figures in perspective, my

The Indigo provides two headphone sockets, allowing easy connection whichever way around your PCMCIA slot is fitted.

Thinkpad's Crystal Sound Fusion chip (with WDM drivers) could manage a 10ms latency only with a comparative CPU hit of about 25-30 percent, as measured by *Sonar*, and dropouts were very easily incurred.

Praise notwithstanding, my experience with the Indigo was not entirely free of troubles. One driver problem revealed itself upon waking my Thinkpad from Sleep mode, when the Indigo began replaying my 44.1kHz songs at 48kHz. Fortunately Echo's tech support was able to direct me very quickly to a driver revision that fixed this bug. I also noticed that the Indigo emitted stuttering, droning sounds when pausing clips played with Windows Media Player after having used an ASIO application. Thankfully, no such artifacts were apparent in any of the serious music creation programs I tested.

A less anomalous but more irritating problem is the Indigo's lack of ASIO drivers capable of sharing the stereo output with standard Windows interfaces. This means you can't run more than one audio application simultaneously if ASIO is used. It *is* possible to run two or more programs at the same time if they use standard Windows interfaces, such as Direct Sound or MME, but not, for example, *Cubase* and *Wavelab* using Steinberg's far more efficient ASIO protocol. It's extremely annoying to see the 'Audio hardware in use' message years after I thought it had been banished, and I hope Echo will fix this soon.

The Indigo And The Layla

I suspect the Indigo will attract a great deal of interest from those musicians who already use either the Echo Layla24 or Mona with a Cardbus interface — after all, both units are rackmount-format and not particularly portable. During this review I had the opportunity to test how these opposing ends of Echo's laptop range integrate with each other, based on the version 6.06 (latterly 6.07 for the Layla24/Mona) drivers. The good news is that the ASIO drivers are completely interchangeable. Once you have powered down, swapped the PCMCIA cards and restarted, your audio applications should not notice that you've changed audio interfaces, because both products appear to use the same 'Echo ASIO' driver. If you use *Cubase SX* and have record-enabled in a song last saved using the Layla24, for example, you will be greeted with a 'Record could not be enabled, because there is no input' error message, but this can be dismissed without any problems.

If, like me, your ideal scenario is one where you can simply put your laptop to sleep, swap the cards and go, you may be a tad disappointed. Currently the Layla24's power management seems to be less 'plug & play' than the Indigo's, meaning you will have to power your laptop down completely, exchange the two cards, and restart, to avoid device recognition problems. This certainly isn't a major difficulty, but if you're lazy and impatient like me, it's a bit of a chore. Similarly, if your laptop has more than one Type 2 Cardbus slot, you might be tempted to run a Layla24/Mona and an Indigo side by side. This would be handy, not least because it carries with it the bonus of shielding the Layla24/Mona's rather flimsy-looking Cardbus cable connection from getting damaged. At the time of writing, however, I can't recommend this approach wholeheartedly, as I experienced rather inconsistent results using the Indigo and Layla24 in parallel. For example, in WDM mode, *Sonar* 2.2 worked reasonably well using the Indigo's outputs alongside those of the Layla24 — but I say 'reasonably well' because the buffer size had to be increased significantly before glitching disappeared. Furthermore, without an 'Esync' connection between them and no S/PDIF on the Indigo, clock synchronisation could not be ensured. Conversely, *Sonar* 2.2 in ASIO mode would only work with one card at once, which necessitated highlighting the required I/O ports in *Sonar*'s 'Drivers' dialogue and restarting the application. On my machine, I could not get *Cubase SX* to work at all with both cards side by side.

As far as sound quality is concerned, I have not been able to distinguish any difference between the headphone outputs of the Layla24 and those of the Indigo, and I suspect they employ the exact same machinery. The major benefit with using the two products in tandem, therefore, is that you really can record your material in the studio, and tweak and mix it using headphones wherever you so please, with just a restart and a quick card swap to bother about.

In Conclusion

With the Indigo's high-quality headphone amp and low-latency ASIO drivers, musicians who prefer to compose or play live exclusively using soft synths and samplers might find that Echo's baby provides all they need. Using the Indigo's ASIO drivers in Propellerhead's *Reason* proved to be a wonderfully painless experience. With a UK MSRP of a penny under £100, the Indigo's closest rival in features is probably M Audio's Sonica — a USB device that also provides an optical digital output. Although the Sonica has greater surround sound capabilities, it currently lacks the Indigo's comprehensive driver support for professional audio applications, and for many the choice will depend on whether they have a free USB port on their laptop. Personally, I was most enamoured with the Indigo's unobtrusive Cardbus design, and the fact that it didn't add to the metres of cable I already have to lug around. Aside from my gripe regarding the current lack of true multi-client ASIO drivers, the Indigo is an excellent product at an attractive price, which could do wonders for the musician who yearns for a slimmed-down studio and the right to roam. SOS

Test Spec

- Indigo driver version: 6.06.
- IBM Thinkpad 600E laptp with Pentium II 366MHz CPU, 288MB RAM, running Windows XP Pro.

information

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Zoom MRS802

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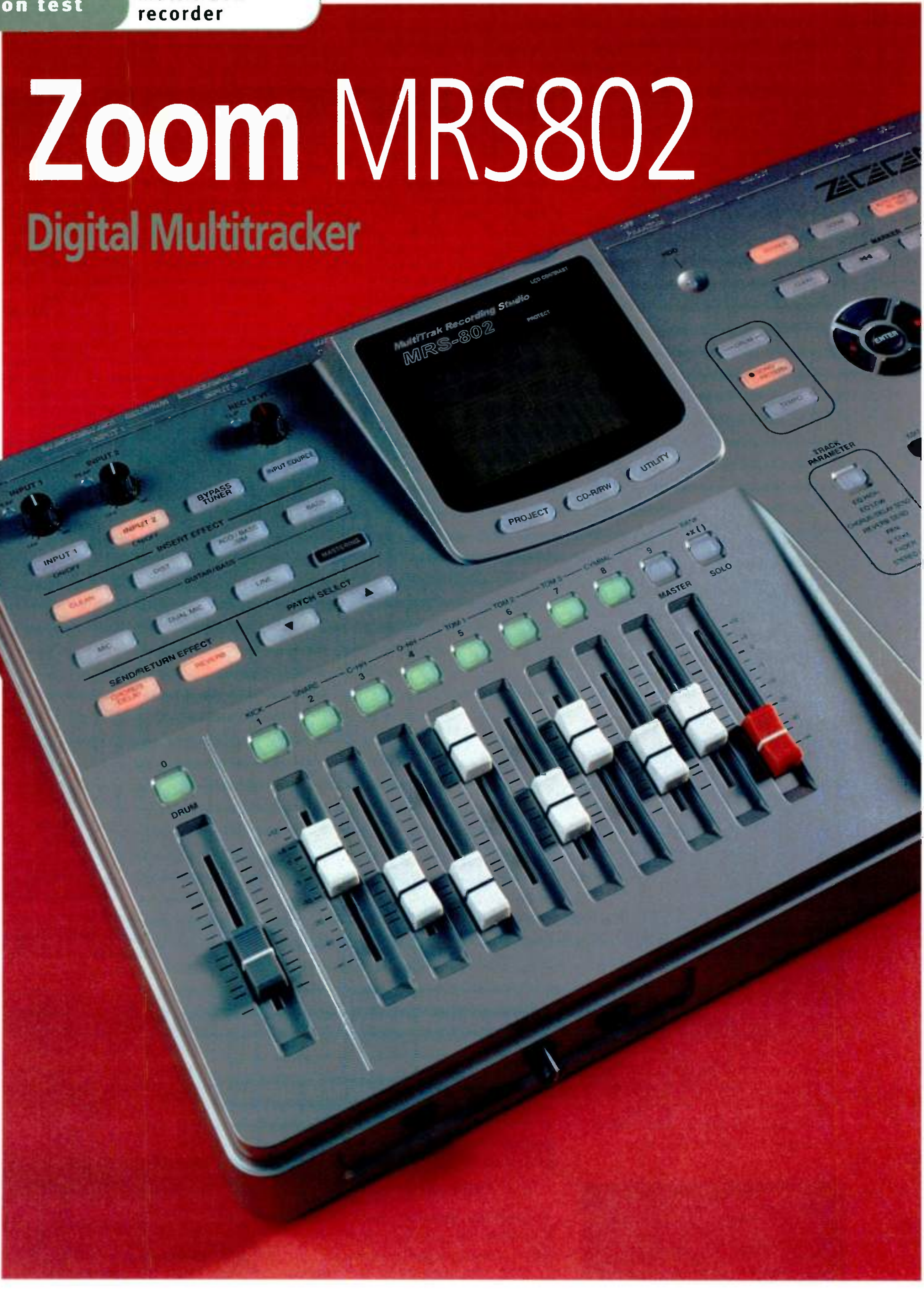


A complete studio in a box, for around the cost of a decent software sequencing package.

Paul White

Zoom started life as an effects company, but today they also specialise in entry-level hard disk multitrack audio workstations, the Zoom MRS802 being a new addition to their expanding range. This time Zoom have gone for a more conventional number of audio tracks, as users seem to have an aversion to anything that isn't in multiples of eight.

Features Overview

The Zoom MRS802 is an integrated eight-track recording system, which features an additional stereo master track for recording your mixes, as well as ten virtual tracks for each track (including the stereo master track) for storing alternate takes. Recording is to a 20GB drive, and internal track bouncing is facilitated for where more than eight parts are required. Bouncing can be to virtual tracks, so you don't need to leave two tracks free for the purpose.

The essential cut/copy/paste editing features expected of a digital multitrack recorder are included, while a phrase looping facility allows imported or recorded audio loops to be played back a specified number of times to create a continuous backing which is recorded as audio. As with previous Zoom multitrack recorders, there's a built-in pattern-based drum machine offering in excess of 400 Patterns that can be used to create either a simple guide track or a part of the final recording. The mixer incorporates snapshot-based automation, and the effects section includes both insert effects and two send/return effects for reverb and chorus/delay.

Despite being an all-digital product, the only way in and out of the Zoom MRS802 is via analogue I/O. A pair of mic/line inputs on combi jack/XLR connectors feeds a pair of 24-bit oversampling converters, while the output (normally used for monitoring) is unbalanced on two phonos, again at 24-bit resolution. You can probably deduce from this I/O arrangement that the Zoom MRS802 is designed to record only one or two tracks at a time, and that all mixing and processing must be accomplished internally.

An optional CD burner allows tracks to be burned as stereo masters. MIDI In and Out sockets are available on the rear panel, and there's switchable phantom power for the mic inputs. An optional USB card is available for connection to a Mac or PC for file management or backup (Mac OS 9 onwards, but not OS X), but this was not fitted to the review model.

Although the main body of the case is constructed from plastic, the unit has a rigid metal sub-chassis and feels very substantial. Power is, of course, from an external adaptor. A centrally mounted LCD window with three mode buttons below it handles metering and parameter display/editing, and all the sections of the machine are separated into distinct areas to make navigation easy. For example, the section to the left of the display deals entirely with the inputs and the effects section while the angled fader section accesses the mixer functions with the channel keys doubling as drum sound trigger keys. A separate mixer fader is included for the drum track, as this is electronically generated and need not be recorded as audio.

The Control section is centred around a data entry dial and four quadrant-shaped cursor buttons with an Enter button at their centre, above which are controls for handling Markers, bouncing, punching in and out, automation snapshot selection and repeat looping. There's also an Exit key. Three buttons access the rhythm section, while a single Track Parameter button enables you to step through the track parameters, using the cursor buttons to access the two-band EQ, send levels, pan, virtual track selection, level and stereo linking. A tape-style transport section falls comfortably beneath the fingers at the bottom right of the front panel, and the illuminated Play and Record keys are sensibly large, with a positive feel. The transport controls include 'go to start' and 'go to end' buttons. A headphone output with level control is located on the front edge of the case, and there's also a jack input for use with an optional footswitch or pedal, to allow hands-free punching in.

Recording With The MRS802

Unlike tape, where you just pick a place and start recording, songs saved to a hard drive don't generally overwrite other songs unless you deliberately erase them. In the MRS802, songs are saved as Projects, where a Project contains not only the audio data you record, but also the settings pertaining to the recorder and mixer sections, plus any rhythm and effects settings used in the Project. Other data is also saved, relating to ▶

SOUND ON SOUND

Zoom MRS802 **£549**

pros
- Inexpensive.
- User friendly.
- Generally good sound quality.
- Good range of onboard drum patterns and effects.
- Not excessively noisy.

cons
- No Undo facility.
- No way to mix using external effects.

summary
A practical all-in-one studio for the musician who works by overdubbing. The CD-R drive or the USB card are essential for backing up and useful for making playable audio CDs of your mixes.

automation snapshots, Markers and MIDI functions. Creating a new Project is a simple exercise, after which you can name it with up to eight characters. The Project also has a number, which is the lowest unused number available. Project data is automatically saved when the unit is powered down via its power switch, during which time the message 'Bye, See You!' is displayed.

The flow of work starts with opening a new Project (up to 1000 Projects can be stored, drive space permitting) and then choosing a guide rhythm Pattern and tempo where applicable. You can then go on to record one or two tracks, with or without insert effects. Inputs to the MRS802 are automatically routed to whichever tracks are currently armed and, if a mono input is used, it routes to both tracks of the available pair so that you can record to either. The track select light serves a dual purpose, showing red for record ready, green for on, and dark for track muted. A single knob adjusts the record level for each input and, after a track has been recorded, you can select the next track you want to record onto and overdub straight away with no routing to worry about.

If you need a guide rhythm part, all you have to do is press the Drum button and then select from one of the available Patterns, after which you can set the tempo. When changing Patterns, you have to press Enter to load them in, which takes a second or so, but you can do this during playback, enabling you to audition the rhythms. Anything more complicated requires the Patterns to be chained as in a drum machine, which is done via Song/Pattern mode. MIDI Clock with Song Position Pointers can be sent by the MRS802, enabling a sequencer or drum machine to be sync'ed up. Start and Stop commands are also sent over MIDI, and it's also possible to use MIDI Controller data to control the drum levels. Another feature which may be useful both in live performance and in the studio is the ability to import Type 0 standard MIDI files, which may be used to play back the internal drum sounds or which can be output over MIDI to trigger an external module. This means it is possible to compose drum parts in a more friendly sequencer environment, then import them into the MRS802, where the data can be used to trigger the MRS802's own drum sounds. However, if you already have a sequencer, you may not feel you need an MRS802...

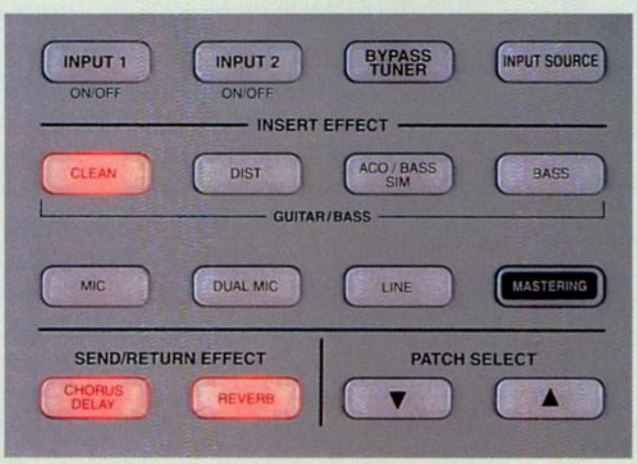
The drum sounds themselves are typical of what you'd expect from a mid-priced drum machine, and cover a variety of

An optional internal CD-RW drive allows Projects to be backed up, and also lets you record audio CDs in track-at-once or disc-at-once modes.

Built-in Effects Processing

The MRS802's insert effects may either be used when recording or may be inserted into any channel of the mixer (mono or stereo), including the drum machine channel. They may also be used in the mixer's stereo out insert points for mastering. There is, however, only one insert effect processor, so if you need to process different tracks differently, it will be necessary to record with the effect applied. There are two send/return effects, one a straight reverb and the other a chorus/delay. These are fed from send controls in the mixer as usual, and can be applied to all tracks simultaneously in differing degrees. Each of the effect types has its own key, so selecting effects is very easy.

The insert effects are actually multi-effect chains, the first of which is designed for DI'd guitar or bass and includes amp simulation as well as Zoom's own noise reduction, EQ, volume pedal (which can be plugged into the control jack), compression and modulated delays. Note that if you opt to use the volume pedal then you can't use a switch for punching in and out at the same time, as there's only one control socket. Mono and stereo mic effects are also on offer incorporating compression, mic preamp controls and vocal doubling. Finally, there's an effect for line inputs, with modulation effects plus a dedicated mastering chain starting with a three-band compressor and lo-fi option, normalisation and something called Dimension/Resonance. Hmmm, sounds like these could do a lot of damage in careless hands! All the chains include Zoom's own ZNR noise gating, EQ and volume pedal capability. Individual effects within an algorithm can be edited to about the same extent as you'd expect from a stomp box effect, but you can't edit the algorithms themselves. Some of the guitar amp emulations are very good, and seem to be based on existing Zoom guitar effect technology, so you can get a perfectly acceptable clean or overdriven rock guitar sound by DI'ing.



traditional drum and percussion sounds, with some electronic hits. Most of the factory Patterns are straight enough to be useful, and overall the kits sound solid.

Mixdown Facilities

Mixing involves balancing the levels of the recorded tracks, setting their pan and EQ values and applying send/return effects such as reverb. At this stage, an insert mastering effect may be applied to the whole mix, which is then recorded to the stereo master track within your Project. If you have the optional CD burner (which can be retrofitted if necessary), you can burn the mixed track directly to CD-R. The same CD-R recorder may also be used to back up Projects.

The mixer faders adjust the monitored level of recorded tracks (or tracks in the process of being recorded) and so are used both to set the monitor mix level and to adjust the final mix. When mixing, wanted tracks are switched on, unwanted tracks are switched off, and the inputs should be turned off if not required so as to eliminate noise coming in from that source. Arming the Master track and then going into record mixes everything to stereo, and a mastering effect may be used if you feel you need it. Personally I felt the mastering effects were most useful as special effects — there are none that I'd use for serious mastering as they all sound rather too processed to my ear.

Using the mixer is very straightforward, though the basic EQ is limited by being only two-band, albeit adjustable in frequency as well as cut/boost amount. Extra EQ is available within the insert effect chains, but you are limited by having only one instance of an insert effect available at a time. Each track has two sends for the reverb and chorus/delay effects, and odd/even channel pairs may be linked for stereo by selecting one of the channels and then following the Track Parameter menu until you find the Link page. Selected tracks may also be soloed using the Solo button.

Mixer snapshots can be saved as Scenes (up to 100 per Project) for partially automated mixing, and there's a dedicated Scene button for this purpose. After pressing Scene, you need to select the Scene number you wish to save into, then press Store to save the current mixer setup. Scenes may be named and though you could mix by recalling Scenes manually in order at the correct time, it's more precise to do this automatically using Markers to show where the Scenes should change. You can set any Scene number to appear at any Marker point, so Scenes don't need to be saved in order, but you do have to remember to save an opening Scene for the start of the Project corresponding to Marker number zero. More complex editing modes are available that allow you to disable only certain elements within a Scene so that these won't change as the Scene is recalled. For example, you can omit the Track Parameter section so that the EQ, sends and pan positions don't change with the Scene changes. Most Scene elements can only be enabled globally, in that they apply to all tracks, but the Track Parameters section can be included or omitted for an individual track or tracks, including the drum track.

Transport & Marker Options

When punching in, I usually opt for the manual (or footual?) option, and the footswitch jack is very handy for punching in and out while performing. However, you can set an auto-punch region by marking the points using the Punch In and Out buttons, and you can test your settings by rehearsing the edit during playback. When you're happy it's going to work, the machine can be set

Internal Pattern-based Drum Machine

The MRS802 includes 43 drum kits, where each kit comprises 24 drum or percussion sounds. Kits can be played via the internal Patterns or triggered from external MIDI data and the track selection keys in the mixer section may also be used to play sounds from within the current kit. The drum sounds feed through the same kind of channel architecture as the recorded audio tracks, so that balance, level and EQ can be adjusted and the send/return effects may also be added. There are 400 preset rhythm Patterns which may be used within a Song, but advanced users might be pleased to know that you can opt to use only selected portions of Patterns where this makes sense, and there are also empty Pattern locations where you can create and store new Patterns.

Patterns may be chained to form what Zoom call a rhythm Song and up to 10 Songs can be saved for each Project, though only one can be played or edited at a time. Each rhythm Song may comprise up to 999 steps and can be composed in step mode, in which case full editing is possible, or in Fast mode, where the whole Song is created in one go, but no editing is possible without going back several steps. Fast mode allows you to express the Song almost as an algebraic formula, where you specify the number of repeats of specific Patterns. Once this is complete, it is written to the rhythm part — if you have made a mistake, you have to go back and edit the formula, then write the rhythm part again.

running somewhere before the punch-in point by pressing Play and Record together, and it will enter record only between the set locations on the armed track or tracks. There's no Undo function, so the rehearse mode is very welcome, although if you're paranoid then you can use the MRS802's temporary track backup facilities, of which more in a moment.

Though fast wind buttons make navigation as easy as with tape, you can also drop Markers into a Project by pressing the Mark button either during playback or while the transport is stopped. A couple of Marker 'cursor' buttons may then be used to skip through the Markers, or you can use the dial to scroll to a specific Marker number. Unwanted Markers may be deleted once you are at their exact location using the Marker Clear key. There's also a Scrub mode that uses the familiar short repeating loop, and that can be used on one or two tracks at a time as selected in the display.

Editing is basic, but covers the necessary cut/copy/paste moves and is accessed via the Utility menu. The modes available are copy, move, paste and trim, though it's also possible to do fades (in and out), reverse sections of audio, and time-stretch/compress sections. As expected, time manipulation over more than a small percentage shows up some processing side-effects. The copy-related functions rely on marking the boundaries of the material that needs to be copied on the source track using Start and End values. An In point must then be defined on the target track. If data is moved to a different location on the same track, the original data will be overwritten. Trim mode allows unwanted noises or material at the start and end of a track to be removed, making it useful for mastering, where mixes can be cropped.

The concept of virtual tracks (V-Takes in Zoom parlance) is well known, as it provides a means to store alternative takes or multiple takes for later comping using copy/paste editing. However, the MRS802 also includes a 'track capture and swap' feature that allows any audio track to be temporarily saved on hard disk, from where it may be restored later. This may be useful when attempting tricky edits, as you can save the original state of the track just in case it all goes wrong, though you should be able to do much the same thing by copying the data to a V-Take track. The main difference is that the temporary 'capture and swap' files are deleted automatically when the Project is saved.

Test Spec

- Zoom MRS802 OS v1.00

When burning a CD, the MRS802 writes the contents of the stereo master track to disc as audio data, and either single tracks or whole albums can be burned. When writing single Projects, the CD-R can be left unfinalised, allowing further tracks to be added, though such discs are not suitable for making commercial CD masters, as these have to be burned in disc-at-once mode. When burning an album, a playlist is first created to tell the machine which mixes to burn, after which the disc is automatically finalised. CD-RWs may be used for jobs where the disc needs to be reused afterwards. The CD burner, where fitted, may also be used to play back audio discs and to import audio into the recorder. Also, very importantly, the drive may be used to back up Projects, across multiple discs as necessary.

In Use

Physical noise can be a problem with digital workstations, though the MRS802 is quieter than most and you could easily record vocals and instruments in the same room provided that you weren't too close to it. I found the recording quality to be clean and uncoloured and, though there is a little noise generated by the effects, especially when overdrive is used, the in-built noise gate is pretty effective at keeping tracks clean. I found the guitar effects to be typically Zoom, insomuch as they sound a little heavily processed, but they are very musical anyway, and several different amp emulations are available.

Every workstation has its own operating foibles, but considering I've not done much work with Zoom recorders before, I found the MRS802 very easy to get around, and most functions could be deduced without recourse to the manual. The most tedious part is creating Pattern chains, but it's no worse than using a stand-alone drum machine — I find copy/paste editing a real chore on any machine of this kind, now that I'm used to my software sequencer's editing. Again, though, it's no worse than any other hardware system in this respect other than the lack of an undo button.

The overall quality of the effects is as good as can be expected from a workstation in this UK price range and most are very usable, though I'd be inclined to give most of the mastering presets a miss. If you do need to use them, then I'd suggest editing them, as most of the presets are too severe, and in any event the MRS802 does such a good job of recording that you shouldn't really need to do much in the way of mastering, beyond some light EQ and compression.

The Final Word

The MRS802 is a good-sounding, easy-to-use eight-track workstation with decent in-built effects and drum patterns. The degree to which effects can be edited is a good compromise between depth and simplicity, and you also have the benefit of snapshot automation, which is useful when a chorus or solo needs slightly different settings to the verse. The single insert effect means you have to do some pre-planning, but on the whole the effects and drum sounds stand up pretty well.

Clearly there are compromises when a workstation is built to a price like this one is, the most notable being that all the mixer functions other than faders have to be accessed individually and then controlled using the data wheel, but, to be fair, the adopted system makes this pretty painless. There's no undo function, but then tape never had an undo either! The other main limitation is that there's no way to use external effects or processors other than to

Optional USB Connection

Where the USB option is fitted, an included CD-ROM provides all the data necessary to recover the factory default settings, including the two demo Projects. However, the most important aspect of having the USB option is that your MRS802's drive appears on the computer desktop just like any other hard drive, allowing files to be copied onto your computer. Audio tracks or mixes can then be edited on your computer as WAV files, though complete Projects may also be backed up. USB mode is accessed by powering up the machine while holding down the Clear key and then pressing the Autopunch In/Out key. However, as no USB card was provided, I was unable to test these functions.

An expansion slot at the rear of the unit accepts an optional USB interface board, allowing you to hook the MRS802 to your computer for backup and audio CD-writing purposes.

treat the main input or output, so you have to use the internal facilities for everything. I'd say the CD-R drive is almost an essential, rather than an option, as it allows you to back up Projects as well as to burn audio mixes, but if you opt for the USB port then you could instead back up your files to a computer and use mastering software to burn more professional audio CDs that way.

Provided that you only need to record one or two tracks at a time, the MRS802 provides a compact and cost-effective solution to recording good-quality demos or even commercial recordings, though you may benefit from a good external mic preamp and computer editing and mastering if you're really serious about making quality albums. On balance, though, I was pleasantly surprised by the sound quality and ease of use of the MRS802. Add a mic and headphones, and you really do have a complete studio at your fingertips. SOS

information

- **£** MRS802, £549; optional internal CD-RW drive, £229; optional USB expansion board, £99.95. Prices include VAT.
- **T** Exclusive Distribution +44 (0)1462 481148.
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TFPro P3 Recording Channel

Photos: Mike Cameron

The latest update of Ted Fletcher's ubiquitous half-rack recording channel.

Paul White

If, in all but colour, the TFPro P3 bears a striking resemblance to an older Joemeek product, it is because designer Ted Fletcher has parted company with the Joemeek name and is now marketing his revised designs under the TFPro name. The Joemeek name has been bought by the company that formerly distributed the Joemeek line in the United States, so some time soon we can expect to see green Joemeek products back on the market, though these will probably have undergone some radical design changes. Apparently Ted experimented with all kinds of colours, as he couldn't go on using green and didn't want to go for something too common like black, so he eventually settled on a rather nice muted shade of red.

Connections & Controls

The TFPro P3 is a mains adaptor-powered, 1U half rack-width voice channel which combines a mic preamp with an opto compressor and a three-band equaliser. It can accept mic-level signals via a rear-panel XLR or line-level signals via a balanced jack connector. There are two outputs on balanced jacks, a feature that comes in handy when you need to set up a zero-latency monitoring feed for a performer doing an overdub in a computer-based setup. If the spare line output is fed to the mixer in charge of the performer's monitor mix, it means the performer can monitor their overdub from its source rather than via software, though you must remember to mute the software-monitored signal to avoid a doubling or flanging effect in the headphone mix. A Mix In jack is also fitted that can be used to mix another signal with the existing mic or line input, plus there's a TRS insert jack that allows additional processing to be inserted between the preamp and the following stages.

The mic preamp is based on Ted Fletcher's current-sensing design and works well with pretty much any dynamic or capacitor microphone. Full 48V phantom power can be applied by means of a rear-panel button (with its associated red status LED also on the rear panel, oddly enough). The benefit of a current-sensing design is that it isn't unduly affected by microphone impedance, so, in theory, all low-impedance mics should 'match' it perfectly. A front-panel button selects mic or line mode and the only other control associated with the preamp stage is a gain control calibrated from 0-60dB. Separate buttons switch the compressor and EQ sections in and out of circuit.

No make-up gain control is required by the compressor, as the system used effectively changes the level of the input signal relative to a fixed threshold using the Compression knob. This isn't a unique approach, though it is less common than the variable-threshold system used in the majority of modern compressors. Two further knobs adjust the

The rear panel offers mic and line inputs and an insert point, alongside a phantom power switch and associated status LED. The Mix In socket allows an external signal to be sent to the outputs alongside the processed signal, and parallel outputs allow easy simultaneous recording and monitoring.

SOUND ON SOUND

TFPro P3 £159

pros

- Clean mic preamp.
- Vintage sound.
- Simple but nice-sounding equaliser.

cons

- Compressor could use a better gain-reduction metering system.

summary

The P3 is sensibly priced, easy to use and has a solid, musical sound with a character all its own.

attack and release time constants, though there's no programme-dependent release option. As with Ted's earlier compressor circuits used during his 'green' period, the P3 uses a lamp/photocell topography, which gives it a somewhat non-linear, soft-knee characteristic that produces a distinct flavour of compression. This works particularly well on vocals where you want an obviously compressed sound, and in most cases it sounds pretty musical and assertive, though there are inevitably occasions where it doesn't suit the source being treated. Setting up has to be done mainly by ear, as the only gain-reduction read-out is a single yellow LED — in most cases, this should flicker only on signal peaks.

The EQ section offers three fixed-frequency cut/boost bands at 80Hz, 1.5kHz and 8kHz, with 16dB of cut or boost in each band. A five-LED bar-graph level meter monitors the output signal level, plus there's an output gain knob that goes from off to +6dB. A red Peak LED in the EQ section comes on just before clipping and, as this comes post-EQ, it warns of excessive EQ boosting. The EQ bands are well chosen, given that they can't be varied in frequency, though I'd generally use the mid-band

Here you can see the optical gain control element at the centre of Ted Fletcher's compressor design: the light emitting diode (LED) on the right acts as a light source, to which the light dependent resistor (LDR) on the left responds. It is the non-linearities of this system which contribute to the characteristic sound of the TFPro P3's compression.

only for cutting, as boosting in the 1.5kHz area usually makes vocals sound harsh and nasal. Of course this harshness could be used as a deliberate effect — just don't expect me to listen to your demos!

Is Red The New Green?

The preamp section of the TFPro P3 is quiet and generally clean sounding with plenty of gain adjustment range, while the compressor has a distinct '60s 'opto' quality that can be a bit over-obvious if not adjusted carefully, but equally it can sound very good on pop and rock vocals as well as on bass guitar and other instruments. The EQ is very simple and quite sweet sounding and is fine for general tonal adjustments, but far less useful for 'surgical' tonal correction. In all, the P3 offers little that we haven't seen before in Ted's designs, but it delivers a quality sound with a distinct character in an affordable and easy-to-use format. SOS

information

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Speed Tests

IDE Drives For Music — RAID Arrays

Traditionally, SCSI drives give high sequencer track counts while IDE drives are more affordable but perform less well. Can you obtain better Mac sequencer performance and save money by using *multiple* IDE drives? We present some evidence...

Mike Watkinson

Owners of computer-based sequencers have known for a couple of years now that although SCSI drives give the best performance (and therefore better sequencer track counts), IDE drives do the job well too, if not quite as well as SCSI models — although at the upper end of the IDE specification there are drives which come close to competing with their SCSI cousins. However, in addition to coming installed as the standard system drives on most new PCs and Macs, IDE drives are much more affordable to buy separately. Because they are now used more widely in general computing applications worldwide, their price is kept low. So I began to wonder — is it possible to put together a high-performance Mac recording system based around affordable IDE drives and make up any performance shortfall compared to a SCSI-based system by using *multiple* IDE drives working together? With this premise in mind, I began some tests to find out.

Using Multiple Recording Drives

The latest Power Macs now come with three IDE drive busses as standard, one rated at ATA33, one at ATA66 and one at ATA100 (see the 'ATA & IDE — What Does It All Mean?' box on the next page for more on ATA). The ATA33 buss (which is an ATA buss running at 33MHz, as the name suggests) is set up for use with optical drive(s) and the layout of controller and power cables facilitates this. The ATA66 and ATA100 busses are intended for use with hard drives, and again the controller and power cables provided line up with the supplied drive mountings. In theory, you have all the connections you need to add extra drives to your system.

With two or more drives in a system serving up data (in this case audio), it is possible to create two types of 'RAID array'. RAID is simply an acronym standing for 'Redundant Array of Inexpensive Disks'. There are many types of array, all of which use a collection of disks to create a single storage volume. The disks can operate independently of each other and the volume can tolerate the failure of a drive without losing data. The most basic type of RAID is Level 1 (known as a 'mirrored' RAID array), which provides 'redundancy' by writing identical data to two (ideally identical) drives. In this scenario, if one drive fails, the other takes over and no data is lost. Hence it is used for data security and not to increase performance (although a slight increase is usually noted).

The second common RAID type, Level 0 (or a 'striped' array) shares data between two drives using mini-partitions (stripes) on each drive, therefore theoretically doubling the performance available from a single drive, since the load on each is halved. Technically, this is not a RAID array at all, because no redundancy occurs, and the failure of one drive results in complete data loss.

So, with the intention of creating high-performance volumes, I used tools available in Mac OS X (in *Disk Utility*) to set up

Drives on Test – Specification

Manufacturer	Seagate	IBM/Hitachi (see note)	Western Digital
Model name	ATA V Plus	180 GXP	WD1000JB
Capacity (GB)	120	180	100
Interface	ATA/ATAPI-6 supporting	ATA/ATAPI-6 supporting	ATA/ATAPI-6 supporting
Maximum quoted sustained data-transfer rate (MB/s)	44	56	—
Quickbench-measured sustained read rate (MB/s)	41	64	42
Average seek time (ms)	9.4	8.5	8.9
Latency (ms)	4.2	4.2	4.2
Spin speed (rpm)	7200	7200	7200
Noise (bels)	3.3 (high performance)	3.0 (idle)	3.5 (idle)
Buffer Size (MB)	8	8	8

Note: When the drives were supplied for review, the company was called IBM, but during the course of writing IBM and Hitachi's Storage Technology Business's have combined to form a new company, namely Hitachi Global Storage Technologies (www.hitachistorage.com). The drive formerly known as the IBM Deskstar is now the Hitachi Deskstar, but is exactly the same drive.

RAID Level 0 arrays on two internal drives in some of the various permutations offered by the multi-buss Power Mac.

Mac OS 9 requires a separate application to achieve the above and, unlike OS X, which requires no additional hardware to be present except the drives themselves, also requires the use of a PCI card that adds the equivalent of two IDE busses to the system. These are known as ATA RAID cards, and there are several on the market. Most are manufactured by Acard and rebadged, as was the one I assessed, the Alchemy ATA RAID 133 PCI card from Miglia. I also used various drive software, such as Lacie's *Silver Lining Pro* (which is generally bundled with Lacie's hard-drive products), Charismac's *RAID Toolkit*, and the imaginatively named *SoftRAID* (by SoftRAID!), which was originally written to format SCSI RAID systems, but is fooled by the ATA RAID card into thinking that it is dealing with a SCSI setup. OS X can of course also create RAID arrays using drives connected to RAID cards.

It's also possible to set up Firewire drives in RAID arrays, and this was another approach I tried. Miglia sent me an MTR Mediabank, an empty housing which contains two Firewire controllers and power supplies for two drives (although it can also be supplied with drives pre-installed). Although modern Power Macs have two Firewire sockets, these are not really separate ports, as they are controlled by the same buss. The exception to this is those Power Macs with the latest, faster Firewire 800 standard, where the 800 socket can be used with older-style 400 hardware using the appropriate adaptor, and is controlled by a separate buss to the 400 sockets. To create a second Firewire buss on the Dual DDR 1.25GHz Power Mac that I used for the tests, I installed a Keyspan Firewire card so that each Firewire controller in the MediaBank had its own buss.

The Tests

In each case, I ran several tests. I used *Quickbench* from Intech software (www.intechusa.com/QB.html), a well-respected benchmarking program, to obtain some comparative transfer-rate data, but, as has been pointed out many times in the past, software benchmarks are not necessarily a reliable indicator of real-world performance from a specific application. As a standard, I chose to use *Logic Audio* as my sequencer package for track-count testing throughout, more due to ▶

ATA and IDE — What Does It All Mean?

- ATA (Advanced Technology Attachment) — Industry-standard disk-drive interface specification.
- ATAPI (ATA Packet Interface) — this is the protocol that allows other devices like CD-ROMS to use the ATA interface.
- IDE (Integrated Drive Electronics) or (E)IDE (Enhanced IDE) — this is synonymous with ATA. The ATA standard was one of the first to design the control electronics for the disk drive on board. Hence ATA/ATAPI is also referred to as IDE.
- ATA/ATAPI — a parallel connection protocol that uses 40 signal lines and is currently at a level called ATA/ATAPI6 (abbreviated to ATA6 by most manufacturers).

There are three transfer protocols for parallel ATA. They are PIO (programmed input/output), DMA (Direct Memory Access) and UDMA (Ultra DMA). UDMA is the current standard and has six modes of operation (0-5) which support transfer rates from 16.6MB/s to 100MB/s. All the drives tested here have ATA/ATAPI6 interfaces and support UDMA mode 5, which means they have a theoretical data-transmission rate of 100MB/s when connected to a supporting motherboard controller. Of course, actual data-transfer rates are a result of the mechanics and electronics of the drive design itself.

Logic's Arrange Window set up to perform the 'Long files' test.

Logic's Arrange Window set up to perform the 'Short files' test.

personal familiarity than anything else.

My first test was the 'large files' test, as used in previous *SOS* test articles (see www.sound-on-sound.com/sos/Mar02/articles/Firewire1.asp for more details). In essence, this consists of running steadily increasing numbers of continuous tracks (like a conventional multitrack tape recorder) until *Logic* flags up its 'Hard Drive Too Slow' warning — at which point I noted the maximum possible track count. The files themselves are recorded at 24-bit 96kHz resoution, and played back at the 96kHz sample rate.

The folder which contained these large files was 1.09GB in size and, as a test of combined read and write capability, was copied from each drive back to the same drive.

The next test was the 'small files' test, utilising tracks made up of separate audio files a 16th note in length, with 16 to a bar. The absolute length of each file is determined by the fact that the song used for 'construction' ran at 120bpm, so each file is 0.125 seconds long, or 12,000 samples. More details of test song construction can be found at the above-mentioned March 2002 *SOS* article.

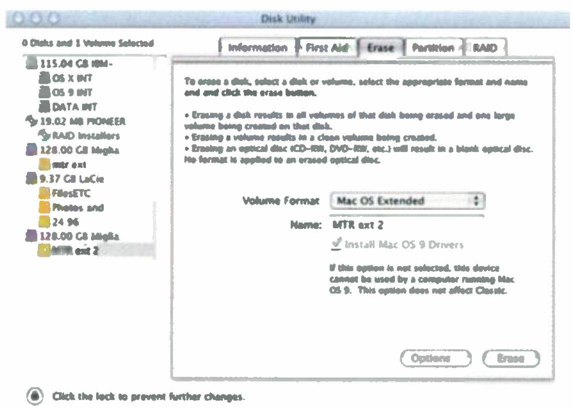
In my testing for previous *SOS* articles this song was played back at 120bpm (eight files per second per track) but due to enhancements made to *Logic* in the interim and the increased performance of the disk drives on test this month, this test no longer posed a severe enough load on the hardware. Therefore I increased the tempo to 480bpm (four times the original speed, or 32 files per second per track). Again, the files used were 24-bit, 96kHz recordings played back at 96kHz.

To ensure a level playing field, each drive was initialised before use by *Disk Utility* in OS X, with the 'Install OS 9 Drivers' option ticked (see screenshot below). The data was copied each time from a Lacie PocketDrive rather than constructing the song as required, partly to save time and partly to ensure that the bits were laid down on each disk in the same order each time.

Disk Utility had many uses during these tests. One which proved invaluable was its ability to mount and unmount drives and partitions. With several drives present, each with identical data, it was often difficult to predict which drive *Logic* was using (without looking in *Logic*'s Audio Files window and checking View/Get Info). Making unused drives invisible to *Logic* removed this potential point of confusion.

The first test was to ascertain the best configuration for a second internal drive. One of the Western Digital WD1000JB drives (known henceforth as WDC — see the side panel for the spec of all the drives on test) was used as the second drive in each case. In Table 1 (on the next page) the system drive is on the ATA100 buss and in Table 2 (below Table 1) it is on the ATA66 buss.

Although instinct would suggest that the ideal setup would have the rarely accessed system drive on the technically slower ATA66 buss and the heavily used audio drive on the superior ATA100 buss, the figures do not bear this out. It appears to make no real difference to *Logic* which way the drives are connected. The *Quickbench* figures for the WDC do show an improvement when it is connected to the faster buss, but strangely only for write speeds. As if to underline the flakiness of benchmarks, the *Quickbench* figures for the other drives in the same positions do not always follow this pattern, and refuse to show



Using *Disk Utility* under OS X to initialise drives.

4

Revere the past.

2

Embrace the future.

"I like the ISA 428 very much indeed. It looks good, sounds fantastic, is completely bombproof and is a joy to use.

In terms of technical performance, the card at least equals - and in most cases outperforms - other comparable converters.

As a stand alone mic pre amp, this unit is very capable indeed, but with the optional converter card installed it makes a very competent and cost effective eight-channel (four mic, four line) front end for a DAW or digital recorder."

Hugh Robjohns - Sound on Sound - May 2003

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any great advantage for either buss.

I was prepared for the fact that drives sharing the same buss should not perform quite as expected, since the buss can only deliver one read/write instruction at time. However, I was disappointed to note that performance did *not* increase when the drives were on separate busses. What this probably indicates is that the system drive is accessed very infrequently during audio playback, and makes few interrupts to the audio drive.

The data partition on the system drive outperforms every second drive in all the *Logic* tests, as in every hard drive test I have ever done. Indeed, with *Logic* the system drive always performs better than *any* other drive in the setup, for a reason I have not yet discovered. I would suppose that since the ATA specification finds it hard to allocate information simultaneously to two drives on the same controller, a single drive doing both jobs (audio files and OS/application) would be easier for the system as a whole to control, even if there are several ATA busses.

My next test was to find out which of my contenders was the best-performing drive. IBM (now Hitachi), Seagate and Western Digital supplied pairs of drives. IBM/Hitachi sent the 180GB version of their well respected 180GXP model. Seagate sent the Barracuda ATA7200.7 in 120GB form and Western Digital sent the WD1000JB in 100GB form. The 'JB' stands for 'Jumbo Buffer', but in fact all three drive types supplied had an 8MB buffer.

Although I had no conclusive evidence from the first test to support my hunch as to the best position for the second drive in a two-drive system, I attached the system disk to the ATA66 buss and one drive from each manufacturer in turn to the ATA66 and ATA100 busses in turn, as shown below.

Table 4 (above) shows that the Seagate has a slight edge over the WDC in the small files test, which supports their claim of a marginally faster seek time (as shown in the 'Drives On Test' box), but the sustained data-transfer rate (when reading) is identical, confirmed by *Quickbench*, which results in equal performance with large file track

	Small Files (Max Track Count)	Small Files — 4x speed (Max Track Count)	Large Files (Max Track Count)	1.09GB Folder Copy (Min/secs)
WDC on ATA66	40	16	40	1'41"
WDC on ATA100	—	16	39	1'40"
Data Partition on System Disk	37	22	66	1'39"

Table 1: System disk on ATA100 buss.

	Small Files — 4x speed (Max Track Count)	Large Files (Max Track Count)	1.09GB Folder Copy (Min/secs)
WDC on ATA66	16	38	1'38"
WDC on ATA100	16	39	1'39"
Data Partition on System Disk	22	66	1'39"

Table 2: System disk on ATA66 buss.

	Sustained Read (MB/s)	Sustained Write (MB/s)	Random Read (MB/s)	Random Write (MB/s)
WDC on ATA66	41	57	34	57
WDC on ATA100	42	81	30	82
Data Partition on System Disk	43	37	38	25

Table 3: *Quickbench* figures with the system disk on the ATA66 buss.

	Small Files — 4x speed (Max Track Count)	Large Files (Max Track Count)	1.09GB Folder Copy (Min/secs)
WDC on ATA66	16	38	1'38"
WDC on ATA100	16	39	1'39"
Deskstar on ATA66	28	60	1'25"
Deskstar on ATA100	27	60	1'47"
Seagate on ATA66	21	40	2'01"
Seagate on ATA100	21	38	2'01"
Data Partition on System Disk	22	66	1'39"

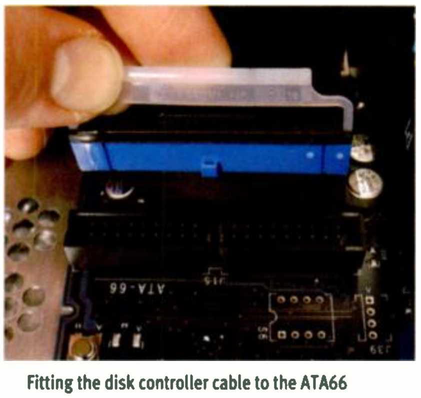
Table 4: *Logic* results for all three drives when fitted as 'second' drives, with system disk on ATA66 buss.

	Sustained Read (MB/s)	Sustained Write (MB/s)	Random Read (MB/s)	Random Write (MB/s)
WDC on ATA66	41	57	34	57
WDC on ATA100	42	81	30	82
Deskstar on ATA66	58	58	40	58
Deskstar on ATA100	64	88	40	45
Seagate on ATA66	41	43	34	35
Seagate on ATA100	41	41	35	36
Data Partition on System Disk	43	37	38	25

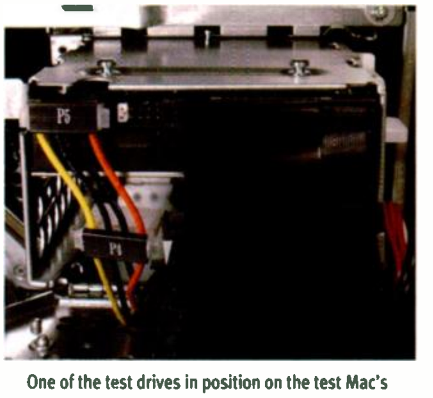
Table 5: *Quickbench* results for all three drives fitted as 'second' drives, with the system disk on ATA66 buss.

counts. Both are beaten by some margin in all *Logic* tests by the IBM/Hitachi Deskstar. Although seek time is fairly similar, the data-transfer rate (both quoted and as measured by *Quickbench*) are in the order of 25-33 percent better than the other two drives. There is an argument that since Power Macs have used IBM Deskstars as system disks for some time now, driver structure may be favourable as a result, but this is pure conjecture; in fact, the most recent range are supplied with the same model of Seagate drive on test here as their system disk!

My next challenge was to see whether two additional drives could be successfully configured as a RAID pair, and if so, which position of the several combinations possible gave the best performance. Theoretically, given previous experience with SCSI RAID, this should give extremely high performance at a fraction of the cost. But read on...



Fitting the disk controller cable to the ATA66 socket on the test Mac's motherboard.



One of the test drives in position on the test Mac's ATA66 buss.

Two Western Digital WD1000JB drives in position on the test Mac's ATA100 buss.

Continuing to act on my unproven hunch that the system drive should be controlled by the ATA66 buss, I tested a pair of WDCs (for no particular reason other than they arrived first for review) in RAID Level 0 array with both drives on the ATA100 buss, and then with one drive on each buss (see above).

In each case *Disk Utility* (OS X) was pressed into service to select the drives and configure the array (see screenshot, below right). Perhaps not surprisingly, with both drives on the same buss, the performance suffers, as you can see from Table 6 above. Now both drives are attempting to make multiple read/write requests simultaneously, and the buss is forcing them to take turns, as only one can be processed at a time. With a drive on each buss, performance increases dramatically, nearly matching that of the data partition on the system drive. For purposes of comparison, the Deskstar pair and the Seagate pair were also tested in these same positions (see Tables 7 and 8).

Unfortunately, the figures for the Seagates and Deskstars in RAID 'across the busses' do not back up the findings for the WDCs, and a distinct instability was noted during operation. With both drives on the same buss, both Seagates and Deskstars performed as poorly as the WDCs.

I cannot therefore recommend this

The System Disk On Test

The 123.5GB of the IBM (now Hitachi) Deskstar that is the original hard drive in the Power Mac was initialised and partitioned prior to testing. The first 10GB partition had OS X (10.2.3) and *Logic* (v5.5.0) installed. The second 10GB partition had OS 9.2.2 and Logic (5.5.0) installed. The remaining partition (103.5GB) was used as the data partition (ie. where the song/project folders were stored).

	Small Files — 4x speed (Max Track Count)	Large Files (Max Track Count)	1.09GB Folder Copy (Min/secs)
RAID across busses	18	63	0'59"
RAID both on ATA100 buss	17	27	1'08"
Single drive on ATA100	16	39	1'39"

Table 6: WDC's in RAID Level 0 array with no separate controller card, system disk on ATA66 buss.

	Small Files — 4x speed (Max Track Count)	Large Files (Max Track Count)	1.09GB Folder Copy (Min/secs)
WDCs RAID across busses	18	63	0'59"
Single WDC on ATA100	16	39	1'39"
Deskstars RAID across busses	22	27	0'57"
Single Deskstar on ATA100	27	60	1'47"
Seagates RAID across busses	16	29	1'08"
Single Seagate on ATA100	21	38	2"01"

Table 7: Comparison of pairs of disks, all three drive types in RAID Level 0 array, with no separate controller card and the system disk on ATA66 buss.

	Small Files — 4x speed (Max Track Count)	Large Files (Max Track Count)	1.09GB Folder Copy (Min/secs)
Deskstars RAID across busses System on ATA33 buss	10	37	0'35"
Deskstars RAID across busses System on ATA66 buss	22	27	0'57"

Table 7a: Comparison of Deskstars in RAID Level 0 array (one drive on ATA66 buss, one on ATA100 buss), with system drive on ATA33 and ATA66 buss.

configuration for creating RAID arrays, where any device has to share a buss with any other device, as a stable solution for high performance. I alluded near the start of this article to the third buss on a Power Mac, the ATA33 buss. For the purposes of testing I disconnected and removed the Superdrive from the TA33 controller and power cables, and substituted the system drive, also adding a Deskstar on each of the other two busses. If this were to be chosen as a working system then the System disk would need to be mounted below the Superdrive in the spare optical drive bay using appropriate brackets.

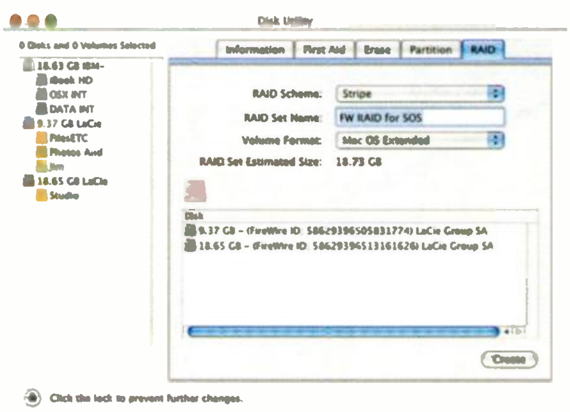
The *Quickbench* results are near enough the same, but the practical tests are revealing. Large file track count goes up by 10 when the system disk is not sharing IDE bandwidth with one of the data drives, but small file count decreases dramatically. Also, folder copying is much faster. These figures go some way to showing improved data-transfer rate at the cost of seek time. Although stability has improved, 'internal' RAID is still not a viable option compared with the use of a simple second drive, even with three busses in operation.

Another possible approach is to install a dedicated RAID controller card, to which end my next test was directed.

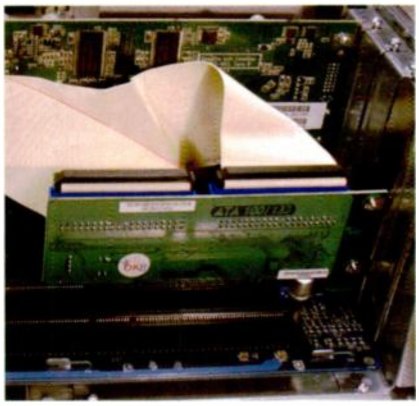
When using two IDE busses in a Power Mac, a third is necessary in order that neither drive in the pair is sharing a buss with the system drive. The Miglia Alchemy ATA133 card shown on the next page provides two further busses (hence its RAID nomenclature) each theoretically running at up to 133MHz, depending on the specification of the attached drives.

Installation is a cinch in a Power Mac, just as with any other PCI card, but finding a tidy path for the controller cables was tricky with the graphics and other cards in the way; I had to make sure that no cables snagged or got crimped when the G4's door was closed.

Tables 9 and 10 (on the next page) compare the performance of a single Deskstar drive connected to one of the Alchemy's



Disk Utility being used under OS X to configure one of the Firewire RAID arrays used in the final tests.



The Miglia ATA133 dual-IDE buss RAID controller card installed in the test Mac, and the necessary connection cables.

	Sustained Read (MB/s)	Sustained Write (MB/s)	Random Read (MB/s)	Random Write (MB/s)
WDCs RAID across busses	79	112	42	106
Single WDC on ATA100	42	81	30	82
Deskstars RAID across busses	113	113	76	111
Single Deskstar on ATA100	64	88	40	45
Seagates RAID across busses	79	81	59	76
Single Seagate on ATA100	41	41	35	36

Table 8: *Quickbench* results for systems shown in Table 7.

	Sustained Read (MB/s)	Sustained Write (MB/s)	Random Read (MB/s)	Random Write (MB/s)
Deskstars RAID across busses System on ATA33 buss	113	109	66	108
Deskstars RAID across busses System on ATA66 buss	113	113	76	111

Table 8a: *Quickbench* figures for systems shown in Table 7a.

	Small Files — 4x speed (Max Track Count)	Large Files (Max Track Count)	Folder Copy (Min/secs)
Single Deskstar/ATA100 buss/OS X	27	60	1'47"
Single Deskstar/ATA133 buss/OS X	29	61	1'20"
Single Deskstar/ATA133 buss/OS 9	23	47	1'25"
Data Partition/ATA100 buss/OS X	22	66	1'39"
Data Partition/ATA100 buss/OS 9	18	48	1'40"

Table 9: Single Deskstar as a second drive, comparing ATA100 and ATA133 busses.

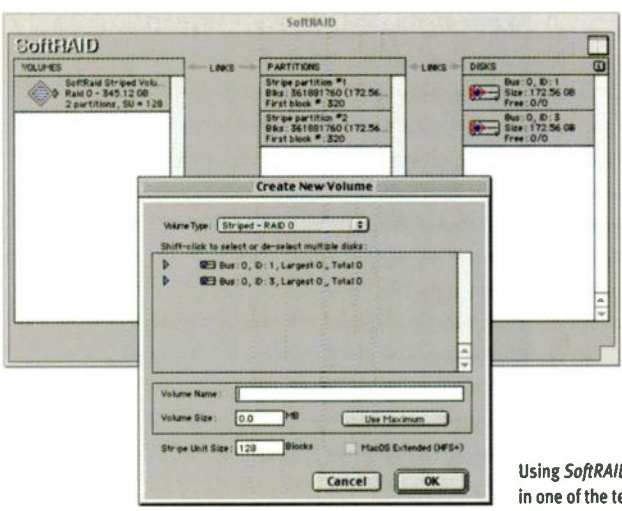
	Sustained Read (MB/s)	Sustained Write (MB/s)	Random Read (MB/s)	Random Write (MB/s)
Single Deskstar/ATA100 buss/OS X	64	88	40	45
Single Deskstar/ATA133 buss/OS X	66	80	43	51
Single Deskstar/ATA133 buss/OS 9	64	77	44	48

Table 10: *Quickbench* figures for systems shown in Table 9.

ATA133 controllers with the same drive connected to the internal ATA100 buss, and with the data partition on the system disk controlled by the Mac's ATA100 buss. Where the second drive is on the ATA100 buss, the system disk was on the ATA66 buss, but in all other cases it was on the ATA100 buss (and of course this has already been shown to make precious little difference!) Tests with this setup were repeated under OS X and OS 9. For the record, the performance of the data partition of the system drive under OS 9 was also noted.

Two clear points can be made on viewing these figures. Firstly, no advantage is gained by attaching a drive with an ATA100 interface to an ATA133 controller. Secondly, performance increase under OS X is marked, especially for continuous files.

In order to assess the RAID capabilities of this card, I tried two different setups. Under OS X I used *Disk Utility* to create a striped RAID Level 0 array between the two Deskstar drives, one connected to each controller on the Alchemy card. With the same connections made I also used Lacie's *Silver Lining Pro* and *SoftRaid* to set up RAID Level 0 arrays in OS 9 (see below).



Using *SoftRAID* to create a new volume in one of the test RAID arrays.

Table 11 on the next page shows that, despite the promise of separate controllers, something is holding this setup back in terms of performance using audio applications. Presumably the fact that the card is PCI and therefore still subject to the restrictions of that single buss would be the root cause of this (each drive is, in fact, still taking turns to respond to read/write instructions).

Table 12 (also on the next page), which shows *Quickbench* results for the same setup, contains some impressive figures across the board (though strangely not as high as for RAID arrays without a controller card — see Table 8). Again the difference between benchmarks (theoretical performance) and applications (actual performance) should be noted. Note that *Silver Lining Pro* allows the RAID array to be set for these two situations. In this case, it makes no notable difference to performance.

The Mediabank MTR

The final connection configuration solution I tested was of great interest after my experiences testing Firewire drives in *SOS* August 2002 (www.sound-on-sound.com/sos/Aug02/articles/firewire.asp). Power Macs dating from before January 2003, as noted previously, do have two Firewire sockets, but these are fed by a single buss, so in order to create a second buss I installed a Keyspan FPCI3 Firewire PCI card.

The Mediabank is essentially a hard drive housing with space for two IDE drives, containing two Firewire (Oxford 911) controllers and two power supply connections, along with a small fan to dissipate the heat created by confining two high-performance drives in a small space. You can purchase a Mediabank with drives already fitted, but I chose an empty housing for review, since I was simultaneously testing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pairs of suitable drives. Squeezing the two Deskstars, chosen as the highest-performing drives in other tests, was no easy task (and the other drives would have been the same since they were of equal external dimensions, although at this point the exposed circuitry of the Deskstars was a worry — the other two brands were completely enclosed), and Miglia would do well to include some pictorial documentation as to how best to arrange the cables, since everything becomes an almost impossibly snug fit once the drive pair is positioned (see picture, right).

The drives are 'piggy-backed' using two supplied mounting plates, whose fixing screws are left slightly loose until the drive assembly is fitted in the casing. Once I had found the appropriate arrangement for each cable I tightened the screws and slid the case cover back on. The MediaBank has a very smart and hi-tech appearance, and with the Deskstars on board is, aside from its performance potential, a compact way of achieving 360GB of desktop storage (though not perhaps as tidy as putting them inside the Power Mac!). Just for the record, Tables 13 and 14 (below) compare the performance of each Firewire buss under OS X and OS 9 using a single Deskstar drive.

Again, *Logic* performs better under OS X than under OS 9, and the Keyspan Firewire buss gives better performance than the internal buss (although the file copy and *Quickbench* figures gives a less distinct picture).

Tables 15 and 16 (bottom) show the performance of RAID arrays set up under OS X using *Disk Utility*, and under OS 9 using Charismac *RAID Toolkit*, as supplied with the Mediabank.

Unfortunately, despite my enthusiasm for the concept and this product, both sets of figures are disappointing, especially those under OS X, and although the *Quickbench* figures are better than for single drives, the *Logic* track counts are worse. Miglia's technical support noted that they have had conflict problems with certain video-capture cards (video being their main market) but have no experience as yet with audio devices. The fact I was using a MOTU 896 Firewire audio interface for these tests may indeed have some bearing on RAID performance, although I had previously noted (in *SOS* August 2002) that the Firewire interface caused no degradation to track count.

	Small Files — 4x speed (Max Track Count)	Large Files (Max Track Count)	Folder Copy (Min/secs)
OS X RAID (*Disk Utility*)	21	28	0'58"
OS 9 RAID (*Silver Lining Pro* — Large Files)	19	28	0'59"
OS 9 RAID (*Silver Lining Pro* — Small Files)	19	25	1'00"
OS 9 RAID (*SoftRAID*)	21	24	1'00"

Table 11: Various RAID Level 0 arrays using two Deskstar drives on the Miglia Alchemy ATA133 RAID card.

	Sustained Read (MB/s)	Sustained Write (MB/s)	Random Read (MB/s)	Random Write (MB/s)
OS X RAID (*Disk Utility*)	88	83	47	82
OS 9 RAID (*Silver Lining Pro* — Large Files)	85	85	48	84
OS 9 RAID (*Silver Lining Pro* — Small Files)	71	83	44	83
OS 9 RAID (*SoftRAID*)	73	85	37	85

Table 12: *Quickbench* figures for systems shown in Table 11.

	Small Files — 4x speed (Max Track Count)	Large Files (Max Track Count)	1.09GB Folder Copy (Min/secs)
Keyspan Firewire buss OS X	21	48	2'23"
Internal Firewire buss OS X	19	35	1'52"
Keyspan Firewire buss OS 9	15	38	2'33"
Internal Firewire buss OS 9	12	34	2'14"

Table 13: Comparing the performance of the two Firewire busses using a single Deskstar drive under OS X and OS 9.

	Sustained Read (MB/s)	Sustained Write (MB/s)	Random Read (MB/s)	Random Write (MB/s)
Keyspan Firewire buss OS X	36	20	31	20
Internal Firewire buss OS X	35	31	29	31
Keyspan Firewire buss OS 9	34	24	30	24
Internal Firewire buss OS 9	34	30	28	30

Table 14: Quickbench figures for systems shown in Table 13.

	Small Files — 4x speed (Max Track Count)	Large Files (Max Track Count)	Folder Copy (Min/secs)
Miglia MediaBank in RAID using OS X (*Disk Utility*)	17	28	1'18"
Miglia MediaBank in RAID using OS 9 (Charismac *RAID Toolkit*)	14	37	1'20"

Table 15: RAID Level 0 arrays created with two Deskstars mounted in the Miglia Mediabank.

	Sustained Read (MB/s)	Sustained Write (MB/s)	Random Read (MB/s)	Random Write (MB/s)
Miglia MediaBank in RAID using OS X (*Disk Utility*)	68	38	50	38
Miglia MediaBank in RAID using OS 9 (Charismac *RAID Toolkit*)	65	46	50	46

Table 16: *Quickbench* figures for systems shown in Table 15.

Conclusions

ATA/IDE is cheap partly because it is the pre-eminent way to connect drives to computers, and partly because of its design. SCSI drives are of more complex design, a higher-reliability specification and offer the benefits of the SCSI interface, that is possible longer cable lengths, faster interface speeds up to 320MB/s, multiple command-queueing for more use of buss bandwidth and multiple drives on a single buss. Of course, all of this comes at a price premium. Some ATA drives, such as the Hitachi, also offer ATA command-queuing capability, but currently this is not supported widely in the industry by the controller and chipset manufacturers, although there are development projects underway.

So, surprisingly, despite the promise of the various setups on offer, the best combination of performance with *Logic Audio* is achieved when using a single drive in addition to the system disk, and where each drive has a unique controller. No RAID solution tested here offered me any advantage in terms of audio performance — in fact in several cases the performance is worse (despite benchmark figures that predict the opposite). If one were to specify the 120GB version of the Hitachi 180GXP (as opposed to the 180GB version on test), then

The Mediabank MTR open to the world with test drives newly installed. It's a very snug fit!

the best performance solution is also the cheapest — now, how often does that happen?

Suppositions that might be teased from this evidence could be as follows. Firstly, the ATA interface design is not the limiting factor on drive performance when running audio applications, or indeed any application. The mechanical and electronic design of the drive (that is, the real media transfer rate and how the drive prioritises tasks) combined with how they interact with other factors in the system is much more likely to affect 'headroom', since multitrack audio data very quickly saturates any buffers/caches available. Secondly, *Logic Audio* requests data in such a way that appears to work well with single IDE drives, but not with multiple drives in RAID array. RAID striping only works well when the total system architecture from drives to memory is not bottlenecked at any point. For example, even if you have a 100MB/s drive on a 50MB/s Firewire interface, the drive will never be able to deliver more than 50MB/s. Also, a poorly implemented RAID (using software or hardware) can create data-processing overhead which negates any theoretical gains, and again this is dependant on the way the application accesses data.

At the end of the day, while the IDE RAID concept may not have worked out, it's nice to know that you can obtain what amounts to 60 tracks of 24-bit, 96kHz audio-recording capability for just under 100 quid, with little point in spending further on multiple drives. That's quite a bargain! SOS

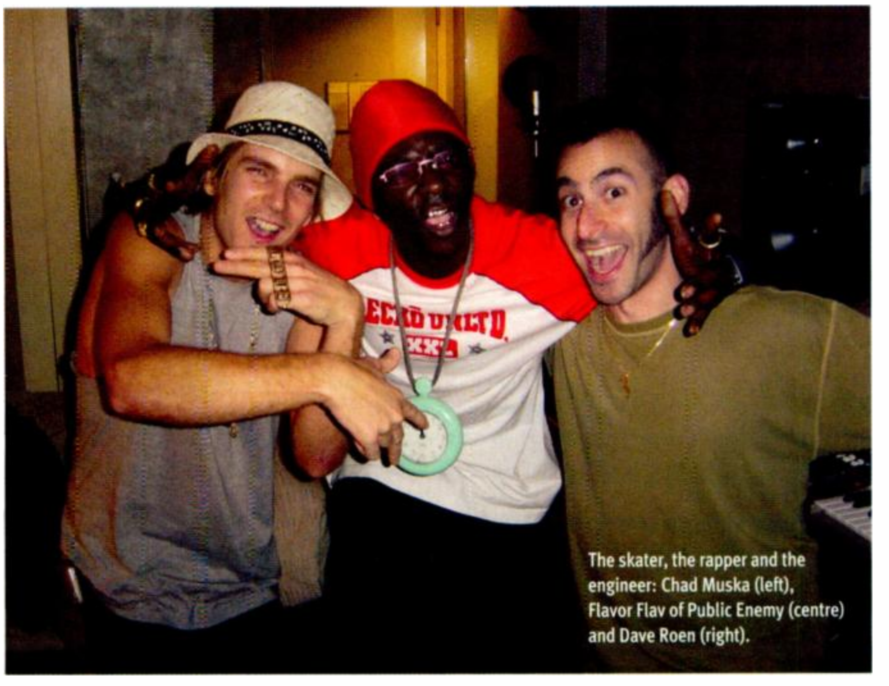
Drive Pricing

Western Digital	**WD1200JB, 120GB, with 8MB buffer.**
Model number	**WD1200JB.**
Price	**£104.50.**
Hitachi Deskstar	**180GXP, 180GB, with 8MB buffer.**
Model Number	**IC35L180AVV2071.**
Price	**£142.01.**
Hitachi Deskstar	**180GXP, 120GB, with 8MB buffer.**
Model Number	**IC35L120AVV2071.**
Price	**£91.00.**
Seagate	**Barracuda ATA 7200.7 120GB, with 8MB buffer.**
Model number	**ST3120026A.**
Price	**£107.00.**

In the absence of standardised pricing from manufacturers, all the drive prices quoted here are from www.dabs.com. They're subject to almost daily change, but this is how they were at the end of May 2003. Prices are in UK pounds and include VAT.

Room Service

Dan Daley



The skater, the rapper and the engineer: Chad Muska (left), Flavor Flav of Public Enemy (centre) and Dave Roen (right).

Chad Muska & Dave Roen: Recording Rappers In The SoHo Grand Hotel

A gaggle of famous MCs, a star of the extreme sports world, and a laptop recording setup in a hotel bedroom: could the *MuskaBeatz* project show the way ahead for rap music?

In the 1920s and '30s, Art Satherly, an Englishman and record label executive working from New York City, undertook regular peregrinations throughout the South-east United States, equipped with primitive recording equipment and a thirst to capture music in its element. It was a sound safari of sorts. Satherly's sojourns left the music world with some remarkable recordings, the best-remembered of which are the so-called Bristol sessions, recorded in a hotel room in the small town of Bristol, in eastern Tennessee, where he recorded the legendary Carter Family and which produced a lasting classic of the era, 'Wildflowers'. Satherly's work inspired countless other field recordists, including Alan Lomax, the BBC field recordist who recorded for posterity (and us) classic southern blues artists and folk performers including Woody Guthrie, Leadbelly, Burl Ives, and Pete Seeger.

Fast forward to the next century. Ensconced at the SoHo Grand, one of the generation of hip hostelries that now dot urban downtowns, a new conglomeration of cultural mavens accomplished much the same thing, without ever having heard of Satherly or Lomax. That's all right, because neither of them would have quite understood why someone needs a posse and a skateboard to make a record.

Both of those were in abundance in May 2002, when Chad Muska, world-champion skateboarder and Gen-X entrepreneur, and Dave Roen, a Los Angeles audio engineer and Muska's partner in 1212 Records, gathered an all-star crew of rappers to jam over Muska's beats, including Ice-T, Public Enemy's Flavor Flav, Prodigy, KRS-One, Biz Markie, Wu-Tang Clan's Raekwon and U-God, Grandmaster Melle Mel and MC Lyte, with the intention of creating an album which would be distributed through a global network of skateboarder and BMX action-hobby shops and chains. Since the record, dubbed *MuskaBeatz*, was released last February, it has sold several thousand units (it's hard to be precise without a SoundScan portal) without ever having been slimed by a major record label or sullied by radio payola. In fact, the entire project managed to stay clear of any conventional music industry or pro audio environment save for a brief stopover at Bernie Grundman Mastering in Los Angeles for a slight touch-up with compression and EQ. In the process, what 1212 Records' owners have done is outline a new paradigm for the music business and the music recording industry for the century to come: one of niche marketing, cross-cultural alliances and audio that is definitely not Abbey Road-approved. And they still managed to get room service.

What A Pair

Roen and Muska could not come from more disparate backgrounds as partners. Roen studied, then taught, music composition and theory after graduating with a masters degree in those disciplines from the University of California at Santa Cruz. The university's electronic music programme caught his ear and his fancy, and pulled him into the realm of audio engineering. He interned at Hans Zimmer's Media Ventures facility in Los Angeles before working as a salesperson at the Sam Ash music store chain in LA, where, in 1996, he met Muska.

By then, Muska was already a household name, assuming your household inventory included a skateboard. (A Google search on his name turns up over 10,500 hits, including an Anti-Muska site.) The Las Vegas native was an early star of what has become the multi-billion-dollar industry known as extreme sports, including skateboarding and BMX biking. Muska understood the power of branding and is a partner in some of the companies that manufacture the gear he uses, such as skateboard maker Shorty's and footwear company C1RCA. Muska was also developing his own instincts as a musician and producer. He had started a home studio based around an Akai MPC2000 hardware sequencer. With Roen's guidance, Muska expanded to a 500MHz Mac G4 running *Cubase* recording software and

Propellerhead's *Reason* software studio.

"We would hook up once a month or so at his house and I'd show him a few things about using the equipment," Roen recalls. "He wasn't very technical but he had a good understanding of music, and he would take things I showed him and then take those concepts way beyond where I might have. He wasn't worried about the principles of professional audio. He was just looking to make cool sounds and beats."

In 1999, the pair started talking about starting a record label, one predicated on the fact that Muska's beats, which he created in his home studio for use during his skateboarding performances, were starting to generate interest among his fans. "The idea was to do an album of Muska's beats — *MuskaBeatz*," says Roen. "But we realised that it had to be an album that had the same basic recording approach as the beats he used in his skateboarding — really, really basic stuff."

So basic was the approach that it could be put into practice on the portable recording setup that Muska and Roen had assembled for the skateboarder's use on the road during extreme sports tours: a Mac G4 laptop using the same software he had at home, a MOTU audio interface, a USB-compatible Roland PC300 keyboard, a Sound Devices USB mic preamp, and an Audio-Technica AT4050 microphone. Roen augmented it with a Mackie 1402 mixer and some self-powered Yamaha monitors.

Thus, 1212 Records was born, complete with its own recording facility. (The label name is pronounced 'one-two, one-two', as Roen explains: "It's what you say when you're testing to see if a microphone is on, and what rap record doesn't have one track at least with that on it?") In discussing what to record, Muska and Roen started dreaming up a wish list of guest MCs, never guessing that virtually all of them would end up on the record. First up was Guru, from Gangstar, who lives in Los Angeles and was reached through mutual friends. Guru's enthusiastic embrace of the primitive coolness of the way the record was being recorded, and the random way that Muska presented beats for rappers to pick from and freelance on, made them realise that the classic rappers of hip-hop's golden era would be equally receptive. "We knew we had to go to New York, which is where rap all started," says Roen.

Studio Time

At first, Muska and Roen considered using a conventional studio in New York. Time had been pencilled in at The Game, a rather hardcore rap studio in downtown Manhattan, perhaps best known for its work on the *Hip Hop Honeys* DVD series, a kind of rapper *Girls Gone Wild*. Chung King, one of the larger Manhattan studios and one which had built itself on rap work in the 1980s, got wind of the project and offered time at a steep discount. *MuskaBeatz* was building momentum before Roen and Muska had the key to a room.

While still considering whether to use a conventional studio for any of the project, the pair set up the portable gear in a room at the SoHo Grand to experiment with beats. When Biz Markie stopped in to meet them, he was immediately enamoured of the idea of doing a rap right then and there. That locked in the idea of a hotel recording marathon, one which would last through early July.

The hotel was chosen by many of the

A New Business Model

The business and legal side of the *MuskaBeatz* project was as unconventional and simple as the technical side. In virtually every instance, each guest artist agreed to a buy-out of the track and the publishing, transferring rights to the song and sound recording copyrights for a one-time fee. The amounts were not disclosed, but this arrangement will probably become far more commonplace as the music industry continues to evolve in the digital era. Attempting to document sales and track royalty payments to nine or 10 artists on a compliation record on a semi-annual basis is task enough for Sony Records, let alone a two-man startup like 1212 Records. But more to the point, it reflects the emerging Zeitgeist, one in which the music has become as commoditised as the disc medium itself has. With piracy and downloading increasingly eating away at the traditional back-end royalty returns, it makes more sense for artists to take a guarantee up front and leave the back end for the record label. It may rankle to some, but it seems to be one of the business models of the future of a very changed record business. But Roen also points out another benefit to having been on the record for the more vintage rappers, whose initial sales heights may be behind them: "This introduces them to a whole new market of kids who like hip-hop," he says. "It's big in the skater market."

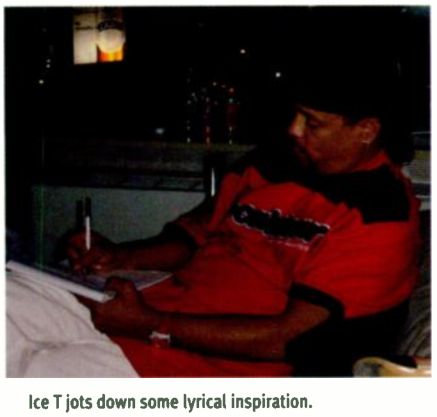
The portable setup used for recording in the SoHo Grand Hotel, with Mackie mixer (left), Apple G4 laptop running *Cubase*, and Yamaha monitors.

Chad Muska with rap legend Afrika Bambaataa.

same criteria one might choose a recording studio: it's centrally located in a hip downtown area, and Muska's girlfriend, a model, was able to secure a reduced rate through a deal with her agency. So in a standard double room at the SoHo Grand, Roen set up the G4-based recording system augmented by a Rolls four-output distribution amplifier for headphones. The stereo output of *Cubase* was fed into the Mackie 1402, and the Rode NT2 microphone Roen chose for the project was lined through a Mindprint mic pre with a touch of compression dialled in, then into the Mackie mixer. That, simply put, was total signal path for the project. Everything else was pure vibe, and that seemed to be equally simple. "We pushed the two beds up against the walls and made couches out of them," Roen says. "We made sure there were plenty of snacks and drinks, and after that it was just one artist after another coming through the room."

There was no need for any acoustical treatment in the room, or to create any isolated spaces. As Roen puts it, the entire room was the vocal booth. Just as well, because the posse effect was often present. Gangstar's Guru brought a crew with him that nearly filled the chamber. "Everyone was drinking and smoking and talking, and every time I wanted to do a take I had to yell, 'Shut up!' to get everyone to quiet down," Roen recalls, laughing at the thought of it now.

Biz Markie adds a keyboard part to his contribution.



Ice T jots down some lyrical inspiration.

KRS-One with Chad, Dave, and Roland drum machine.

Some of the sessions were planned. Biz Markie, the first of the New York-based artists to record, did his tracks in a few hours, choosing from an array of beats Muska had looped and then launching into a rap. Other sessions were decidedly *ad hoc* and of the moment. "We weren't sure if Ice-T was going to cruise by or not," Roen says. "There was a video crew also staying at the hotel and they were friends of ours and were working with Ice-T. [*T is now a regular on the hit prime-time television detective series* Law & Order, *which shoots in New York*.] We were just finishing our tracks with U-God when Ice-T knocked on the door. He was with his girlfriend. We played him a couple of tracks and he was psyched immediately. This was around 10 o'clock at night. Then the guys from the video shoot came in and it was mayhem for a while. We had to eject as many people as we could. It was just me, Chad, Ice and his girlfriend Coco. Ice picked a couple of beats out and just started writing. It went down quickly."

Instant Takes

Roen used a couple of techniques that are peculiar to rap sessions. The nature of the music is often totally spontaneous, with an inspired rapper ready to grab the microphone without warning to break in on a track. Roen, who had worked with rap artists as an engineer in LA, knew beforehand to always keep a certain amount of compression and limiting on the microphone. "You don't get a chance to do a level check every time," he cautions. "You don't know what's coming at you, so you want the compressor to grab it to tame it a bit."

Secondly, Roen ran a Tascam DAP1 DAT deck continuously during the sessions, with one channel being fed from the computer and one from the microphone through the mixer, not even turning it off between takes. "I lost some freestyle raps that Raekwon had done for the record at Chad's place in LA when the computer crashed," he explains. "It was really good stuff, too, and I was pissed off. So from then on, I always run a DAT deck during sessions constantly. On a rap session, you never know when the next take is coming. You always have to be ready. And I got everything, even the sound of people kicking back and hanging out, which is also cool stuff to put on a record like this. Some of that kind of stuff is on the Biz Markie track, just him and us talking."

There was no isolation or soundproofing in the hotel room, and the playbacks at four in the morning over the Yamaha monitors sometimes crept up in volume. However, over the course of a month's recording, there was not one noise complaint. "We got away with murder on that," laughs Roen. "What was even funnier was that each room has a crummy shelf-type stereo system in it, and when the guys in the video crew first checked in, they got complaints from the front desk to turn it down within a half an hour. Meanwhile, they never said a word to us."

The next step in the project was to bail on New York and head back to Los Angeles to 'mix' the record. "Really, there wasn't much to mix," says Roen, noting that the looped beats were on stereo tracks on the laptop and the vocals were mono, making for a grand total of three tracks. But there was editing at Muska's studio. Working on a Mac G4 with dual 800MHz processors and a TC Electronic Powercore DSP card as a signal processor, the two cut some new beats around the vocals, muting them here and there to create dynamics. "It wasn't so much mixing as it was arranging after the fact," says Roen. "More like rearranging."

When mastering legend Bernie Grundman got the record, Roen says he also appreciated its primitive rawness. "He couldn't believe it was recorded in a hotel room," he says. "It barely needed any EQ or compression."

Still Learning

'Barely' is a word that aptly describes the entire *MuskaBeatz* project. Yet it comprehensively encapsulates the paradigm shift the process and business of making music is undergoing. It probably seems natural to someone like Muska, who lives and breathes new models of business in a sporting sector that barely existed a decade ago. Roen, on the other hand, is traditionally trained and admits to being somewhat conventional in his thinking, and says he's still on a learning curve for the whole thing. But his sense of humour will certainly help him along.

"Everyone is used to working in a certain way," he says. "It's ironic: I have a master's degree in music and I've composed a symphony. Yet the one thing I'm most known for musically is that I wrote a couple of original tracks for the *Jackass* movie. I look at this whole project as though I was still in school. I'm still learning." SOS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Creamware Noah

Instrument-modelling Hardware Synth

Software instruments are flexible, upgradeable, and frequently offer better user interfaces, while hardware offers better tactile feedback and is easier to use live. But what if you could combine the plus points of both? Enter the Noah...

Mark Wherry

One of the oldest criticisms levelled at computer-based music systems concerns their use in live situations, where stability and reliability are of paramount importance — factors that some people have problems associating with computers. This is becoming less of an issue with modern, stable operating systems such as Windows XP and Mac OS X, as evidenced by the number of musicians now taking to the stage with laptops. However, there are still some things you just can't do with a laptop — the most obvious being that you can't use PCI cards. This means that if you want to use a DSP card in a live setup, such as Universal Audio's UAD1, TC's original Powercore or Creamware's SCOPE (not forgetting Digidesign's TDM-based Pro Tools systems) you really have to use a desktop computer, or attempt to tame an expansion chassis.

TC have now come up with a solution for this situation in the form of a Firewire-based Powercore system, which goes some way towards solving the problem of using computer-based DSP systems without having to bring your desktop computer along for the ride. However, you do still need a computer to drive the Firewire Powercore, of course, which doesn't do much for those people who'd rather not use a computer in the first place. And it's not just in live situations where computers have traditionally not been welcome — some people simply don't want to use a computer in the studio, or they don't want the computer to be at the centre of everything, running instruments, effects, mixers, and the program to generate the week's lottery numbers.

This situation leads to an interesting paradox: musicians want to be able to use the instruments and effects available for computer-based systems, with or without additional DSP power, but not every musician wants to use a computer in every situation. Two companies have announced products to cater for this need: Manifold Labs, whose Plugzilla is a stand-alone rack unit capable of running a selection of VST plug-in instruments and effects, and Creamware, whose new Noah is the focus of this review.

SOUND ON SOUND

Creamware Noah/Noah EX

£1599/£1999

pros

- Versatile system with excellent instruments and effects.
- Well-thought-out architecture makes using SCOPE Fusion Platform instruments easier than with a computer-based SFP system.
- Support for both audio and MIDI I/O via USB.

cons

- Expensive.
- Digital audio input only possible via USB.
- Noah doesn't automatically support other SFP instruments, and porting them will take time.

summary

Noah is a big step in bringing the concept of an open DSP platform from the desktop to the rack, and with its generally excellent initial complement of instruments, it should be very popular with musicians who want Creamware's high-quality synths without the complexity and nature of a computer-based setup — provided that they have the budget.

Genesis

Creamware's SCOPE Fusion Platform (SFP) was reviewed in last month's issue of *SOS*; and as regular readers will be aware, it's a platform that utilises SHARC DSP chips to run an environment supporting a range of virtual devices, including mixers, instruments and effects. Until now, the SFP has run on Creamware's family of PCI cards, using a host computer to provide a user interface. However, since the code for the SFP devices is written specifically for SHARC DSP chips, rather than for the host's Motorola or Intel processor, it would, in theory, be possible to build a stand-alone platform to host the SFP, doing away with the need for a computer — and this is exactly what Creamware have done to create Noah.

It's worth pointing out that although the Noah is an open platform, in that it can run different devices, conceptually it's not that different to any other DSP-based synth currently available, such as the Access Virus, Clavia Nord Lead, and so on. The big difference is that where these instruments are designed for a single purpose, Noah's open design gives it the potential to be any instrument Creamware can dream up, whether that's today or a year from now.

Noah's chameleon-like nature creates something of marketing problem for Creamware: how do you sell an instrument that has the potential to be *any* instrument? The answer lies in the first instruments

Creamware have chosen to ship with the Noah — since the company has a great deal of experience in modelling classic synths, the Noah concept is based around the idea of providing an ark for all your favourite synths. Ark, Noah; get it? Well, it's worth remembering that Germans are as noted for their sense of humour as we English are for our extravagant national cuisine. Although, perhaps not by coincidence, we *were* experiencing torrential rain when the Noah arrived at the *SOS* office...

The Audio Went In Two By Two

Noah is available in two versions: Noah and Noah EX. Both are 2U rack units — the original plans for a keyboard version seem to have been put on hold at present. Indeed, the only difference between the two current models, as we'll investigate later, is that Noah has six DSP chips (the same as Creamware's Pulsar II card), while Noah EX has 10. We were sent a Noah EX for review, although I'll refer to both models generically when I talk about 'Noah' in this review, unless otherwise stated.

The front panel has a fairly distinctive appearance with a metallic silver finish screen-printed in navy blue, and rounded off ▶

with a 2 x 40 LCD display, a Compact Flash card slot, some buttons, and a set of rotary controllers, which are coloured, rather tastefully, in black, white, and, er, yellow. To be fair, Noah is quite pleasant to look at, although some of the controls — the four yellow rotary controllers in particular — aren't stepped and do feel a little bit fragile given its overall cost, which raised a few eyebrows amongst colleagues in the *SOS* office. However, I didn't have an issue with these controllers when I was using Noah — it was the headphone control that had my eyebrows adopting a more Vulcan-like shape.

Remote Control

As mentioned in the main text, you can connect Noah to your Windows computer via the USB interface and remote-control the operation and configuration of the hardware using the supplied software. As you'd expect, Noah's software is based on Creamware's SCOPE Fusion Platform (SFP) software, which makes getting to grips with Noah's editor a piece of cake if you've ever used a Creamware system before (for more on this, see the SFP review in last month's *SOS*). Once installed, the familiar Live Bar appears showing all the devices currently running on your connected Noah — which, by default, includes the Mixer, a MIDI Manager, the auxiliary effects, step sequencer and arpeggiator editors, and the currently loaded instruments or effects.

Although you don't get the SFP's Routing window (unlike with a PCI-based SFP system, you aren't able to change the way Noah's devices interact with each other because of the fixed architecture), the editor windows for Noah's instruments are all identical to the desktop versions, and this highlights the big advantage of using the computer-based editor. Rather than having to navigate pages of menus and figuring out which rotary controllers to turn, you can see all the parameters on the screen in front of you in one window, and this really does make life easier. The same goes for setting up MIDI, programming the step sequencer and anything else you might want to do, especially since you can see all these editor windows at once if your monitor is big enough!

However, while editing on Noah is unsurprisingly easier via a computer, the neat thing is that you can edit everything and prepare your presets via the computer, but then disconnect it and take Noah on stage or to another studio, with all of your settings stored.

The only problem I encountered with the remote-control software is that some of the actions you perform in the software aren't always reflected visually on Noah. For example, if I was playing preset 1 on *Minimax* and changed to preset 10 on the remote-control software, I found that its display would still be set to preset 1, although Noah would play preset 10 correctly. This wouldn't be such a big issue, but if I then pressed the increment button on Noah's front panel, preset 2 would be selected rather than preset 11. I couldn't find any way to solve this, no matter how I set up the synchronisation features for communication to and from the Noah and the remote-control software.

Looking to the future, Creamware have some exciting plans for Noah and their SFP software, as used by all other SCOPE systems. With the next release of the SFP software, SFP v4, anyone who uses a SCOPE card and Noah on the same computer will be able to manage these separate devices within the same SFP software, rather than having the SFP and Noah remote-control software running separately, which is how the situation works at the moment. This should add a great deal of flexibility and ease of use when it comes to sharing presets between a SCOPE card and a Noah, because although SFP instruments aren't themselves compatible with the Noah, as mentioned elsewhere in this review, their presets do appear to be.

The computer-based remote-control software for the Noah.

On the front panel is a quarter-inch headphone jack with an associated volume control that doubles as a button for triggering previews of the currently selected instrument and preset. I never knew it was possible to be intimidated by a headphone volume control before using the Noah, but for some reason Creamware have opted to use the same type of endless rotary controller they do for Noah's other front-panel knobs, providing no obvious indication of how loud the headphone output is going to be. The plus side is that the headphone level will always be the same as you left it when you last switched off Noah, regardless of whether you've accidentally touched the headphone control in the interim — the down side is that you won't actually know whether this level will be set to silence or ear-shredding before it's too late.

Turning our attention to the back panel, we find Noah's fairly comprehensive selection of inputs and outputs. For audio, there's two pairs of left and right mono quarter-inch jacks for analogue I/O, with a gain switch to change the sensitivity of the analogue inputs between -10dB and -24dB. When it comes to digital audio connections, there's a dual-purpose S/PDIF-ADAT output, but no 'conventional' digital input at all (there's a computer-based workaround, though, as I'll explain in a moment). This is mostly excusable, since Noah is primarily designed for playing synths, but as it can also act as an effects unit, it seems a shame that the only way you can get audio into Noah without a computer is via an analogue input.

Noah has a built-in power supply with a fan, although it's pretty quiet — you'd notice the fan if you were in a room with just Noah, but it's not at all distracting in the studio. Rounding off the rollercoaster ride that is the back-panel connections are a BNC word-clock input, a footswitch jack, MIDI In, Out, and Thru connections, and a USB port.

Computer Love

While the introduction to this review described the Noah as an instrument that doesn't require a computer, which is true, it's still possible to connect Noah to a Windows-based computer, enabling you to edit and prepare patches using the same attractive on-screen interface designs as if you were running a PCI-based SCOPE system (see the Remote Control box for more information).

But the USB connection does more than just facilitate editing by remote-control — Noah can also act as a MIDI and audio interface for your PC, potentially eliminating the need for an additional computer interface. Although Noah is designed to help musicians do away with a computer in some situations, ironically, it also has great promise for being the centrepiece of a computer-based production system — a Noah with a laptop would be a great portable system for SFP instruments, for example.

At this point, I just want to mention that despite Creamware's advertising for Noah, where the unit is pictured with a sexy Apple G4 Powerbook, I didn't leave anything out in the last paragraph where I mentioned connecting Noah to a Windows-based computer. Unfortunately, Mac support isn't here yet, so Mac users really will have to use Noah as a stand-alone device, although support is planned in the not-too-distant future.

As mentioned just now, Noah can also be used as the world's most expensive MIDI interface, although the serial MIDI (five-pin) ▶

Effects Rack

In addition to the reverb, delay and chorus effects available as aux-send effects, Noah includes an additional 35 insert effects, grouped into Filter, Modulation, Distortion, Dynamics and Other types.

FILTERS

- ***Stereo EQ*** **and** ***Parametric EQ*** **are both four-band equalisers with gain, frequency and Q controls — the only difference is that bands one and four on** ***Stereo EQ*** **are high- and low-shelving filters, while** ***Graphic EQ*** **features eight bands with fixed frequencies.**
- ***Auto-Wah*** **speaks for itself.**
- ***Resonator*** **is a comb filter with an optional LFO, and a ring modulator.**

MODULATION

- ***Ensemble*** **is a simple chorus effect, while** ***Master Chorus*** **gives you more control over the tone of the effect.** ***Harmonic Chorus*** **allows you to process only frequencies above a threshold which you set, and** ***Triple Chorus*** **and** ***Hexa Chorus*** **provide three and six delay lines respectively.**
- ***Master Flanger*** **is a versatile flanging effect that also forms the basis of** ***Random Flanger*****, where the modulation waveform is a random signal.** ***Harmonic Flanger*** **works in the same way as** ***Harmonic Chorus*****,** ***Space Flanger*** **modulates both the delay position and delay length, and, finally,** ***Step Flanger*** **makes it easy to adjust the modulation signal in small to large steps, thanks to a sample & hold feature.**
- ***Master Phaser*** **takes its lead from** ***Master Chorus*** **and** ***Master Flanger*****, and Creamware also include** ***SSB Phaser*** **(Single Side-band), as explained in the SFP review last month, where the same frequency shift is applied to all the signal's component frequencies.**
- ***Auto Pan*** **and** ***Tremolo*** **are pretty standard.**

DISTORTION

- ***Amplifier*** **simulates the characteristics of a tube amplifier, while** ***Overdrive*** **and** ***Tube Processor*** **both model the characteristics of an overdriven tube amplifier.**
- ***Decimator*** **allows you to process audio so that it plays back at any bit depth or sample rate, independently of the rest of the system — Creamware's take on a 'bit-crusher', in other words.**
- ***Distortion*** **is fairly self-explanatory!**

DYNAMICS

- ***Compressor, Expander, Limiter*****, and** ***Gate*** **are separate effects, while** ***Dynamics*** **combines all of these processes into one effect, which is useful since you can only use two insert effects in the signal chain.**

OTHER

- ***Stereo Pitch-shifter*** **and** ***Two-voice Pitch-shifter*** **are both two-channel pitch-shifters, but where** ***Stereo Pitch-shifter*** **can shift the two channels independently,** ***Two-voice Pitch-shifter*** **shifts both channels identically.** ***Feedback Pitch-shifter*** **sends, as the name suggests, an adjustable amount of the pitch-shifted signal back into the effect again.**
- **Finally, both** ***Soft Clip*** **and** ***Stereo Expander*** **are also fairly self-explanatory.**

There's no digital input *per se*, though digital input is possible via the USB connection. Otherwise, though, Noah has the potential to be very well connected, with USB and word-clock sockets, a digital (S/PDIF or ADAT) output, footswitch jack, the full complement of three serial MIDI ports (which work independently of the USB MIDI functionality), stereo analogue input jacks with associated low/high-gain switch, and stereo analogue outputs.

ports on the unit and the MIDI I/O via the USB connection are treated as two separate devices. This allows for a degree of flexibility in that you can trigger different parts of Noah from different MIDI interfaces, but, by default, your computer won't receive MIDI input from a keyboard attached to Noah until you route that input to USB.

When it comes to using Noah as an audio interface, the USB interface provides two inputs (as a stereo pair) from the computer to the Noah, and six outputs (as three stereo pairs) from the Noah to the computer. While the Noah doesn't have any digital inputs, as discussed earlier, it's possible to get around this by using your USB connection. However, Noah's aspiration to be at the centre of your computer-based setup is currently hampered by the lack of ASIO drivers — only ordinary Windows audio drivers are provided at the moment. ASIO drivers 'are planned for a later release', according to the manual, but I was a little disappointed by this, as it does limit Noah's current usefulness as an audio interface.

While some could argue that a device in Noah's price range should really use Firewire, or at least USB 2, instead of USB 1.1, it's worth remembering that there are still some platform issues with the choice of interface at this time. Only the most recent Windows-based computers have USB 2 or Firewire connections, and while all Macs have Firewire ports these days, there aren't many Macs with USB 2 connections, so USB 1.1 is actually the most logical choice for a cross-platform device. Those who require eight outputs from the Noah could always get a cheap ADAT soundcard and use Noah's ADAT out port, for example.

User Interface

One of the design points of the Noah that surprised me was the fact that every function and parameter is accessible from the front-panel controls. Before actually using Noah, I had previously thought of it in a similar way to an instrument like Clavia's Nord Modular, for example, where a computer was required for full editing, and the unit itself would be more of a 'playback' device when used in stand-alone mode. But every function and parameter is indeed accessible from the front-panel controls, which presents both advantages and also a couple of disadvantages.

On the plus side, when you aren't using Noah with the computer-based remote-control software, you can make any adjustment you like to the instrument without having to get annoyed that you don't have your laptop alongside. The disadvantages are that getting to every parameter through the menu system requires a little practice, and what you can do with the Noah is limited to whatever's possible from the front panel, unlike the Nord Modular, for example, where the computer unlocks the instrument's full potential. However, this isn't meant as a criticism to Noah's designers, since I was surprised to find that the interface was reasonably easy to use once I got my head around the way the menus were laid out. One of the things I discovered during this process was that in addition to being able to play an instrument's preset from an attached MIDI keyboard or, when Noah is hooked up to your computer, from a PC-based sequencer, you can also select from two other sound trigger sources: an arpeggiator and a step sequencer.

The Slot System

The challenge Creamware faced with Noah was how to take the open-ended DSP architecture seen in the SFP and adapt it for use in the Noah's more rigid architecture, which is suitable for a stand-alone instrument. One of the great advantages of the SFP is that very few limitations are placed on what the user can achieve: any number of objects can be created and routed to (and through) any other object, until you run out of DSP power. However, such flexibility would be unsuitable for a stand-alone instrument, where the user requires a certain degree of predictability — if you're playing the Noah with a certain amount of polyphony, you don't want this to change if you add an effect, for example. And similarly, you wouldn't want an error reporting that all your DSP resources were used when trying to add an effect.

So to make Noah predictable to the user, Creamware have indeed designed a more rigid architecture. This means that Noah has less flexibility than a computer-based SCOPE system, but also that Noah is far easier to use and understand, which may endear it to some musicians who previously found the SFP too complicated.

Noah's DSP resources are divided into a series of so-called 'slots', and each slot can either be used independently, supporting one instrument with up to two insert effects, or linked to another slot to give more resources to a single instrument. And since

Slots & Polyphony

This table appears in Noah's manual and shows exactly what polyphony you will get for each instrument — note that the three- and four-slot configurations are only available on Noah EX.

DEVICE	1 SLOT	2 SLOTS	3 SLOTS	4 SLOTS
Minimax	3	6	10	13
Lightwave	6	12	16	16
Pro One	2	5	8	11
Six String	2	6	9	12
B2003	16	16	16	16

Creamware's instruments get through more DSP power than Nanette Newman does dirty dishes, the number of slots is quite small: Noah has two slots and Noah EX offers four slots, which makes these devices two- and four-part multitimbral respectively.

When Noah uses each available slot to run a different instrument, this mode of operation is known as Multi Mode. However, as I've already mentioned, slots can be linked together to give more DSP resources to a single instrument, giving you more polyphony for that instrument, and when all of Noah's slots are linked to run just one instrument, this is known as Single Mode. The difference between Multi and Single mode is easily understood on the two-slot Noah, since you either have two slots running two instruments (Multi Mode), or two slots running one instrument (Single Mode). But with the four-slot Noah EX, these modes become a little more interesting.

Single Mode on Noah EX is exactly the same as Single Mode on Noah, in that all slots are used to run a single instrument — the only difference is that whereas Single Mode gives you double the DSP power on Noah, it gives you quadruple the DSP power on Noah EX. However, in addition to being able to run four independent instruments in Multi Mode with a Noah EX, you can also link the available slots in any configuration required. For example, you could link the first two slots and run one instrument with double the DSP resources, while still using two additional instruments independently in slots three and four. You could group slots one and two, and slots three and four, which would effectively give you the equivalent of two standard Noahs in Single Mode; and, finally, you could group slots one to three together, running one instrument with triple the normal DSP resources, and leave slot four to handle a second instrument with the usual single slot of DSP power.

What's In A Slot?

While Noah's slot-based architecture is obviously important, the most crucial factor overall is whether the devices included to run within these slots are actually any good, and the short answer is: they are. The slightly longer answer is that Noah currently includes eight devices, consisting of six instruments and two advanced effects.

The instruments, as mentioned earlier, are mostly recreations of classic synths, including *Minimax* (the Minimoog emulation to die for, as reviewed in last month's *SOS*) *Lightwave* (a wavetable synth based on the Prophet VS), *Vectron Player* (a vector synthesis instrument), *B2003* (a Hammond B3 model), *Pro One* (a recreation of the classic Sequential Circuits synth, which is an optional instrument shipping free with Noah until the end of June) and *Six String* (a physically modelled guitar synth). Although Noah includes a healthy dedicated effects section (which runs in addition to the slot system, and of which more in a moment), two of the effects, *Vocodiser* and *Interpole*, are rather more hungry and are required to run within a slot.

Lightwave uses two wavetable oscillators that model the waveshaping technology used in the Prophet VS. The oscillators are mixed together before being passed through two flexible 12dB-per-octave filters, used either in series or parallel. However, it's in the modulation department that *Lightwave* really shines at creating interesting sounds. Here you can use a variety of sources for modulation, including two LFOs, a multi-stage envelope generator, and most MIDI controllers.

Vectron Player is an instrument that uses vector synthesis, where you can mix the output of four oscillators with a joystick-like control in a way that's conceptually similar to a surround panner. The joystick is implemented across two of Noah's yellow front-panel knobs, or as an on-screen vector control in Noah's remote-control software. As the name suggests, *Vectron Player* can only play back existing presets, but even though you can't create your own, this instrument still provides a great deal of interesting sonic possibilities.

B2003 is a useful Hammond B3 emulation that models 92 tonewheels, key-click, percussion, overdrive and the famous Leslie rotary speaker, and although it sounds good, Native Instruments have already set the standard by which others are judged with their popular *B4* virtual instrument. I have to confess that I'm a not a huge Hammond *aficionado* — while I love that sound, the only organs I've ever played properly (in the musical sense, so no sniggering at the back) are the kind with 32-foot pipes rather than spinning speakers. However, as many other Noah users have noted, *B2003* doesn't seem to have as lively a sound as *B4*, and although *B2003* is a faithful model of a real B3, most people seem to prefer the slightly brighter sound of *B4*. For this reason, Creamware will shortly be updating *B2003* to compensate for this, offering the original *B2003* model in addition to a slightly brighter version.

Pro One is a model of Sequential Circuits' classic synth, and is based on Creamware's so-called Circuit Modelling technology, using two oscillators capable of producing different waveforms simultaneously, which are mixed with optional white noise through a 24dB-per-octave low-pass filter. You can also modulate the filter and amplifier via two ADSR envelopes. And, last but not least, *Six String* uses physical modelling to recreate the sound of a guitar, with two models provided for creating electric and acoustic guitar patches. While *Six String* isn't bad in terms of realism, a good sample is going to sound far more realistic. However, *Six String* shouldn't be considered the poor relation in the collection of instruments because it's capable of creating some unique sounds based on the models that were designed for more traditional guitar sounds. The included presets show off these capabilities admirably with all manner of ethnic-instrument, drone and other curious sounds.

Turning to the slot-based effects, *Vocodiser* is a vocoder which takes full advantage of the audio-input capabilities of Noah, enabling you to use a signal from the analogue or USB audio input as the carrier signal, and while it sounds like an organisation devoted to combating ▶

Some of the software instruments bundled with Noah (clockwise): *B2003*, *Lightwave*, *Interpole* and *Pro One*, which at the time of writing is supplied with Noah, but this is a limited-edition offer. After the end of June, it will be a $99 optional download.

international crime, *Interpole* is actually a rather cool filter that also utilises Creamware's Circuit Modelling process to model the design of an analogue 24dB-per-octave low-pass cascading filter. *Interpole* is especially versatile, and it's great for adding an analogue character to incoming signals and making sounds more interesting with its modulation abilities.

Although the bundled devices are generally excellent, there's unfortunately no way to run any of Creamware's other synths on Noah at the moment, since a little bit of tweaking is required to make an existing SFP device Noah-compatible. As a result, the only Noah-compatible devices currently available are those supplied with the unit itself. One Creamware instrument I would love to see running on Noah is *Modular 3*, although I imagine that this would also be the most complex instrument to port. However, to get around the issue of every parameter being accessible from the front panel, which would surely be impossible with *Modular 3*, it would be great if you could build your *Modular* patches on the computer and have only the 'playback' controls available on Noah, as on Clavia's Nord Modular.

Mixer & Effects

Noah's Mixer is pretty flexible, supporting from three to six channels (depending on whether you're in Single or Multi Mode), which are routed to a stereo Master/Mix channel: one to four channels for the slots, a channel for the stereo analogue input, and a channel for the stereo USB input from the host computer. Each of these channels, including the Master/Mix channel, can be routed to any of the available ADAT, USB or analogue outputs, giving you eight stereo output pairs in total.

When it comes to effects, Noah's send effects are hard-wired as reverb, chorus and delay; and although there's only one choice of reverb (based on Creamware's *Masterverb* SFP plug-in), the chorus can be switched to a flanger if required, and there are five types of delay to choose from, taken from Creamware's extensive collection of SFP effects. These effects are very usable and add an extra dimension to the otherwise dry synth sounds — the delay effects are particularly useful since, as with the normal SFP versions, you can set the delay times in terms of millisecond or note values, with the tempo based on the Noah's internal setting or an incoming MIDI Clock signal.

Each channel on the Mixer has three auxiliary send controls, and there are three master send controls on the Mix channel to control the overall amount of signal being sent to the effects, rather than sending a portion of the Mix channel to the effect, along with three return controls to set the level coming back from the effect, and buttons for disabling the effects altogether. Since each channel on the Mixer can be routed to a different output, it seems a shame that the output of the auxiliary effects can only be routed to the Mix channel, and the obvious workaround — routing a channel to a different output from the Mix channel so you're only left with the effects on the Mix output — doesn't actually work.

Noah also allows for the use of two insert effects from a choice of 35 (see the Effects Rack box for more information), and although there is a selection of chorus and flanger effects here, you can't use a reverb or delay as an insert effect. Although each channel, including the Mix channel, can accommodate two insert effects, only two insert effects can be used on the Noah Mixer simultaneously. This means if a channel is using the first insert effect, no other channel can use the first insert effect space without it being reassigned from the other channel, although there's no problem in using the first insert effect space for an effect on one channel and the second insert effect space for another channel.

While the way in which effects are allocated in Noah sounds potentially limiting at first, it didn't appear to be when actually using the unit. While Creamware could arguably have allowed the user to run more effects when only a single slot is in use, for example, this would have complicated the use of the instrument, making the experience less predictable, as discussed

Test Spec

- Noah EX OS/Firmware revision reviewed: v1.0 (dated 6/5/03). Version 1.3 was made available at the very end of the review period, as detailed in the Stop Press box, but not in time to be evaluated for this review.

earlier. As it stands, you know you're always guaranteed five effects, no matter what you do with Noah.

A further nice touch with the mixer and effects is the ability for them to be automated via MIDI Controllers, as described in the Technical Reference Manual. The functions of the instruments, arpeggiator and step sequencer can also be similarly automated. In the case of the mixer and auxiliary effects, these generate outgoing and respond to incoming MIDI data, which can be configured in the MIDI Manager, part of the computer-based Noah remote-control software described in the Remote Control box.

Stop Press! Firmware Update v1.3

Just after the Noah EX we were sent for review had been whisked away from the *SOS* offices, an email from Creamware arrived containing a firmware update to upgrade the Noah's operating system to version 1.3 — the review model had v1.0. As a result of this unlucky timing, it wasn't possible to put this update to the test (it would have been nice to know if the discontinuities I observed between the Noah's display and the remote-control software had been fixed, for example), but the revision notes indicate that Creamware claim to have fixed three potentially annoying problems, which answered a few suspicions I'd had while playing with Noah.

The v1.3 update should apparently eliminate occasions where you get the odd stuck MIDI note, or random MIDI notes or delays during playback. I had noticed this happening a few times, but wasn't sure if it was to do with the way my sequencer was communicating with Noah or Noah itself. According to Creamware, the new update should also fix random freezes, which sounds rather serious, but wasn't something that happened during my time with the instrument.

Here Comes The Flood

There isn't anything I really don't like about Noah, and there are many, many things I *do* like. Nevertheless, there are a handful of issues that have stuck in my mind, preventing me from writing the unabashed praise I'd be tempted to write by just listening to the sounds coming from the unit. On the one hand, Noah is a great concept done well: the remote-control software adds a great degree of flexibility, the built-in audio and MIDI USB interface is a nice touch, and the instruments sound great, which is surely the most important point.

On the other hand, there are some niggles with the first versions of the operating system. Creamware claim to have fixed these in the v1.3 firmware upgrade, but there wasn't time to test this during the review period (see the 'Stop Press' box above). I also heard from other users who've had their Noahs already that there have been some clocking oddities and incompatibilities with the remote-control software itself — but it's important to point out that I didn't experience any of this with the Noah unit *SOS* was sent. The current lack of Mac support is a shame, since so many musicians use Apple-based laptops, and the USB audio interface aspirations hadn't fully materialised at the time of this review, with the ASIO drivers just approaching the beta stage. However, I think I'd probably overlook these more minor points if the unit wasn't as expensive, or perhaps if Creamware had ported more instrument devices to Noah (which they say they plan to do at a later date when it has become more established).

Considering the price issue in more detail, with a UK retail price of £1599 for its basic version, Noah is an expensive product; and it's actually at one of those awkward prices where it's not over-priced, but neither is it the bargain of the century. For £1599, you could now buy a Clavia Nord Lead or an Access Virus, for example, or a new Nord Modular G2 with a fair bit of change for the bus ride home — and while I've made several comparisons to the Nord Modular in this review, I do think that if Creamware got *Modular 3* running on Noah, it would be the potential killer application. This would allow musicians to compare Noah with the Nord Modular and say 'Hey, look how much more Noah does for only a little more money,' easing the way Noah's price tag is viewed.

The real problem with the issue of cost, in my opinion, is caused by Noah EX, because I think most people who are looking to buy a Noah will opt for the more expensive version. The reason is simple: while Noah EX costs £400 more than Noah, Noah EX effectively gives you a second Noah for that £400 (with the exception of effects); and if you were going to spend £1600 on Noah, I think you'd be mad not to buy a Noah EX instead. The point here is that the difference in price between Noah and Noah EX needs to be greater; I think lowering Noah to around £1300-£1400 would tempt more musicians to purchase the cheaper model.

However, these points aside, I don't want to end on a negative view of Noah because I really, really like it. As should have been more than apparent from the SFP review in last month's *SOS*, I'm a great fan of Creamware's synths, and there are many other useful practical aspects to the Noah as well. It's great being able to have such quick access to *Minimax* and other instruments when you just want to come up with some ideas, and you could use Noah alongside your SCOPE card for those times when you want to add a few more instruments. There's also the portability factor for live performance, which is previously unheard of in a Creamware system. I have to admit that I'm a confirmed advocate of software-based systems, and I wouldn't normally be seen dead with a 2U hardware rack unit or effects unit in my studio. However, I'd make an exception for Noah — it's a highly desirable instrument. SOS

information

£ Noah £1599; Noah EX £1999. Prices include VAT.
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Blue Sky Pro Desk

Active Monitoring System

Blue Sky provide their innovative speaker technology, first seen in the Sky System One, in a new and more compact system.

Paul White

The Blue Sky Pro Desk system belongs to the same family as the somewhat larger Sky System One reviewed back in *SOS* February 2003, the obvious difference being its lower power rating. The Pro Desk 2.1 stereo system under review here comprises a pair of two-way satellite speakers (model Sat 5) plus a single subwoofer/bass unit (model Sub 8). A Pro Desk 5.1 system is also available for surround applications, and owners of 2.1 systems can upgrade their systems for 5.1 use at a later date if they wish.

Blue Sky's approach differs from that of other manufacturers insomuch as the sub is not an add-on to extend the low end, but rather handles all the low end for the system in much the same way as the bass driver in a conventional three-way system would. The satellite speaker drivers are fed an 80Hz, 24dB/octave high-pass filtered signal via integral active filters and so behave exactly like the mid-range and high-frequency drivers in a three-way system.

Sub 8 Subwoofer

Both satellite speakers and the subwoofer have IEC mains sockets to provide the necessary power, while conventional balanced XLR cables (not supplied) are used to connect the satellites to the sub. All the subwoofer's connections and controls are on the rear panel, along with the power amplifier's heat sink. The audio feed enters the sub on a pair of balanced XLRs and there are two more XLRs providing an input to, and an output from, the sub for integration into other systems or for driving an additional subwoofer. A gain knob controls the level of the subwoofer, with a calibrated reference position, and a computer-style connector allows an optional remote volume control (called the Functional Volume Control) to be connected, a feature that will be particularly welcomed by anyone trying to set up a computer-based system without a mixer.

Overall, the subwoofer cabinet measures a modest 13 x 16 x 15.76 inches and it sits on four screw-in conical feet. It's pretty heavy at 22kg, not least because it is built from 0.75-inch MDF with an inch-thick baffle. It is powered by a single eight-inch, long-throw ▶

SOUND ON SOUND

Blue Sky Pro Desk 2.1 **£999**

pros
- Easy to install.
- Well-integrated bass end with deep bass extension.

cons
- High end wasn't quite as smooth as I'd have liked, but many users might actually prefer this to absolute neutrality.

summary

A practical and affordable full-range monitoring system that should fit the space and budget of many a project studio.

BLUE SKY PRO DESK

Second Opinion

Having reviewed the larger Blue Sky Sky System One, I was intrigued to see how its younger and smaller sibling stacked up. I found Sky System One a mildly frustrating product back in those dark winter days — the classic curate's egg. Really good in parts, especially its bass performance, but let down partly by being balanced a little too far on the bright side of neutral and mostly by displaying a distinctive and not particularly pleasant mid-range coloration. It was a shame, because the level of engineering competence and value for money in System One was impressive. I particularly liked the decision to go for closed-box loading for both sat and sub, the choice of tweeter, and the design of the bass/mid-range unit.

Pro Desk still features closed-box loading, with all the time-domain and dynamic advantages the technique potentially offers over reflex loading. It still incorporates some impressive driver technology, including the recently launched 19mm version of Vifa's ring radiator tweeter, and it's still dead easy to install and use. However, I liked it much more than System One, and if I were in the market for a compact active satellite and subwoofer monitor system I'd find it a hard proposition to turn down. Pro Desk shows no sign of the characterful mid-range that tainted System One. It's still a little on the bright side, but is open sounding and uncoloured in a way that the bigger brother could only hint at. And with bass performance and sub-woofer integration that seems even better the second time around, Pro Desk is a very competitive product — frighteningly so, I'd have thought, if you're one of the competitors!
Phil Ward

driver with a two-inch voice coil and, unusually, this has an aluminium cone. The amplifier, a discrete bipolar device, is rated at 100W and in a typical room it covers the 20-200Hz part of the audio spectrum. A fabric grille covers the front of the unit.

Sat 5 Satellite Speaker

Because the satellite speakers only have to handle from 80Hz upwards, they are relatively compact at 10.88 x 6.62 x 10.23 inches, but again they're heavier than they look (11kg), because of their heavy MDF construction and integral power amplifiers. Their bases are fitted with threaded inserts that are compatible with industry-standard Omnimount stands and brackets. The tweeter has a curious nipple-like profile, described by Blue Sky as dual concentric and is, like so many of the best nipples, of Swedish origin! Its ferrite magnetic assembly is magnetically shielded, and the design utilises a soft diaphragm 0.75 inches in diameter within an integral waveguide.

Like the subwoofer, the 5.25-inch mid-range driver has a concave aluminium cone, this time driven by a 1.5-inch voice coil. This driver is also magnetically shielded, and is driven by a 60W amplifier with an electronic crossover, though no details of the crossover characteristics are offered. Each satellite has a rotary gain control with a reference position, and the quoted frequency response extends from 200Hz to 20kHz (±3dB). When set up with the sub, that means it's possible to achieve a system response of 20Hz to 20kHz, though the exact figure depends on the room characteristics to some extent. A switch on the rear panel allows the 80Hz high-pass filter to be bypassed, allowing the speakers to be used in

different systems where the bass/mid-range crossover is placed before the speakers in the signal path.

Performance

I set up the system with all the gains in their calibrated positions, and in my room this produced the best overall tonal balance. It didn't take long to find a location for the sub box that gave a nice even bass on all notes, after which it was just a matter of selecting material and listening. The first thing you notice about this system is that it has tremendous bass extension, though the bass doesn't sound deliberately hyped — it's just that everything has so much depth. Dance music engineers and producers should appreciate this especially, as it provides some insight into how the bass balance will hold up on a big club sound system. What's more, the bass end doesn't feel detached from the rest of the spectrum, a failing I've noticed in some other systems that use a sub.

Higher up the spectrum, the sound is clear and very well-defined, though perhaps just a trifle bright for my own taste. However, it's nothing too serious, and I imagine the speakers were voiced this way because a lot of people like to work with this kind of sound. The stereo imaging also holds up well and, even if you place the sub box right over to one side, there's still no sense of anything being out of place.

The Blue Sky Pro Desk provides a compact and sensibly priced solution for anyone in the UK wanting full-range monitoring in a project studio where there isn't room for conventional full-range monitors. This system behaves like a physically much larger installation and it can play loud as well as deep. You could also do worse than use the system as the front half of a home cinema surround system — I tried this for a few days and was well impressed. Because this isn't a 'two baby full-range speakers plus a sub' system, but rather a three-way system where the bass speaker just happens to be in a separate box, there is none of that lower mid-range boxiness you sometimes get from other small speakers and a great deal more bass extension than you get from most sensibly sized full-range monitors. What the speakers lack in high-end smoothness, they more than make up for with solid, dependable bass end, and the bottom line is that the sound you get is very much a sound you can work with. SOS

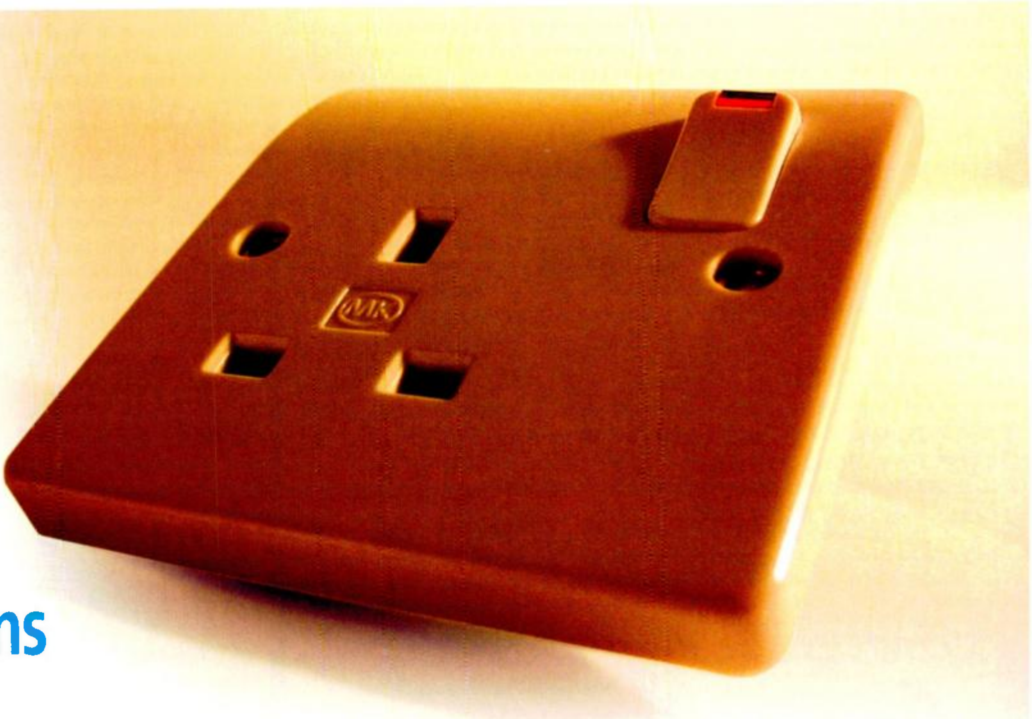
Blue Sky Loan Scheme

If you like the sound of this review, and you fancy checking out the sound of Blue Sky's monitors for yourself in your own studio, Sonic Distribution in the UK are willing to loan systems free of charge for evaluation, before purchase. This offer is available through participating dealers only, and a list of these is available by contacting Sonic directly. Terms and conditions apply, which differ depending on whether you're an account holder with the dealer in question.

information

£ Pro Desk 2.1 system, £999; Pro Desk 5.1 system, £2460; Functional Volume Control, £82.25. Prices including VAT.
T Sonic Distribution +44 (0)1525 840400.
F +44 (0)1582 843901.
E james@sonic-distribution.com
W www.sonic-distribution.com
W www.abluesky.com

In The Mains



Identifying & Solving Mains Supply Problems

Ben Duncan

Most of us never have trouble with the electricity in our studios — but as with everything we take for granted, fixing problems can be difficult when they *do* arise. We look at some of the main problems and solutions.

The mains (AC electrical) supply in the UK is one of the most stable, generally pure and reliable across the world. The downside of being so lucky is that unless there are blatant problems, the energy supply becomes transparent — we readily forget just how reliant we all are on its smooth continuity for musical creativity.

Why Be Troubled?

Problems with the mains supply may be easy to measure but go unrecognised, or they may be hard to measure, but have a surprising effect on the sound quality you get. What constitutes a problem depends on the sort of equipment you are using, and its sensitivity to varying degrees of imperfect mains. The complexity of all the equipment you're using may well mean that you prefer to stay focused on producing music, and never find time to look into optimising mains connections. This is fine if your problems are trivial, but what if they're not? Your money and time can be wasted buying successive pieces of equipment, and finding you're never quite getting the results you wanted, while all the time the problem is elsewhere — in the quality of your studio's mains supply.

Another scenario is that you buy 'mains-improving' equipment that achieves nothing. This may be because it's a 'snake-oil' product (meaning it will do very little under any circumstances, but having paid for it, you find yourself believing it

Mains Problems With Foreign Equipment

Making equipment that works smoothly on any national AC supply is not hard, but it requires a little knowledge and also forethought. The first detail is an understanding of how mains voltages vary across the world. More than one overseas manufacturer has been fooled into thinking that the mains voltage in the UK (& Eire) has been reduced to 230V, and some are careless enough to sell products made for 220V for use EU-wide. These may be stressed or unsafe when used in the UK & Eire.

The second, more major problem is that the US supply frequency is 60Hz — compared to 50Hz in the UK, Europe and many other territories. The effect of the higher frequency is that mains transformers for use in the USA (and Canada) can use smaller magnetic cores and less turns of wire to do a given job. This gives higher-powered US-made products (where transformer cost is a substantial part of the whole) a 'triple-whammy' unfair economic advantage — of lower size, weight and cost — in the world market. It also causes endless problems for users in 50Hz territories, including Britain, Ireland, most Commonwealth, and all western-European countries. It's easy enough to provide transformers with correct voltage tappings, but unless the transformers for use in the international market are designed to operate at 50Hz, a bigger problem remains. That's because when a transformer is tightly designed down to a price so that it only just works at 60Hz, then even if the mains voltage setting is correctly set, the transformer will run hot(ter), draw a higher background 'magnetising' current, adding to studio mains pollution. The saturated core will also generate high third-harmonic noise, meaning added levels of buzz at 150Hz. Such a transformer is also likely to buzz acoustically loudly. The sole remedy is to fit a correctly made, 50Hz-rated transformer, which is not cheap. It's worth noting that this is most likely to arise as a problem with grey or illegal imports, and also with equipment that is newly imported to the UK. Established and reputable US and Japanese manufacturers are less likely to forget to design and fit transformers made for 50Hz to export products — but the buyer should still beware.

However, using a transformer at a *higher* frequency than it was designed for will rarely disrupt it; thus most 50Hz transformers will work fine at 60Hz. Effectively, products made to work on UK and European 50Hz mains are 120 percent overdesigned and more expensive when sold into the US, and likewise, US products can be cheaper because they've unwittingly (or not!) sold us an under-designed transformer. Naturally, because imports from 50Hz countries into the US are unlikely to heat up, buzz mechanically or generate noise, American designers remain blind to the grief/hassle that their unadjusted 60Hz products cause to the rest of the world!

must have enhanced something) or equally, because it's just the wrong equipment for the problem at hand. This article is intended to help you identify the sort of mains-related problems that will affect sound and what you may be able to do to rectify them.

Voltage Problems

One of the more common causes of studio gear misperforming is high (excessive) mains voltage. This can cause the following symptoms:

- Acoustic buzzing from transformers. This may come and go with high voltage, and is a sign that a transformer is getting saturated (for more on this, see the box on page 232). This is harmless, in the sense that it's reversible, but upsets the proper operation of transformers.
- Gear running hotter than it's designed to. In some cases, equipment may be failing regularly, or likewise blowing fuses.
- 'Lifeless' or 'gutless' sound over monitors.

A series of light bulbs blowing prematurely is a tell-tale sign that voltage is higher than ideal on a regular basis. In the UK, the official 'declared' voltage is in practice 240V (although it's officially described by the EU as 230V with a different 'tolerance'). In the real world, the mains voltage at your studio's intake varies anyway with loading. In ideal circumstances — a 100-percent healthy mains source — it will centre on 240V and vary from say 235V to 246V throughout the day, as other users switch on and off. Most users are unaware of the mains' continuous and ceaseless fluctuation. Whenever the supply's loading in your immediate block, village, street, or the greater area is low (particularly late at night), the voltage will tend to rise above 240V. The maximum legally allowable is 254V, but voltages as high as 275V have been recorded.

High-voltage Problems & Solutions

You can tell how likely you are to have problems with excess voltage if you can find out how far away from your studio the nearest electrical sub-station transformer is. If you are right next door, you can expect your studio gear to be regularly subjected to mains voltages above 250V at the times of lowest public power usage.

To know for sure if the mains voltage is too high, you need to measure it. Unless you're electrically competent and have time to make lots of readings, it's safest and easiest to hire an electrician to set up some voltage monitoring (logging) at your supply's intake.

There are three ways to deal with excess voltage.

Firstly, if the supply repeatedly exceeds 254V, then your REC (Regional Electricity Company) is legally required to take steps to rectify the situation. They will be most likely to take rapid steps if the mains is seriously high, say continuously above 255V, since there would then be an element of increased fire risk, and the REC would be seen to be liable.

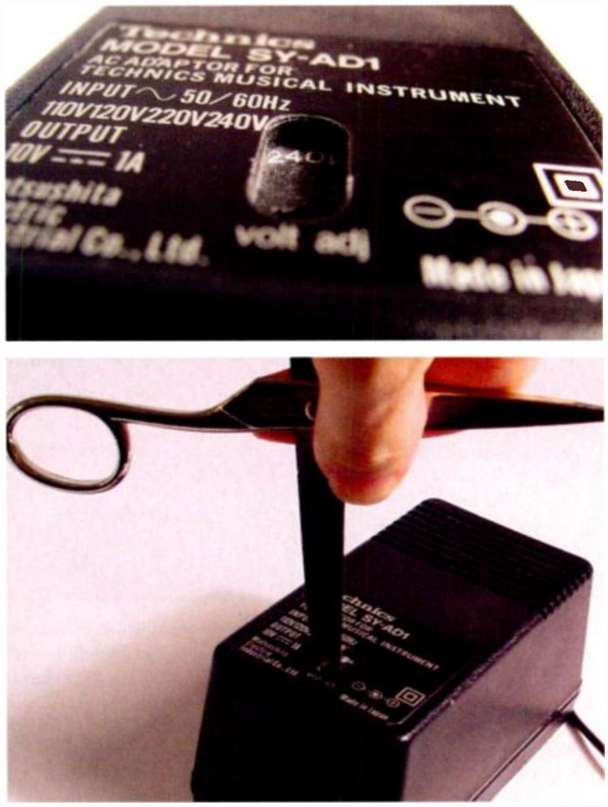
Secondly, you may be lucky enough to own older equipment with mains voltage adaptors allowing fine adjustments. On such gear, you can move a plug on the rear panel to 200-210, 220-230 or 240-250 volts, for example, or sometimes the external power supply itself is adjustable (see picture below). Today, however, the problem of raised mains voltage is one that most equipment manufacturers don't even recognise. Instead, most equipment is presumed to work smoothly (without adjustment) over the 200V to 260V worst-case range of voltages that occur across the UK and Europe. As most manufacturers test their equipment in the daytime, when the supply is most heavily loaded, few will ever encounter the effects of high mains. And yet how many musicians work at night, when the voltage is at its highest?

Third, an easier approach than either of the above is to use an overall suitably rated voltage-adjusting transformer that's able to power all your studio gear. Usually we think of transformers as making big voltage changes, for example 240V down to 15V, but the purpose here is to reduce the supply voltage by just a few percent. However, in order to power a small studio, with its high current peaks, transformers of this type need to be quite large, and they're not normally available off the shelf. What's more, you may find that they reduce the voltage too much at other times of day, although the side-effects of a low voltage, as we'll see next, are less serious.

Low-voltage Problems & Solutions

The UK's 240v mains can legally fall as low as 216V. In my 19 years of product and review testing, I've found that most audio gear will work to considerably lower voltages with few or no ill effects or functional degradation. The odd exception — a tape machine here, a power amp there — will suffer problems before the legal minimum voltage is reached. Symptoms and problems include relays not closing or opening (which affects controls, mechanical functions, powering-up sequences and so on), and audio signals clipping in power amps at maybe 1dB lower than you would expect (earlier overload = lower maximum output). You might also experience hum and related noise, as regulated supplies lose their ability to regulate and smooth the equipment's DC power sources. Finally, low voltages can cause some appliances with motors to burn out, particularly fridges and freezers. This won't affect your studio, but can add a domestic thrust to the desire to get the problem rectified!

If you suffer these problems, you should again have your supply voltage logged at the intake by an electrician, as above. If the supply is regularly going below the 216V legal limit, the REC is required to raise the supply so that it doesn't. But before notifying the REC, make sure that your voltage logging tests are taking place at the supply's intake, as near to the REC's (main)



Some musical gear can be adapted to run on various national voltages, either by changing over plugs on the equipment itself, or by altering an adjustable power supply, as here.

fuse as your electrician can get. This is because the voltage can drop significantly further between there and your studio.

You can avoid this drop in voltage between your house's main intake and your studio by giving your studio's electronics its own dedicated supply (a radial or spur, rather than the usual ring circuit), and reduce voltage losses by using lower-resistance wiring than usual. If your studio is located a fair distance — say 100 feet — from the supply intake, surprisingly fat wires will be required. These will be nominally rated at unfeasible-sounding high currents (maybe over 100A), but what counts is the low total resistance. In electrician's language, very high current wiring is required to meet the part of the IEE (Institution of Electrical Engineers) regulations, which cite a maximum four-percent voltage drop between the mains intake and the furthest fixed outlet, when operated at the anticipated current. However, what electricians will find hard to understand is the high current peaks that studio gear draws (see below).

"There are few accessories for mains-related problems which completely fail to work, but some will be sold to you when their best possible effect is well down the list of your setup's priorities."

All UK studio gear should now come with a fused, moulded UK plug — if not, you should return it at once. For older gear with plugs you've fitted yourself, it's worth ensuring you upgrade to decent brands with the British safety standards kitemark.

Subtle Problems

It's a little recognised fact that nearly all studio gear pollutes its own mains supply. It does this mainly by drawing a peak current that's a lot higher than the average level. A particular studio — including several PCs — may consume only (say) 750W in all, which is about 3A at 240V. This is the average level, and an electrician, from theory, would expect the peak current to be about 4.2A. But in fact, it's more like three to five times bigger (depending on the quality of your local supply); 9A to 15A. You'll be pleased to know that this has no adverse safety implications, because although these high currents repeat 100 times per second, they don't persist long enough to cause any overload. However, the 'spiky' loading that the high peaks represent causes extra voltage drop as well as harmonic distortion. The latter can be seen when almost any mains waveform is inspected — the smooth peaks of a pure sine wave are absent and replaced by less tidy shapes. This is especially true of current waveforms. It means that the mains voltage that the gear is itself using includes harmonics of the mains' fundamental frequency (50Hz in UK and the EU). The added frequencies (mainly odd-numbered harmonics, at 150, 250 and 350Hz) are the primary causes of the richness and complexity of hums and buzzes.

The resulting 'squashed-peak/clipped distortion' can also cause saturation in audio equipment's mains transformers — leading to effects similar to excessive mains voltage. You can now see your studio afresh as a high concentration of DC supplies that creates a localised area of degraded, noisier mains — with high harmonic content. Simplistic brute-force 'solutions' to this problem, like regenerating clean mains just for the studio (as touted by some US manufacturers) don't work very well, since it is the audio gear itself that's causing the problem.

Blatant Problems

If you have the following, something is badly amiss:

- Regular or repeated total outages (power dropouts);
- Periods of sudden and quite deep voltage dip, noticeable as periods when lamps dim and some equipment may malfunction or stop.

Such supply 'outages', as they are known, signify a problem that could be serious — for example, something involving arc'ing, which might cause a fire. For this reason alone, your REC should be keen to rectify matters promptly. Before reporting to them though, do first make sure that neighbouring users are suffering the same problems at about the same times. If not, an electrician should instead be called in first, to check that the cause is not in your premises, past the intake point where the supplier legally hands over to you. No attempt should be made to cure problems as extreme as these above, by using any power-conditioning accessories or techniques. Doing so is like using any good tool for entirely the wrong job.

Having dealt with the most common problems, we can now look at power-conditioning products in context.

Uninterruptible Power Supplies

Also called backup power supplies, these devices will mainly prevent power interruption and also deal with deep voltage dips. While achieving this, supply quality (for audio gear) will usually be low, due to high harmonic and radio-frequency (or RF) noise levels. Ordinary mains can be purer.

As the mains can fail unexpectedly at any moment, some form of UPS can be vital for computers, other digital devices with volatile memory and hard drives, to give you a chance to save your work before the power runs out.

An important aspect of choosing any UPS is that the stated 'VA capacity' of the device needs to be about five times greater than the nominal power rating of the equipment being connected (in Watts or VoltAmperes). This is because of the high peak currents that audio gear, PCs and other technical equipment draw. If you do not make this allowance, the UPS may or may not be harmed, but the gear will certainly be 'power-starved' and may work sub-optimally.

Mains Filters

Also called 'filter blocks', 'mains cleaners' and similar trade names. The common types (if they work at all) remove noise only at radio frequencies well above audio. As you cannot hear such noises directly, nor always know whether they may be strong in the vicinity, it's often impossible to tell whether or not mains filters will achieve anything.

Despite this, they are sold in some instances as if filtering is a universal cure-all. You are most likely to benefit from them if your recording room is near to any RF (radio-based) equipment that's well inside the frequency range that the filter happens to cover. This could result from a nearby broadcast aerial alongside an ambulance or other public-services station, to an FM mains-connected baby alarm in the house next door.

Mains filters commonly 'backfire' by coupling noise into the equipment, which is exactly what they're supposed to deal with! This happens because the mains earth wire that the filter's 'earth' terminal has to be connected to doesn't look like an earth to the RF signals that the filter is trying to deal with. Instead, it can look more like an aerial! Also, when several items have filters, all the RF noise that's picked up in cable shields and by metal cases is summed together, but has no particular place to go. Meanwhile, each filtering capacitor inside contributes a current leakage (including noise) from the live side, through the mains earth wiring. Worse, due to thoughtless design, the use of many items of gear with RF filters may cause any Residual Current Devices you have in your fusebox (or trip/safety switches, as you may know them) to be tripped out, preventing you using your equipment, or meaning you have to remove the RCD, and forego the safety it offers. The bottom line is that RF filters are best avoided. If they must be used, get an electrician's help (or that of another expert) to isolate the filters from the existing mains earth wiring and to connect them both separately back to the mains intake earth point, and also to a solid RF earth (there's more on RF earthing later in this article). If you have much equipment fitted with mains RF filters, this gear may have to be powered from separate circuits if you want to power it through RCDs, for personal safety in the studio.

What Is Transformer Saturation?

As the name implies, when a transformer is saturated, the magnetic core that carries energy from one side of it to the other gets choked up. The transformer can't pass as much energy as may be demanded of it, and this can make the equipment mis-perform in a range of quite subtle ways that add up to degraded musical performance. When saturation occurs, the transformer generates large amounts of third and fifth harmonics — effectively adding distortion to the mains current and voltage waveforms. This then pollutes the supply locally.

The mains transformers used in studio gear will all saturate above a certain mains voltage level. As a result of keeping costs down, the transformers in some makes saturate when the mains is only ever so slightly high. Others are able to take higher voltage levels with no problem, since the makers have spent money to make them work properly over a real-world range of conditions. Some transformer types, particularly high-quality toroidal types, saturate suddenly (at a given maximum voltage) while with others, the saturation is more gradual.

Voltage Dependent Resistors

VDRs are also called 'surge (or spike) suppressors', and similar trade names. These are commonly disc shaped, wire- or tab-ended components that can work rather like an audio limiter, limiting (ie. clipping off) spikes that occur on the mains supply. They are commonly sold in a pre-connected format hidden away inside mains filters, mains adaptors and extensions. The combination with a mains filter can be good, as the filter's inductors helps the VDR to clamp some surge voltages better. A simple type of VDR product is one mounted between Live & Neutral inside a mains plug which plugs in, but isn't connected to anything. This arrangement connects the VDR across your supply without you having to make any wiring changes. British 13A plug pins, with their large contact area and high contact pressure, are particularly suited to this, but it's a rather less stable and workable solution for the weedy pins on most other national plugs, like the European type shown below.

VDRs will provide some protection against outright damage from lightning and switching surges when your local supply is re-energised during a storm or after a system breakdown. But VDRs offer no protection from smaller or longer-lived excess voltages. Typically, they can reduce spikes of thousands of volts to a peak level of about 700V, but no lower. And 700V is still quite damaging, compared to the official peak value of 240V mains, which is 340V. In fact, it's only sustainable if it occurs for a few millionths of a second, just as some people can pick up a hot object without harm so long as they're very quick. As with filters, VDRs can make noise problems worse if they involve connections (from either Live or N, neutral) to the mains earth, unless the earth wiring is sturdy and also happens to be close to the mains earthing stake. VDRs are sometimes held to make sonic enhancements which don't square with levelling a few spikes. The reason is that they also perform as small capacitors that may help suppress, redirect or alter RF signal levels in the wiring. In other words, they can do the same job as an RF mains filter.

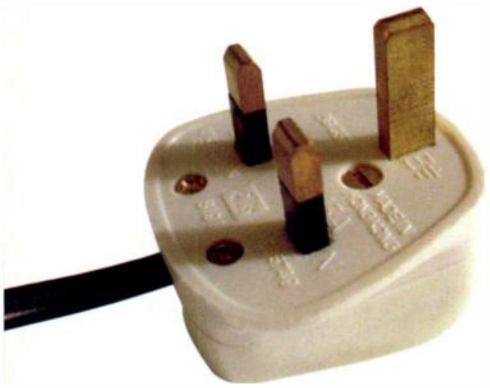
The upshot is that VDRs are mainly benign, but also only one tool in the mains enhancement armoury. They are most likely to be useful if you live in a rural area, where lightning strikes have ready access to power lines. They can also help when you're near a factory or other establishment using lots of electric tools, motors, welders, or in a tower building near the lifts, for example. These operations can create a series of spikes on the mains that the capacitance and level-limiting efforts of VDRs should at least help to curtail.

Heavy-duty VDRs may be placed at the supply intake, with a direct, very heavy-duty connection to the earth. Such 'lightning

protectors' cannot protect from a direct lightning strike (nothing can), but they do give enhanced protection to rural studios if a strike occurs to the overhead mains wiring, provided it's no closer than 50 feet away.

Enhanced Mains Connectors

These may include better grades of standard connectors, particularly ones with gold-plated contact surfaces. These may seem to make little sense until you realise that most of the time (outside the mains' peak voltage points), the mains connection to your gear can be acting as an aerial for all manner of RF signals that pervade the air around us. At this point, mains plug and socket contacts have only tiny leakage and



transformer-magnetising currents flowing in them, and can easily act as detectors (demodulators) of radio signals (when high current is flowing, it can break through the layer of grime and oxidised metal on the contacts, but when the current level is low, the oxidised grime's radio-detecting effect can be stronger). The detected signal then makes its way into the audio path, adding background noise. It is therefore unsurprising that some people who fit gold-plated mains plugs, which don't corrode or oxidise, report enhanced sonic clarity, depth and the like. However, if the area in which your studio is located doesn't suffer from the kind of RF problems that cleaner connections reduce, you may reasonably regard a mains plug with specially plated pins as 'snake oil'. On the other hand, if there's an AM transmitter nearby (long wave, medium wave, or short wave), such plugs could have quite an effect. Even so, if the sort of music you're involved in doesn't benefit from depth or inter-transient silences, you might hear nothing at all.

A practical approach for studios with lots of gear is as follows. When you have a major maintenance session about once a year, upgrade your cheapest mains plugs to decent domestic ones, such as MK or Duraplug types, and wipe down all the mains plug pins with Caig Labs' Dexoit, or any other decent contact enhancer. Also, when equipment is unused, it can be good to leave it plugged into a mains outlet (but switched off). This protects both the plug and socket pins from oxidation and the grime build-up that causes the radio signal/noise detection problem.

Shielded Mains Cables

These are like shielded signal leads, in principle. The braided metal shield intercepts ambient RF signals, but the mains earth wiring can let down the effectiveness of the necessary earthing at radio frequencies — unless you have a low-impedence RF earth connection (more on this in a moment). This matter affects the shielding of leads carrying audio signals less, because the signal connections have higher impedance — so good results can be had without needing such a low-impedance earth. A metal cable shield around mains wires can also create dangerous situations, for example if it gets disconnected, touches live, and pokes out of the mains plug, for example. Therefore extra work is required to sleeve the cable ends, regularly inspect cables, and so on.

The upshot is that metal-shielded mains wiring is recommended only in extreme situations (say, if your studio is next to a powerful radio transmitter), and only when professionally installed (an electrician is probably needed to help you install a low-impedance RF earth, for example).

Having said all that, a new class of shielded cable (called 'Lo-Rad') has recently been developed by Jenving in Sweden using conductive plastic, which is is designed to reduce any shock risk enough to be safe for home studio and domestic use.

Woven Mains Cables

Typically protected inside plastic tubing, cables of this type comprise several loosely and openly woven strands for the live and neutral sides, with the mains earth wire running up the centre. This arrangement may be hidden by curling and placing inside a jacket, but the hosepipe size of the resulting cable gives the game away. The process of weaving the cable is extremely tedious without special, huge jigs — manufacturers have huge, Victorian-style weaving machines — and towers that cost enormous sums. This is one reason why this sort of cable sells for such high sums (usually over £20 per metre).

The apparent benefit of these cables is that they provide a ratio of inductance to capacitance (also called 'characteristic impedance') that is quite different to the ordinary wiring that precedes and follows the short lengths you might use in a studio as gear cables. The result of this is that RF noise already on the line is filtered out and/or damped down. Also, due to the weaving (the live and neutral are broken into many strands which repeatedly cross each other at right angles, so that any mutually induced fields cancel) such cable acts as a sort of 'anti-aerial', meaning that the pickup of ambient RF is reduced.

Isolation Devices

Also known as audio-grade isolating transformers, isolation devices are specialised mains transformers which introduce no change in voltage and stop noise leaking through to the audio path through the mains connections, by 'balancing' the mains leading to audio gear, so both sides (live and neutral) are equally 'above' earth, over a wide range of frequencies. This can make a massive difference to the sound quality in a studio, and is achieved by reducing what's called 'common mode noise leakage'.

Spur Wiring

As described earlier, this involves providing your studio with a more direct, dedicated, mains connection. The beefier wiring helps reduce the voltage drop and interaction between equipment caused by the high peak current being drawn. With a home studio, the spur (which should be a dedicated supply with no loop to tempt circulating

A contact enhancer like Caig Lab's Deoxit really can help to reduce RF interference if used to keep your plug pins clean on a regular basis.

Though essential for electrical safety, the mains earth connection is also a common source of RF contamination.

earth currents, which only add extra hums and buzzes) also reduces noise from and interaction with other appliances that are being used on the house's other circuits. It's worth pointing out, though, that unless you have electrical knowledge and can interpret the present safety regulations, an audio spur needs to be fitted by a qualified electrician.

RF Earthing

This comprises a solid and suitably deep earth stake (deep enough to stay in the ground water all year round), plated with a fairly inert metal which doesn't make a good RF detector when the groundwater corrodes it. Such stakes are hard to find. Ordinary copper-plated steel ones are not much use, as the surfaces turns to rust in a few years or months. Stainless steel stakes don't corrode, but the surface oxidation acts like a radio detector when bonded with the copper earth wire. Suitable wires (flat braids or fat conductors) provide low-inductance connections to the point being earthed, usually some critical, central part of the studio system, such as your mixer's metal casing. The RF earth wire and stake then provide a lower-impedance bypass for RF. To avoid safety hassles, a mains safety-approved capacitor is placed in line with the RF earth, at or near the point of connecting to the casing of the mixer (or other centrally placed metal-cased gear). This special earth wire and stake will then only pass RF noise, not mains faults currents.

For best results, the gear being earthed needs to be near to ground level, so this technique is not really applicable in studios that are several floors up, or where there's no access to the soil. For electrical safety and to avoid insurance hassles, the existing safety earth connections must all be left in place.

Earth Chokes

These are inductors (coils) placed in line with the green/yellow earth wire between your studio's power wiring and the mains supply. The idea is to prevent RF noise from other sources getting to the audio gear. This makes sense, since mains earth wiring, as already noted, isn't good at providing a good RF earth, but is good at spreading RF contamination. The choke has to be a type that will work well up to quite high radio frequencies, while also being made to handle high fault currents and have a low resistance to these, to maintain safe conditions in the unlikely event of a mains earth fault. It also has to be securely connected, and suitably placed where the earth wiring passes between clean equipment (like mixers) and dirty equipment (like PCs and the rest of the house wiring).

Mains-Frequency Regenerators

Some rotational devices, such as the motors in hi-fi turntables and older tape machines rely on the mains frequency for correct speed. Mains frequency variations are heard as pitch variations, usually as 'wow' (a slow speed variation) or as a sudden, one-time speed and pitch change. The mains frequency has to be stable and quite accurate for any national AC grid system to work. In addition, millions of mains transformers across the grid would also be upset (drawing high current and ultimately failing) if the mains frequency were to drop more than a few percent. When a national AC mains supply is under heavy load, the frequency starts to drop, and below a few percent droop, some of the load has to be 'shed'. This explains why wider-than-normal frequency variations are associated with power outages, and all occur most in countries with inadequate national mains capacity.

Since frequency is a national matter (in the UK it's controlled by National Grid plc) there's not a lot you can do if it is varying too much for your gear and ears. If it's outside the statutory one percent above and below 50Hz, then all you can do is take comfort in knowing that everyone in the country will also be affected. The answer is to use a 'frequency-regenerating conditioner', or turntable power supply, such as are made for hi-fi vinyl turntables.

Snake Oil

There are few accessories for mains-related problems which completely fail to work, but some are ill-designed while others are ill-sold, meaning they'll be sold to you when their best possible effect is well down the list of your setup's priorities. As the performance benefits of many mains-enhancing products can only be determined by trying them out, a reputable supplier will have to work on a sale-or-return basis.

It may be worth being wary of mains-conditioning products that are nicely presented with impressive packaging, but have absent or jumbled technical details. A recent example is a box that plugs into a spare power outlet, and merely consists of a large capacitor placed across the supply. The simplistic idea is that the capacitor 'absorbs' any noise on the mains. Equally, it provides power factor correction — but only for a very specific load condition! There are several problems caused by the brute-force of the approach. One of these is that a large capacitor across an AC supply causes voltage rise. This unexpected fact may lead to damaged equipment, especially if the voltage in your area is already high. Second, the cap will exhibit both resonances or power factors that cause overheating in isolation transformers, as well as creating illegally high 'power factors' — essentially 'borrowing' lots of energy off the mains supply without paying a higher tariff for doing so. Thirdly, a capacitor connected from the live side to the earth, can, as noted earlier, cause more noise-related problems than it cures. SOS

Email your queries, comments and tips to: sos.feedback@sospubs.co.uk
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Visit the SOS Forums at www.sound-on-sound.com/sosforum.htm

Return To Sender

Following a couple of very stressful weeks since my Digidesign Control 24 desk stopped working properly, I felt I should write in to warn other readers who may own a similar desk about what *not* to do if they ever experience a breakdown. I also need to vent some frustration!

I bought my Control 24 in November 2002 from reputable audio company 'X', and it's a great little interface which makes using my Pro Tools rig much easier. A couple of weeks ago however, I realised that the automated faders had stopped working and I called the company to ask their advice. They told me that the cause was likely to be a dodgy power supply inside the unit, a fairly common problem, and that the unit would have to be sent to their repair company. They said it was a simple problem to fix and fully covered by the warranty. As a small studio owner whose setup is based around the desk, I was worried about losing business and asked them to supply a replacement for the repair period. They said they didn't have another one, and that I would probably get the unit back in a couple of days anyway, so I accepted the situation. (Rick's Word of Warning 1: In his review of the unit in *SOS* March 2001, Paul White rightly pointed out that returning it for repair could be a potential hassle, and this is definitely something to consider when purchasing a desk if your livelihood depends on it.)

The company organised for a courier to pick up the unit and take it to their repair company, and told me to put the unit in its original box. I no longer had the original box, as it was accidentally thrown out by builders working on my house, who thought it was rubbish. (Rick's Word of Warning 2: Always keep the original box, and find a safe place to store it, even if you have to stick it in your gran's loft!) The company said that was fine, and that I should package up the unit as well as I could. I wrapped it in thick polythene sheeting, then three layers of industrial bubble wrap, and then boxed it up with a combination of cardboard and high-strength gaffer tape to form a thick box. The courier picked up the unit and that was that — or so I thought!

The next day I phoned to find out how everything was going, and was told that the repair company had received the unit but that it had been badly damaged in transit. The metal casing of the unit had been dented in several places, screws had been sheared off, and the packaging had been torn. They said it was also likely that some of the circuitry had been damaged, given the force that would have been necessary to create the dents. They

Damage to this Control 24 was "purely cosmetic".

told me I was looking at extensive repair costs that would not be covered by the warranty, and as good as called me an idiot for not sending the desk in its original box. They also told me it would take at least another two weeks to repair, which meant two more weeks' lost business.

A call to the courier was fruitless. Since I had not paid for the delivery, I could not make a claim. Another call to company X was equally disheartening — they told me they would pursue a claim with the courier, but that the courier would probably try to wriggle out of it by claiming the package was poorly protected. Even if they were successful in their claim, that particular courier pays out only £15 per kilogram on damaged goods. That would translate as about £300 compensation for a £5000 unit — not good news. (Rick's Word of Warning 3: If you do have to send a large, valuable bit of audio gear back for repair, I would always consider taking it there yourself, even if you do still have the original box.)

Next, I talked to a solicitor friend of mine who advised me that, technically speaking, company X were liable for the damage. Since they hired the courier, the damage to the desk essentially happened when it was in their care. He told me it would not be unreasonable to ask for a replacement desk. I did this, and was told in no uncertain terms that company X were *not* liable for the damage and that if I was serious about trying to force them to replace my desk then I would have to contact their solicitors directly, and prepare for an expensive and drawn-out dispute. They suggested instead that I check any insurance policies I had to see whether the desk was covered. (Rick's Word of Warning 4: Always contact your insurance company straight away to inform them of a new purchase you want to cover. This is something I had repeatedly put off. I had insurance on my other, older items but not the desk and when it came to the crunch, quite literally, I wasn't covered.) While all of these unpleasant phone calls and emails were bouncing to and fro, the repair company got back in touch and said that the desk was actually functioning OK following their replacement of the PSU, and that the damage to the desk was purely cosmetic!

I suppose I should be happy that the desk wasn't irrepairably damaged and that I'm not facing the prospect of lengthy legal proceedings. But I'm not. At the end of the day, I sent off a desk that was in pristine condition (bar a faulty PSU) and I'm getting one back that looks like it's been attacked with a sledgehammer. Sure, it works, but it's not exactly going to impress my clients. I'm also concerned that the desk might develop problems in the future, considering the battering it received. What's more, the repair company have declared that they "cannot be held responsible or accountable for any related failures that may arise subsequent to our handing over the C24 in question". There is nothing more I can do save to warn others about my experience.

Rick Parkhouse

Adventures In Stereo

I read Eric James' article on stereo microphone techniques in the March issue with interest, and have just finished a solo harpsichord recording of the Goldberg Variations using a pair of Sonodore RCM402 mics, Sonodore preamps and a Lucid 9624 A-D converter.

We were recording in a fairly reverberant country church close to Melbourne, Australia. The initial problem was reducing the mechanical noise of the instrument. After trying virtually every miking position, we ended up moving the instrument from a nicely resonant wooden floor to a carpeted one. This not only virtually eliminated the noise but also helped to focus the harpsichord's sound. We ended up using much the same A-B pair arrangement as described in the article, only with the mics spaced a little closer together.

The Sonodore mics have an excellent transient response, and are supplied with a constant voltage from their own power supply that can handle any sudden peaks. As they produce such a realistic sound, the quality of instrument, location and performance is of utmost importance. We were lucky enough to be recording a fine instrument in a nice location, and found the resulting recording had a pleasing stereo spread, with the harpsichord well centred yet with a sense of space around the instrument. Even listening to takes where you can hear a car driving past outside is a pleasure! SOS

Thomas Grubb, Mano Musica

sample shop

Scoring Pack 1

SONIC FOUNDRY
ACID

As regular users of Sonic Foundry's *Acid* will be aware, the company now have an established line of loop library 'packs', consisting of four or five individual CDs that have a common theme. The most recent releases are the *Scoring Pack 1* and *Scoring Pack 2*, the first of which is the subject of this review.

As the titles suggest, these packs are clearly aimed at those producing music for picture. *Scoring Pack 1* consists of five individual loop libraries; *Cinematix Volume 1, James Johnson Spektral Minimalism, Robert Rich Ambient Atmospheres & Rhythms, Numina Emotional Peak Sounds For Cinema* and *Orchestral Series 3: Cinematic*. In total, the pack contains 1460 loops totalling some 2588MB!

Cinematix Volume 1 ■■■

This title contains 300 loops in three sections. Music Beds provide some almost complete (if short) music pieces. Titles such as Funky Mama, Cop Show and Dark Beat give an idea of the eclectic content. The Sound Effects section also includes a wide range of styles, with some true sound effects, but mostly dark-sounding synth-based atmospheres. The Individual Instruments & Grooves section is split into a number of smaller groups covering guitar, horns, pads, strings, keyboards and drums/percussion and there is more here for music construction, rather than the textures that dominate elsewhere.

James Johnson Spektral Minimalism ■■■■■

This library comprises four sections: Complete Beds, Complex Wave, Pad FX and Sublime Keys. There are only 123 loops in total, and these are essentially a collection of long soundscapes, some running to 20 or 30 bars. These mostly provide a fairly relaxed and dream-like feel. The only real 'melody' loops are within the Sublime Keys section, with some gentle piano, vibe and bell loops. The whole CD is excellent — it just leaves you wanting more!

Robert Rich Ambient Atmospheres & Rhythms ■■■■

The 500 loops here are in seven sections and give a strong eastern feel. For example, the Clean Flutes section consists of short (and quite haunting) flute melodies mostly of an eastern origin, rather than western orchestral. The Drones & Abstract section provides mellow and soothing textures based on flute, piano and bell sounds, along with a few creepy ones. The Ethnic Percussion section offers a collection of various djembe, dumbek and frame drum loops, while the Melodic section includes various short phrases played on kalimba and marimba (amongst others), and these could easily add an ethnic touch to any 'East meets West' type arrangement. The Synthetic Pulses section is completely free of man-made vegetarian food, but does contain various processed sounds with a strong rhythmic feel, while the Metal and Twinkles sections consist of various steel and wind chime-like objects being hit.

Numina Emotional Peak Sounds For Cinema ■■■■■

As with *Spektral Minimalism*, the *Numina* set contains a smaller number of loops that are longer. Both the Beatz and Drones sections set a fairly unsettling mood — think monsters creeping up on the unsuspecting heroine and you will be in the right ball park.

This dark theme continues into the One Shot, Sound FX and Themes sections, with the latter providing some really nice textures that also contain a melodic component. The Quirps section is also dark, but the loops here contain a high-tempo rhythmic feel. This whole library is definitely aimed at those who need to suggest that something unpleasant is about to happen — put these loops up against the right footage and your audience is going to be drawn slowly closer to the edges of their seats!

Orchestral Series 3: Cinematic ■■■■

The final library is the *Cinematic* title from the *Orchestral Series* and, as this was covered in Sample Shop back in *SOS* October 2001, a quick recap is all that is required here. Essentially, the library contains 27 small orchestral construction kits with an emphasis on moody, thriller-style underscore — again, you just know something bad is going to happen!

CONCLUSION

Scoring Pack 1 is very much about dark and disturbing atmospheres. These loops would work really well under footage that requires an unsettled or tense feeling, or perhaps for dark corridor scenes in a first-person computer game. In using this collection, I found that loops from the first four libraries could be blended together quite easily, mainly because there was little in the way of chord sequences to worry about matching. What was a little more surprising was how well many of the orchestral loops could also be made to sit over some of the textures and drones.

Given the type of content, this collection is not going to appeal to many straight-ahead music producers, but for those using *Acid* as a music-to-picture tool and who want to give their audience the creeps, it has plenty to offer. *John Walden*

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Flatpack ■■■■

LAPJOCKEY
REFILL

This is the first release from Lapjockey, a group of musicians and producers which includes our very own Pro Tools Notes columnist Simon Price. Presented in Refill format, *Flatpack* contains a large selection of patches and sample sets for each of the modules in *Reason* v2 and, though largely electronic in character, it steers clear of catering for any single, specific style or genre. That said, there is a strong bias towards the analogue synth sounds of

Star Imperial Measurements

■■■■■ Rood
■■■■ Virgate
■■■ Perch
■■ Peck
■ Bushel

yesteryear, with 30 classic drum kits for *Redrum* including Roland's TR808 and TR909 drum machines and the Korg ER1. Elsewhere, there are Moog, Korg MS20, Oberheim Expander and Casio VL Tone patches (amongst others) for the *NN19* sampler. These samples are endowed with plenty of vintage character, although, in the case of some of the Casio sounds, their vintage might not be to everyone's taste.

For the superior *NNXT* sampler there are 'classic keys' sample sets of a different variety, with a selection of Rhodes, Wurlitzer, melodica and organ patches. Of these, the Rhodes comes out on top, offering a realistic and highly usable sound without retreading any of the ground covered by the Rhodes patches in the *Reason* Factory Soundbank. Five versions of the Rhodes are included: two tremolo variants, a bright version, a sustained version and the 'plain Jane' original. Although the tremolo's speed is consistent all the way across the keyboard, there's no way to alter it, but the sensible inclusion of an uneffected version allows you to add your own effects.

The Wurlitzer is, to my ears, much less successful. While the Rhodes' lower notes are pleasingly warm and fuzzy and its upper register is suitably bell-like, the Wurlitzer sounds like a Clavinet at the bottom end and like a harp at the top, giving the impression of a synthesised imitation of the real deal.

There's a nice Lorenzo accordion patch and a couple of brash melodicas which aim to accurately replicate the original instrument's tuning instabilities, although, in my opinion, these stray too far towards reality! There are some pleasantly full and fruity organ sounds for the *NNXT* too, although my favourite organ presets in the Refill are for the *Malström* module.

In general, *Flatpack*'s *Malström* patches make excellent use of the module's two oscillators and oscillator presets, giving a good demonstration of its capabilities. Helpfully, two template Songs included with the Refill demonstrate how *Malström*'s modulators, filters and shaper can be used to create radical effects on external sources (*Flatpack* features five such effects patches) and a third shows how the *Subtractor*'s electronic drum sounds can be triggered from Redrum using Gate signals. There are around 100 patches each for both *Subtractor* and *Malström*, ranging from basses, synths and leads to some evolving loops and pads. There are some interesting pads for *NNXT* too, and the folder labelled Pads (One Finger Chord) does exactly what it says on the tin, for that instant ambient house breakdown. The audio files used to create these soundscapes are included unpacked in a separate folder in WAV format.

Among more than 200 REX-format loops there are a number of acoustic percussion samples which provide some contrast to the electro slant of the Refill's other rhythmic content, though here too you'll find plenty of bleeping retro quirkiness. All in all, *Flatpack* impresses by virtue of the sheer quantity of material on offer and the creativity that has clearly gone into its assembly. If you can't find at least something here to tickle your imagination, then you'll have a hard time finding it anywhere else, and for £50 in the UK you could do a lot worse.
David Greeves

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Sonic-O-Tool 2

AMG
REFILL

Considering the kinds of cutting-edge dance music styles that are usually associated with *Reason*, the idea of a General MIDI Refill might appear to be a bit of a strange one. However, Propellerhead have already demonstrated that *Reason* is much more than a one-trick pony with the *Orkester* Refill. If *Reason* is your main music-making environment and your studio is not overly stocked with top-notch soundcards or hardware synths, then some facility for playing back basic MIDI files (commercial ones or those acquired via the Internet), might also be quite useful.

The CD is split into three parts. The central feature is the main set of General MIDI sounds for the *NNXT* sampler. The rest of the Refill contains some 200

multisampled, multi-layered *NNXT* instruments that are variations on the core GM sounds. These are also supplied in less memory-intensive *NN19* forms (370 patches). A collection of 26 *Redrum* kits rounds off the Refill. Finally, some 2900 samples, upon which the Refill is based, are supplied in WAV format. All the original samples were recorded at 24-bit resolution and the synths used include the Korg 01W, Waldorf Microwave, Yamaha DX7, Roland TR808 and Linn 9000, amongst others.

The *NNXT* GM sound set offers 128 patches named and numbered according to the GM convention. These stick fairly faithfully to the sorts of GM sounds found on many sound modules — so you can expect that the *NNXT* Soundtrack patch to be very much like that in other GM sets. This is important for those who might use this Refill as a means of playing commercial MIDI files written with GM in mind. In the main, this works well enough in practice, and AMG include a few template Song files to help this process. However, it is still something of a pain that, having opened a GM Standard MIDI File, you then have to manually assign the appropriate patches to each channel.

The quality of the GM sounds themselves is pretty good. Most feature multisamples and some also multi-layers. The quality is certainly better than you would find on most soundcards and, for example, I preferred many of them to the equivalent XG sounds on my own Yamaha SW1000XG — perhaps just a little less polite? In particular, the pianos, brass, synth bass and pads all worked well. For the latter, adding in the sound-shaping capabilities of the *NNXT* offered some very functional sounds.

One or two things were less satisfactory. For example, a couple of the guitars were a little lumpy (but then really good GM guitar patches are rarer than rocking horse manure) and the solo violin is a little heavy on the vibrato for my taste (although the fiddle is useable enough). Melo Drum is straight out of Eastenders!

The additional *NNXT* patches (and their *NN19* equivalents) are all in GM territory, but add to the variety on offer. For example, the Arp/Sequencer section contains a small number of really nice patches that sync to tempo. The usual GM-type sound effects are also present, but are far more convincing here than on your average soundcard. The *Redrum* kits cover the usual TR808, TR909, funk, acid and 'straight' collections, plus one or two that are more adventurous (the Daisy Kit is quite interesting), and the quality is good.

In conclusion, *Sonic-O-Tool 2* is perhaps not going to set the Refill world alight, but those needing access to a very useable GM-style sound set via *Reason* need look no further. What the sounds might lack in originality, the CD makes up for in value for money — £30 is a steal. Now where did I put that 'Baby One More Time' MIDI file? *John Walden*

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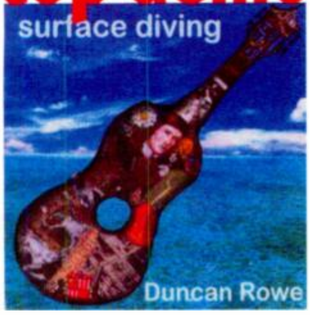
demo doctor

The main demos featured this month can be heard in both Real Audio and MP3 format at www.sound-on-sound.com/sos/july03/demodoc.htm

Resident specialist John Harris offers his demo diagnosis and prescribes an appropriate remedy.

top demo



Duncan Rowe

Equipment: **Apple Mac G4 running Steinberg *Cubase* v4, Adaptec *Toast*, Mackie 24:8 desk, Tannoy DC100 monitors, Alesis ADAT XT eight-track recorders (pair), Alesis 3630 and Joemeek compressors, Roland RSP550, Lexicon MPX1 and Alesis Quadraverb effects, Rode NT1 and NT2 mics, Tascam DA30 DAT.**

This project began as a recording of acoustic guitar and voice, and then grew as other instruments were added. Predictably, there were difficulties with overdubs as the tempo of the original guitar and vocal tracks has a tendency to drift. This is discernable in the way some of the earlier songs on the CD are mixed — with percussion and bass kept fairly low in volume. Click tracks were toyed with but abandoned because of Duncan's inexperience at playing to them. Some songs don't lend themselves well to being recorded with a click track in any case. Later on, the guitar, vocal and percussion were all recorded at the same time. It's a shame that this tactic wasn't used earlier in the making of the CD.

The better mixes sound fine in stereo, but in mono they sound quite muddy and, at times, there's some obvious phasing going on. Some of this could be blamed on the modulation effects being used. These effects often employ some kind of stereo phasing trick in order to artificially widen the stereo image, such as anti-phase feedback on one side and in-phase feedback on the other. Another possibility is that phasing is being introduced at the point where outboard compression is applied to the mix. To eliminate this possibility, the cables running to and from the compressor should be checked for phase compatibility. It's also worth checking the cables at other points in the signal path (such as the DAT machine) just in case.

In fact, the compressor was reported to be a little unpredictable by engineer Chris Bond, working sweetly on some mixes, but applying too much compression to others, even at what should have been low compression settings. There could be a couple of reasons for this. Firstly, some of the mixes could contain more transient peaks than others which would certainly cause the compressor to work harder. I suggest they check the gain reduction meter as the mix is playing; at the point where it's hitting its highest levels, they should to try to identify the sound or sounds that are driving the compressor hardest. It may then be advantageous to reduce the level of these sounds at certain points, which, of course, can be done with the mixer automation features in *Cubase*. If there are any suitable compressor plug-ins available, another option is to apply individual compression to the offending sound's mixer channel, in addition to the compression being used across the entire mix. Another possible reason why the compressor is working too hard at low settings is a signal level mismatch between stereo output and compression input. If the stereo output signal is at +4dB and the compressor is set to receive a -10dB input, this could be contributing to the problem.

With all this technical advice, I probably haven't said as much about the music as Chris and Duncan would like. In a nutshell, there are some fine songs here, showing good use of effects (the rotary modulation on the guitar and keyboards, for example) and I particularly liked the way the female backing vocal is arranged and performed. The album's title track, 'Surface Diving', is probably the best-produced song on the CD, but it cries out for real drums — perhaps it would have been worth the trouble of trying to record a real drummer. ■

Overdubbing and over-EQ'ing

One of the hardest aspects of learning your trade as a sound engineer has got to be mastering the art of overdubbing sounds and making them gel with the rest of the mix. It's a skill that can only be acquired through practice and there are no real short cuts. Yet some things immediately become obvious if you are able to take a step back from the track and give it an objective listen. One is that you simply can't record a lot of energetic or complex sounds that occupy the same frequency area without running into difficulties. Many of the recordings this month seem to be over-equalised in the upper-mid region. This is often an attempt to give the instrumentation clarity by the means of EQ boost. But EQ can only do so much and, rather than over-EQ'ing, some different and more complementary sounds should have been chosen.

Cosmic Puppet

Venue: **Home**
Equipment: **Apple Mac G4 with Emagic *Logic Gold* v5, *EXS24* sampler and *Waveburner Pro*, BIAS *Peak*, TL Audio 5021 compressor.**

The opening track is tight and punchy thanks to some good programming, and the deliberate use of heavy quantisation gives the music a mechanical, quasiindustrial feel. Indeed, this is very much in line with the general concept of the CD, which is based around a novel by science fiction author Philip K Dick.

Track two of this entirely instrumental CD uses simple bass lines and loops, leaving enough space for the rhythmic echo effects on these sounds to come through. These effects create more activity and interest in the stereo image, and when the original sound is being filter modulated, the delay follows the resulting changes in tone.

Having said that, all too often the delay is set up and then just left to run throughout the course of a track. I wonder how effective it might be if the delay didn't follow the filtered sound, but was instead applied to a copy of the original that was unaffected by filter changes. This copy would have to be inaudible in all but its delayed form. This could be achieved by applying a delay plug-in to the copy with the Mix control set to 100 percent, or 'Wet'. Another idea would be to use a longer delay at key points in the arrangement, where space allows. This could even be set up so that the repeat occurs a whole beat or more after the original filtered sound. Needless to say, the part would have to be musically fairly basic for this to work successfully, but it could be used very effectively at certain points in the arrangement.

This is by no means a CD of warm-sounding mixes, but neither does it confuse the term 'industrial' with an abundance of harsh upper-mid EQ boost and distortion. Tension is created by the uptight, heavily quantised rhythm track, occasional slow attack sounds and repeated metallic notes, and there's some excellent use of detuned oscillators, in a manner reminiscent of a sci-fi movie soundtrack. SOS

How To Submit Your Demo

Demos should be sent on CD, DAT, Minidisc or Cassette to: Demo Doctor, *Sound On Sound*, Media House, Trafalgar Way, Bar Hill, Cambridge, CB3 8SQ, UK. Please enclose a band/artist photograph and/or demo artwork (which we may use here and on our web site to illustrate your demo review). Including contact information, such as a telephone number, web site URL or email address, will enable anyone who is interested in your material to contact you.

QUICKIES

Fret

It's clear from this band's name and sleeve artwork that this is a guitar-based project, and this CD album features some exceptionally good playing. There's some interesting use of electric guitar textures, in particular the creamy-sounding harmony lead guitar parts played over chords. Elsewhere the arrangements are tight and inventive, with just a touch of jazz rock creeping in to extend the scope of the CD beyond familiar classic rock territory. However, the mixes are generally bass light and it feels like there's a lack of confidence in the bass end of these recordings. This may well be down to poor monitor placement, as the monitors — Mackie HR824s and Yamaha NS10s — should be fine for the job. However, the equipment list includes enough post-production software to fix the problem without having to remix the album. I'd use a graphic EQ to bring up the bass at around 100Hz and I'd also cut a little mid range at around 800Hz where there are too many instruments vying for the same space. In combination with a maximiser or some limiting, this should bring up the overall level and give these tracks a sound as professional as the cover artwork! ■

Martin Rigby

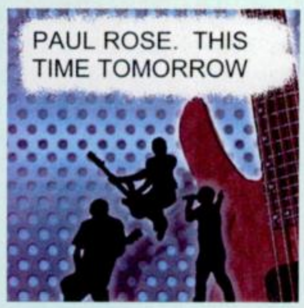
Unusually for a guitar-based album, the instrument itself is pretty understated here unless playing an obvious solo, so much so that I can't easily pick out the rhythm guitar on either of the opening mixes! Despite this, the sequenced arrangements are very well programmed and the lead guitar work, although highly derivative of artists like Santana, is excellent. Overall, I felt that the mixes placed too much emphasis on the 800Hz-2kHz region, with clean guitar, bright keyboards and equalised drums all vying for space. Some more attention to sound layering is required, and a more mellow keyboard sound would allow the guitar to come forward in the mix without having to lift it in level. Despite this album's emphasis on guitars, there are some skilful solos from other instruments too, like the piano on track four, and there's a nice Latino feel to the drum and percussion programming which keeps the foot tapping. ■

Ben Mayer

Ben's written some good songs and performs the vocals well, getting a good sound from his Rode NT2 and Joemeek mic channel. The mixes are consistently good as far as the drums and voice are concerned, but the other instrumentation is just too low in the mix. The picture on the CD cover of Ben in his studio may offer some explanation as to why this should be. I couldn't help noticing that the mixing area is surrounded by flat, hard surfaces with no acoustic treatment. Naturally this would give the engineer a false impression of the mid-, and especially upper-mid frequencies in the listening area by over-emphasising them. However this doesn't explain the generally bright sound of the mix, unless Ben is trying to copy certain commercial albums in his collection. I'd suggest a little acoustic treatment (a couple of HF acoustic tiles would probably do the trick) and a remix. ■

Paul Rose

Paul's CD is stylistically pretty diverse which makes me wonder who it's aimed at, or whether he's just recorded it for his own satisfaction. With no continuity in terms of musical style (even though it has a rock-oriented cover), it can't be pigeonholed, but neither is it that saleable on the open market. Image aside, he's done a decent job with a fairly basic setup. The AKG C3000 microphone is probably his most expensive bit of kit and it helps give the vocals a suitably classy sound. Like many of the other demos this month, there seems to be a tendency towards overly bright-sounding mixes, although Paul does manage to get a solid bass-end going on his, with a good balance between the punchy kick drum and bass synth sounds. I was a little surprised at how thin some of the synth sounds were given the sound sources — a Roland MC303 and a Korg Poly 800 recorded to analogue four-track — so this must be the result of over-equalisation in the upper-mid region. Some of the arrangements are a little self-indulgent and don't warrant their long running times, offering little in the way of development, and relying on the same bass groove throughout. At the very least, some extreme effects like comb filtering and delay here and there would help to vary the dynamic. ■



The main demos featured this month can be heard in MP3 format at www.sound-on-sound.com/sos/jul03/demodoc.htm

Business End enables you to have your demo reviewed by a panel of producers, songwriters, musicians and managers. If you want your demo to be heard by them, please mark it 'Business End'. This month's industry panel is drawn from the MPG (Music Producer's Guild).

The Audio Journal

Matt Ward (MW): "These songs have a nice groove and the band definitely have an idea about song structure in terms of how they are varying the tempos and sounds, although I'm not sure if they are quite pulling it off because at times it's all over the place. Sometimes it goes a little bit country, other places it's a little darker, all of which could develop into their own sound, but they need to work on that.

"The vocal ideas are there but they are a bit limp in relation to the backing so I think the vocals need more power. The packaging is good and looks it professional already."

Jon-Paul Harper (JP): "The second track didn't do much for me, it was as if it had been thrown in there to have something obviously different from the first. The third was more in keeping with the first."

Dave Fowler (DF): "It's difficult to know where they are coming from when they list their influences as being everything from Nine Inch Nails to Mike Oldfield, which are at completely different ends of the spectrum. The problem is that there are too few clear harmonies, and no distinctive melody to make it catchy. I haven't come away singing any of the riffs or the vocal tunes.

The demo starts off with a very Beatlesque 'Come Together'-type sound but it doesn't have anything like the sing-along melodies of the Beatles. Even though it is well done and they have well-crafted songs, every song needs something catchy.

"The vocalist uses a low range on one track and a very different high range on another and I thought it was two different vocalists until I read the letter, so if he wants to work on the vocal melodies, he has the skills to do it."

Sam Shemtob (SS): "I was interested by the cover straight away, it is artistically done and it made me want to listen a bit harder. Songs two and three caught my ear because they have some interesting emotion that wasn't completely obvious and simple. Having said that I wasn't bowled over by either of them."

JP: "I don't like spiders and so out of all the CDs it was their cover that drew my attention immediately. You have to give them credit for that because getting noticed is the name of the game."

DF: "I must say that I am still of the opinion that for demos, the more plain and factual the cover the better. I like to see just a band name, a list of songs, the names and roles of the people in the group and a contact number. Artwork just makes me expect to hear something that is not there or stops me wanting to listen to the CD."

JP: "I disagree with that, purely because a demo arrived here the other day with marker pen scribbled on the cover saying 'sorry about the cover but my printer is bust', and that is remarkably bad. I *do* think that the first bite is with the eye."

DF: "When I say simple I don't mean badly presented, I imagine just a very clean piece of white card with all the information printed in a nice black typeface."

Izernega

JP: "Keeping it simple is good if you have a strong idea, but I found this very repetitive and monotonous, because there isn't much going on and not much of a riff to latch onto. Using a simple rhythm when the main focus of the track is the words is understandable, but I found it hard to make out the words, so it's a bit of a no-score draw for me. As the words are quite important in this case, I would like to have had the words written out with the letter.

"The one thing that was interesting and unusual in this context was the Eastern instrument sample — I think it was a balalaika — they were using in places."

MW: "By its very nature, hip-hop is stripped-back, has a very repetitive beat and a certain aggression to its vocals, so to call it down for being simple would be unfair, but when something is stripped-back it has to be extremely good to stand out.

"It was quite a nice idea to try to blend the world music elements with hip-hop, and it is definitely something they should explore, but they haven't managed to do it successfully yet — it is still a bit like hip-hop by numbers. This demo isn't really keeping up with the developments that have happened recently in terms of hip-hop production, groove and style. For example, this kind of simple riff-over-a-beat production is like something the Wu-Tang Clan would have done, whereas new producers like Timbaland have messed around with the beat structures so that they are not just a plodding 4/4.

"The cover is very nice and they have some strong ideas, but they need to lift the vocal performances and put something more in there because at the moment it doesn't draw you in, and there are any number of hip-hop MCs who rap in that slow drawling style."

SS: "It would have been interesting to read a few lines of their story — maybe just a paragraph — and there is space on the letter for it. The CD-R broke down at one point in the second song, and some cheap CD-Rs are prone to this, but if I was making a demo I would make sure I had tested it thoroughly before sending it out. I also thought that the third track sounded really tinny and trebly.

"In terms of sound the balalaika is intriguing, and I agree that there are some nice ideas there, but for the genre the arrangements still need more depth. Having said that, for me that's more of a polish thing and basically I think they are quite interesting.

"I got the feeling that some of the songs don't work that well on CD but they might be quite interesting live. I'd like to know if these guys can get a crowd behind them because I think the live element could be important."

DF: "There is a lack of dynamics in the music. It doesn't build up to a point and then break down to another point; even lyrically it is sung with the same rhythm and tempo over and over again. Regardless of the style of music a song needs dynamics, otherwise there is nowhere to go, and all three songs are very much the same all the way through.

"The presentation is nice and plain but the covering letter has very few details so I am left with no idea about these guys. Curiously it did have a line saying 'featuring Killabeat', but there is no mention of who this guy is or what he does."

Pho Aureel 3.2

MW: "The first thing to say is that he's written that he wants to collaborate and work in films and he doesn't want to release CDs at all, so we have to consider this from a slightly different perspective to a standard demo. The presentation, in general, is very nice, smart and clean, but for film music he should include some idea of the mood. Generally speaking, directors expect some sort of reference to the situation it should be used in because they don't want to sit through an eight-minute track to pick out one particular snippet. He could include some kind of sync'ed movie on the CD so we can see how it works in comparison with the visuals.

"The music is very clever and atmospheric. He shows occasional flashes of real talent, particularly in the third track where it has the feel of Lemon Jelly, but without any visuals I am unsure of how it is supposed to work.

"The first track takes ages to come in and then it doesn't do very much, and that is a bad move; he should swap it. I appreciate that this is for a visual medium but the tracks should still be cut down."

DF: "Given a little more work I would probably buy this, I can imagine playing it at home while I'm busy. There are some very catchy bits in the first and third tracks: the guitar line on the third piece is instantly catchy, as is the two-note keyboard idea on the first. He's used two notes where others would use hundreds.

"This demo has dynamics, ideas, and most interesting of all, for a bloke who writes 'I am not a trained musician, probably not really a musician at all,' he has shown more musicianship than either of the previous two submissions. He is using his lack of training to throw sounds in to see what happens. As a producer I am swayed by his interesting use of effects but setting the technical stuff aside, this is the only demo of this three I would have happily listened to all the way through without skipping, although I suggest that the tracks could be divided into sections on the CD, like movements.

"He says that he has had one piece on TV, so he is close to achieving his goals, but if he did decide to change approach, then he could bear in mind that William Orbit did well with his *Pieces In A Modern Style*, which isn't dissimilar in its feel. Also Aqualung have done similar spacey mellow music but with added vocals, so I can imagine he could go that way too, with a decent singer."

SS: "I also really liked track one and I noticed that we all sat up when the guitar in the third track began. I like the fact that all the tunes have lots of live instruments even if they are, perhaps, off a CD or put through a processor. In the first tune I was thinking of DJ Shadow, which has to be a compliment, but I was also wondering if it was going anywhere."

JP: "I remember seeing an interview with Mick Jones when he had just started Big Audio Dynamite, and he was explaining that he had a keyboardist who wasn't really a musician so he had coloured dots on all his keys to help him play. Mick thought that there was a certain quality gained by having somebody in a band who didn't know how to hit something properly, and that has come out in aces on this demo. It has an atmosphere, and it's doing interesting things even without words.

"It has a great contrast between the use of contemporary plug-ins and real instruments. He has a vibe and a feel and not many people seem to approach music that way these days. For example, classical composers were painting a picture with sound or creating a mood, whereas a lot contemporary tracks just grab a riff and strangle it for five minutes! This did sound like a backing track to something on TV and it was almost like painting a picture with music, in that respect. I feel that I have heard something different, which is very refreshing, and that's what music should be about.

"If there was any criticism then I would say he should keep the track lengths down and make sure there is a little more contrast. In places it builds up and then drops away. Sometimes an anti-climax can be just as effective but I think he could try bringing in some vocals or samples so there are some sort of landmarks." SOS

This Month's MPG Panel

Dave Fowler is the Studio Manager and owner of Rogue Studios in South East London. At Rogue, Dave acts as producer and engineer for bands and artists using the venue. He is also the singer and guitarist of the band Juice who have recently been touring Europe playing at festivals alongside Feeder and Coldplay. Dave currently runs the small band promotion company called Bandnet (www.bandnet.co.uk) which organises gigs in some of London's most respected venues.

Matt Ward worked for the MCPS as a music licensing consultant for several years before moving to Kickin Music in London, where he heads their publishing arm, Haripa Music Publishing. Kickin/Haripa's catalogue and writers include Chiller Twist, Kings of Tomorrow, Deep Dish, Decoder, Blaze, Future Disciple, Lemon D, Carl Faure and the Early Dance Masters.

Sam Shemtob is press officer for the Association of Independent Music (AIM), the trade body for UK independent record labels. He does other press work, under the Name moniker, for Musicindie, Recordstore and Musictank, and has recently begun a venture to license music for new media uses. He'd like to be in a position to send in a demo of his own one day.

Jon-Paul Harper was the owner and principle engineer of Rogue Studios for five years. Dave now runs Rogue's rehearsal studios but continues working as a engineer. His other experience includes working as a roadie, seven years engineering front-of-house and ten years playing guitar in a touring band around Europe.
Jon-Paul has been both a record label owner and an artist signed to an independent label. He has a home studio running Emagic *Logic Audio Platinum*.

Many thanks to Rogue Studios (www.roguestudios.co.uk) who hosted the session.
The MPG's web site is at www.mpg.org.uk

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Spec: VSTi for all current Windows and Mac platforms. Audio Units. Price: £39.95

DR-008's official moniker is an Advanced Rhythm Production Station for both VST 2.0 and DXi hosts. The most advanced VST 2.0 and DXi drum module on the market offering an astounding array of features and including GMEDIA Music's custom built library of kits and MIDI gooves. "This is an impressive drum synth in one very easy to use package... it emerges as a fantastic long term proposition." Computer Music Performance Award May 2002

Spec: VSTi and DXi for all current Windows platforms. Price: £99.00

The Oddity is based upon the classic Arp Odyssey, a faithful recreation offering some breathtaking and innovative enhancements to the original's feature list. This is a stunning, unique instrument at an impulse price. Tune-in, turn on and be amazed! "I have to say I unreservedly love the Oddity... If you were an Odyssey fan you have to have this. And if you have never tried an Odyssey then treat yourself." Future Music, March 2003

Spec: VSTi for all curent Windows and Mac platforms. RTAS, MAS and Audio Unit versions. Price: £99.00

This VST to AudioUnit Adapter for Mac OS X is a highly-integrated plug-in wrapper that allows users of AudioUnit compatible applications such as Emagic's Logic Audio 6.x and MOTU's Digital Performer 4.x to integrate VST plug-ins and instruments seamlessly in to their working environment. Once installed and configured, the wrapper creates "virtual AudioUnits" for each of your VST plug-ins, which behave exactly the same as true AudioUnits in every respect. It uses virtually no CPU (less than 0.1% per instance on a 550MHz PowerBook G4), and offers total compatibility with a huge selection of VST plug-ins from a wide range of vendors. It's also incredibly easy to use!

Spec: Software for Mac OSX. Price: £49.95

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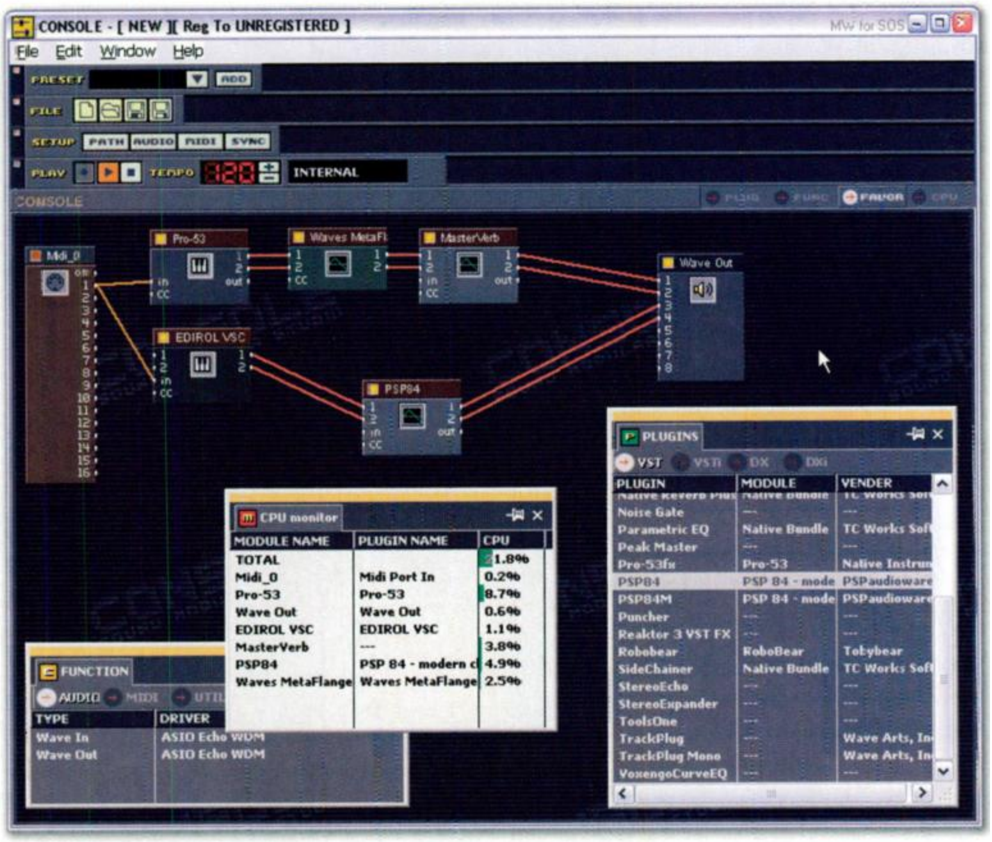


Fancy running a second computer as a stand-alone instrument or real-time effects rack? *Console* can serve as a host application for your existing instrument and effects plug-ins, in addition to acting as a wrapper inside other hosts.

This month we offer a few suggestions about using your old computers as stand-alone effects and and instrument racks, and look at a utility to remap incoming MIDI velocities.

Martin Walker

With technology continuing to advance all the time, many users will be tempted to replace their computer every couple of years with a newer and faster model. In some situations, you can keep some of your components during the upgrade to save money; but with so many new varieties of RAM, faster hard drives, cases, power supplies and so on, it's often more tempting or sensible to replace the whole lot.

In this situation, you may find yourself with an old (but fully working) second PC that almost no-one wants, and so long as it has a processor of around 800MHz or faster, along with 128MB or more of RAM and a 10GB or larger hard drive, you could consider using it to run additional software effects and instruments alongside your main machine. There are various ways to connect the two machines, ranging from a simple MIDI lead if both have a suitable interface, to a variety of LAN-based solutions, such as Musiclab's *MIDI Over LAN* (www.musiclab.com) and Helmut Eberhart's freeware *MIDI Via Net* (the author's web page at http://24.3.46.20/software.htm doesn't seen to be active, but his software is available at various other sites). I've even heard of such utilities being used via a USB network cable, without involving extra hardware.

The Virtual Rack

Unless you're going to run a stand-alone software instrument or sampler like *Gigastudio* on your second computer, you'll need a suitable host application to run your soft synths. Of course, you could run another copy of your favourite MIDI + Audio sequencer on the second PC. Some non-dongle-protected programs will allow you to install a second version without problems, for example, while users of *Cubase SX* who've upgraded from *Cubase VST* may still have their old *VST* dongles. However, this can be an expensive option if you need to buy a second copy of a fully fledged sequencer, and you'll be wasting many of its resources as well. A cheaper solution might be one of the various entry-level versions of popular MIDI + Audio sequencers, such as *Cubasis* for instance, but perhaps the best option is a simple host application that supports both plug-in instruments and effects.

A good approach for anyone

PC Snippets

Back in *SOS* June 2002 I mentioned Ian Lyon's *Dspctrl* software for the Yamaha DSP Factory soundcard. Apparently, many users have previously been unable to use the card fully until they downloaded this utility, and many have also suggested further improvements which now appear in the latest version 2.13. These aim to make the mixer software faster and more intuitive to use, and Ian hopes that *Dspctrl* will become an open source project with other contributors in the near future. A free demo is available, and the full version only costs £22.
W www.xpi.btinternet.co.uk

Anyone who uses a convolving reverb such as Sonic Foundry's *Acoustic Mirror*, *Samplitude*'s *Room Simulator*, or even the freeware *SIR* (http://home.t-online.de/home/520073787260-0001/sir076b.zip), will be pleased to hear about two web sites where you can download further impulse responses. Noisevault (www.noisevault.com) had 79 impulses when I last visited, covering many of the most popular manufacturers, including EMT, Eventide, Lexicon, and TC Electronic, along with a wide variety of real acoustic spaces, mics and preamps. Echochamber (www.memi.de/echochamber/responses) also offers a good range of impulse responses, including some created by effects plug-ins from Arboretum and Waves.

Have you ever come across two files with identical names and sizes, but not been sure whether they are identical and if one can be deleted? I've used the DOS File Compare command in the past, but entering the full pathname of each file in its command line can be tedious to say the least. However, a far easier approach is to use Trombettworks' *File Comparer*, from the same developer as the velocity curve utility mentioned in the main text, where you can simply drag the two files in question into a window and click the Compare button.
W www.trombettworks.com

using *Cubase SX*, *Nuendo* 1.6 or *VST* 5.2, and who already has a soundcard in each PC with digital I/O, is Steinberg's proprietary VST System Link. For those who just want to run a virtual rack on another machine, Steinberg have developed *V-Stack*, a virtual rack for up to 16 VST Instruments that locks to the main application with sample accuracy. *V-Stack* is available to download from Steinberg's web site for 50 Euros.

However, there are various other VST host applications out there, and the K-v-R web site provides a comprehensive selection at www.kvr-vst.com/host.php. FXpansion's freely downloadable *Simple VST Host* (www.fxpansion.com/skunk/svh14.zip) supports up to 16 VST Instruments and VST effects, arranged as four channel strips with four insert points, and is compatible with MME or ASIO drivers. For $40, Spin Audio's *ASIO FX Processor Standard Edition* (www.spinaudio.com/products_asiofxpse.html) also has four chained slots for VST plug-ins, one of which can be loaded with a VST Instrument.

I've also recently discovered *Console*, a host that not only provides an attractive 'drag and drop' graphic interface, similar to *Reaktor* and *Tassman*, but also supports VST and Direct X plug-ins, and can use MME, Direct Sound or ASIO 2 drivers. *Console*'s support for Direct X plug-ins gives it a much wider appeal, especially to *Sonar* users, and in addition to being able to run as a stand-alone host, *Console* can also be run as a plug-in from within other host applications to add Direct X support to a VST-only host and *vice versa*. *Console* is a $54 shareware product from Art Teknica (www.console.jp/eng), and it's very flexible, as you can see from the screen shot (left). After just a couple of minutes' use I managed to create a setup that ran the *Pro 53* VST Instrument with Edirol's *Virtual Sound Canvas* DXi, along with a selection of VST and Direct X effects plug-ins, and with no great overheads in terms of resources, as you can see from the CPU monitor in the screen grab.

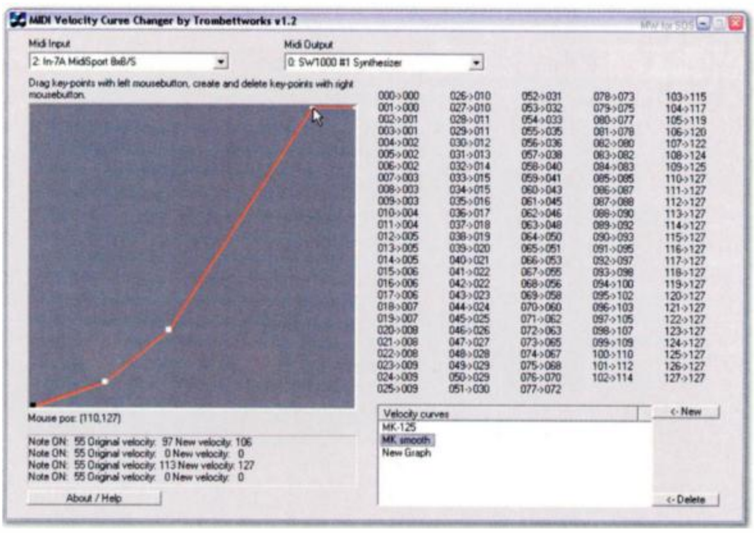
Changing Curves

Have you ever wondered if your master keyboard is letting you extract the maximum range of expression from your various hardware and software synths? Well, depending on your keyboard style, you may not be hitting it hard enough to generate the maximum MIDI velocity level of 127 (and some keyboards rarely output higher velocities than 110 even if you hit them with a sledgehammer), while very soft playing may still be outputting velocities in the 40 to 50 range, leaving your MIDI performances restricted to a range of 40 to 110 instead of 1 to 127.

Some keyboards already offer a selection of velocity curves to suit your playing style, but even if yours doesn't, it's nearly always possible to do this on your computer instead. Suitable real-time functions sometimes appear in sequencers, but are often limited in scope or difficult to set up, like the *Cubase SX* MIDI Input Transformer, which would need strings of logical command lines. Both *Cubase SX* and *Sonar* users already have access to real-time MIDI effects for velocity control, but these only affect playback, and not the recorded performance.

The answer is to use a separate MIDI utility, and since this can process all your incoming MIDI performances for maximum expressive possibilities, it's well worth taking the trouble to optimise it once and for all. A good one that I came across recently is Trombettworks' *MIDI Velocity Curve Changer* (www.trombettworks.com), a freeware utility with a graphic interface that allows you to create your own 'transfer characteristic' by drawing in extra points on the curve and dragging them where needed. By monitoring the velocity of an incoming MIDI performance, you can determine the useful range of your keyboard decide how to enhance it. My own keyboard, for instance, 'topped out' at about 110, so I compensated by creating a new point (indicated by the cursor in the screenshot) to expand the input range from 0 to 110 to generate an output from 0 to 127. Most of my quietly played notes ended up at velocities between 40 and 50, so I also drew in a couple of extra points, ending up with a curve that generates a typical performance range of 20 to 127.

The only complication of Trombettworks' very useful utility is that you need an additional utility to pipe its output into your sequencer as a MIDI input. Most PC musicians will already be familiar with *Hubi's Loopback* for use with Windows 9x, and *Midi Yoke* for Windows 9x/NT4/2000/XP, although, as I mentioned last month, Windows XP and *Gigastudio* users can have problems with this combination. However, once you've tweaked the curve to suit your keyboard and playing style, it can be a revelation to switch between the original and optimised curves and hear what you've been missing.



Trombettworks' *MIDI Velocity Curve Changer* lets you modify the velocity of your incoming MIDI performances.

A Case Of Mistaken Heatsinks

I've seen a lot of comments about the advantages of using aluminium cases, and for me, the three advantages of my Lian-Li PC60 are its good looks, light weight, and the ease in which you can install new drives and expansion cards using its modular bays and chunky thumbscrews. However, despite many claims to the contrary (including my own mistake way back in *SOS* September 2001), it's generally not true that aluminium cases can act as a huge heatsink.

While it is true that plenty of heatsinks are made from aluminium, and there are probably thicker panels in an all-aluminium case than a similarly sized steel and plastic construction, a sheet of metal can only dissipate heat effectively if it's bolted to the hot object. So, in my own system, the cooling of the Silent Drive sleeves and the PSU casing may be subtly enhanced by being bolted to all that aluminium, but all the other components, including the CPU heatsink, any other motherboard chipset heatsinks and cooling arrangements for expansion cards, will be totally unaffected. So, in conclusion, an aluminium case is probably no more effective at cooling a PC than any other type — it's primarily the combination of heatsinks, vents, airflow, and cooling fans that does this.

It's history in OS X, but extension management is a subject every OS 9 user has to deal with. This month, we offer some final advice on the subject and see if tweaking extensions really can boost your system's performance.

Mike Watkinson

Extensions are additions to the core system designed to add functionality and enable and enhance communication with both built-in and third party hardware. Many parts of Mac OS 9 are supplied by Apple as extensions, such as QuickTime, for example, and third-party hardware and software manufacturers provide other extensions to add the necessary functionality for their products. As an example, to use a Lacie Firewire drive, you simply plug it in, and the Mac OS Firewire extension will allow the drive to mount and be formatted with Apple's disc drivers, which wouldn't happen if the extension was absent. Installing the Lacie software adds a Lacie Firewire support extension to augment Apple's own Firewire extension, providing communication between the Lacie disk utility, *Silver Lining Pro*, and any Firewire discs.

Catching Conflicts

Although extensions don't make permanent changes to the operating system stored on your hard disk, they can cause conflicts with the operating system and with each other when loaded into memory by the OS at start up. Even if you think you've got your extensions co-existing amicably, it's quite possible for an upgrade to either the operating system (if you're not already running the last version of Mac OS 9) or the extension to cause a conflict. Obviously, the default set of extensions supplied with any machine is thoroughly tested before Apple release it, but as soon as you add third-party extensions, or an Apple extension is updated, there's the potential for a conflict.

Apple supply a utility for organising the list of enabled extensions called Extensions Manager, which can be found in the list of Control Panels in the Apple Menu. With a freshly installed system, Extensions Manager will list three sets of extensions under the Selected Set pop-up menu: 'Mac OS All', 'Mac OS Base' and 'My Settings', the name of the unlocked set that's active by default.

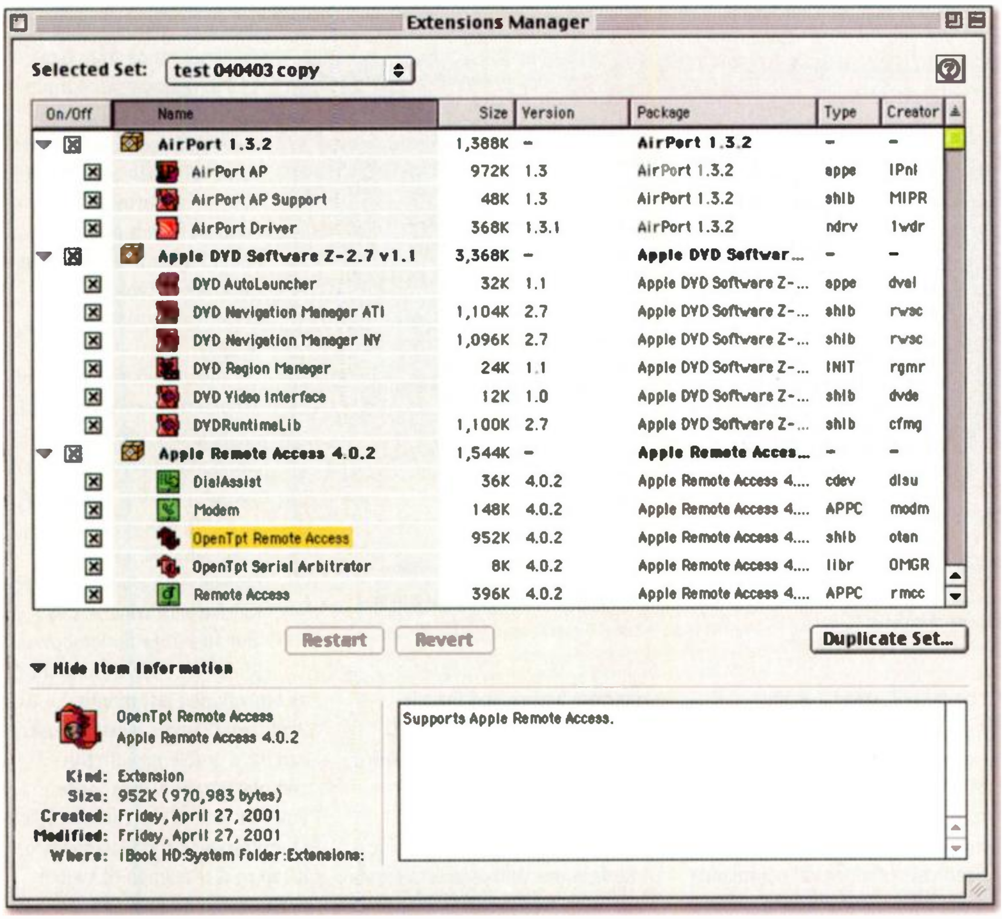
If you had the time and patience, a good habit to adopt would be to duplicate the 'My Settings' extension set before each installation, and rename it with the name of the software or hardware you're about to install. This way, every addition to the active extensions list can be tracked, and getting back to a version of the loaded OS plus extensions that operates without the symptoms of extension conflict would be a simple case of choosing the right extension set.

In the real world, however, installations occur without this zealous housekeeping. An extension conflict (which could be between hardware and extension, extension and extension, or extension and OS) could reveal itself gradually, as an emerging pattern of 'soft' crashes (requiring Command/Apple-Alt/Option-Escape-to force-quit the application) or full freezes of the computer, requiring Control-Command/Apple-Power button to restart (newer machines will power down if you hold the power button in for about five seconds).

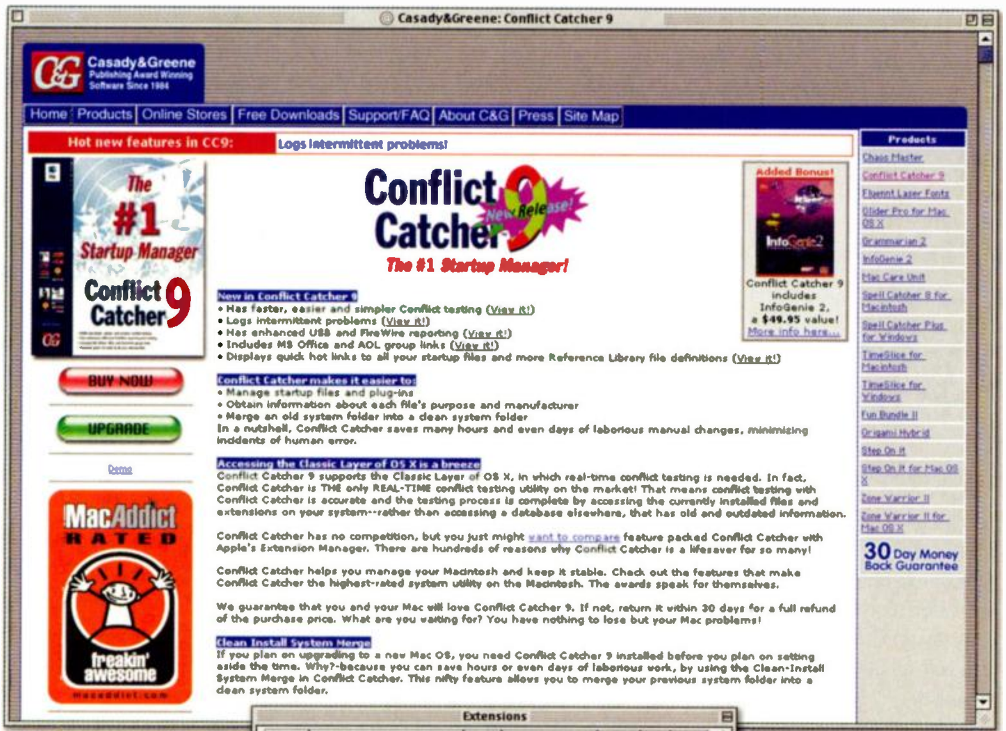
Using Extensions Manager, choose 'Mac OS Base' or 'Mac OS All' as the extension set and restart. If the problem still occurs, then restart with the Shift key held down. This disables all extensions (disabling them all in Extensions Manager only disables those that EM

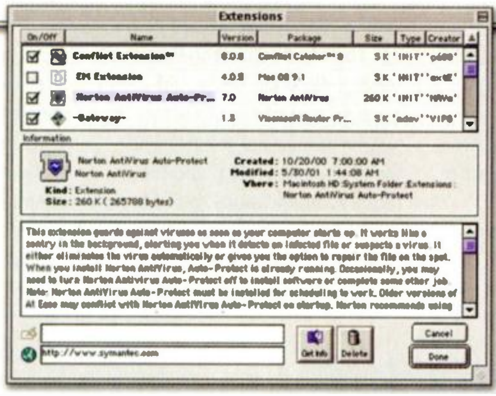
Test Specs

- Beige G3 tower with 333MHz processor.
- 384MB of RAM, with Virtual Memory off.
- Appletalk off.
- Default Disk cache.
- Emagic's *Logic* v4.8.1 assigned 100MB memory.
- Track-count tests had *Logic* set to 64 Tracks, Larger Disk Buffer on.
- Plug-in tests had *Logic* set to 16 tracks, Larger Process Buffer on.
- Audio tracks are continuous 16-bit 44.1kHz recordings.
- All plug-ins are *Logic*-native, set to stereo.



Extensions Manager showing extensions viewed as packages, and original system extensions labelled red. Control panels have been labelled green.





If Extensions Manager doesn't provide enough information or control for you, there are third-party applications that do basically the same job with extra features.

recognises, but this may also do the trick, since third-party extensions are usually in this group). If your problem still doesn't disappear at this stage, it can't be down to an extension conflict — a fresh install of your OS is then recommended. If the fault does disappear when restarting with a locked set, duplicate and name it. The procedure is then to re-enable extensions in small groups (restarting each time) until the problem reoccurs. This step can be made easier to track and understand if you use the 'View as Packages' option, which shows the extensions grouped according to function and/or install source.

Having found the suspect group (or package) you can narrow your search until you have found the offending extension. If you're lucky, the conflicting extension is not vital to the operation of your system and can simply be disabled. However, if the offending extension is related to the operation of vital software or hardware, there may be an update available on the Internet. Many extensions get updated as a result of reported conflicts arising from hardware/software combinations unforeseen by the manufacturer. If this brings no joy, then contact your supplier and the manufacturer's technical support as to possible fixes, such as suggestions as to which extensions may be in conflict. The ultimate solution would be to abandon that piece of hardware, but it is extremely rare that an extension conflict cannot be resolved.

Third-party Cover

If the method for extension management just described makes wet paint watching sound like the more enticing activity, there are many third-party utilities that can assist and automate the extension-conflict identification process. Perhaps the best known of all third-party extension managers is Casady and Greene's *Conflict Catcher* (available from www.conflictcatcher.com), which can find duplicate extensions and is a really good source of information as to the role of each extension.

Extension OverLoad (from www.extensionoverload.com) allows you to see and control

Extension Tips & Tricks

- **Extension Manager is loaded at boot-up by holding down the spacebar.**
- **Most audio software requires the QuickTime extension set (which you can easily see if you View by Packages in Extensions Manager) for features such as importing CD Audio and video playback. However, if you've disabled the networking extensions, the QuickTime Streaming extension will cause an alert at each boot-up unless also disabled.**
- **One neat tip I've picked up as a way of being better prepared for future problems after a clear install is to navigate to the Extensions folder in the System Folder, select all the files (hold down Command/Apple and press 'A'), and label them with a colour. This way, any third-party extensions added subsequently are easy to spot since they will appear with a different colour to the original set, and these colours will be visible in Extensions Manager.**
- **Mac OS 9 to 9.0.4 contains four Altivec-enabling extensions, which are built into later versions of the operating system and are removed automatically when you update Mac OS to a version greater than 9.0.4.**
- **Some versions of *Toast* had trouble with conflicts between the *Toast* extension and Apple's Firewire and USB Authoring Support extensions. From Version 5 this has been resolved — *Toast* now automatically disables these extensions on install.**
- **Some USB to SCSI adaptors require extensions that have been reported to cause freezes in *Logic*. Users of these devices, such as those who connect a legacy SCSI CD writer via such an adaptor, are recommended to set up two extension sets: one for *Logic*, the other for CD burning.**
- ***Recycle* v2.0 requires the following extensions to be enabled: OpenTransportLib, Open Transport Library, Shared Library Manager, Shared Library Manager PPC; and either Open Tpt AppleTalk Library, Open Tpt Internet Library, or OpenTpt Modem.**
- **The Extensions Manger can also be used to disable Control Panels. Steinberg's *Nuendo* requires both Appletalk and TCP/IP control panels to be switched off in the Extensions Manager (as well as turning off Appletalk in the Chooser).**
- **If you spot ObjectSupportLib in your extensions, disable it or, better yet, delete it from the Extensions folder, as it's known to cause instability to Mac OS 8 and disk corruption in Mac OS 9.**

extensions that exist in the Extensions folder but are not visible to Extensions Manager. There are good reasons, however, why Extensions Manager might exhibit this limitation. For example, Text Encoding Converter is not visible to Extensions Manager, but is to *Extension Overload*. Turning it off, however, will render the System Folder unusable in some versions of the Mac OS. If you wish to delve deeper than Apple's Extension Manager allows, one of these utilities might be worth investigating.

Mac OS 9.1	Tracks	Number of Stereo *PlatinumVerbs*	Number of *Distortions*	Mac OS RAM usage
'Mac OS All'	48	7	97	39.2
'Mac OS Base'	47	8	102	38.1
Extensions Off	48	8	98	20.4

Table 1: Test results under Mac OS 9.1.

Mac OS 9.2.2+	Tracks	Number of Stereo *PlatinumVerbs*	Number of *Distortions*	Mac OS RAM usage
All Extensions	47	8	98	46.2
Mac OS All	47	7	98	39.8
Mac OS Base	47	7	99	39.4
Extensions Off	48	6	98	20.9

Table 2: Test results under Mac OS 9.2.2 (with extra extensions added).

Performance Tests

Fine-tuning extension sets is often thought to be one way to achieve higher performance with OS 9, especially for those with older Macs that have limited bandwidth, memory and processing resources. There are countless threads on Internet forums devoted to the ideal extension set for the many combinations of hardware and software, so I decided to test some of the suggestions to see if real gains in performance are achievable.

Using a 333MHz beige G3 Power Mac and Emagic's *Logic* v4.8.1 as the basis of a typical older system, I tested track and plug-in count under a number of different circumstances. The purpose of trying a low CPU useage plug-in like *Distortion* was to confirm that the *PlatinumVerb* results were indeed arbitrarily different and not indicative of any trend (a four-percent change does not indicate a useful performance gain!). To begin with, I installed a new copy of Mac OS 9.1 onto the initialised system disk followed by *Logic*. My personal experience has proved that this is the most straightforward way to achieve a fast and stable system, since optimising a system already overloaded with unnecessary applications can often be a futile exercise. I tested this configuration with the default 'Mac OS All' and 'Mac OS Base' extension sets, along with no extensions loaded at all (by booting the Mac with the Shift key down). After that, I installed Mac OS 9.2.2 along with a collection of third-party extensions to simulate an overweight system and repeated the same tests, but with the addition of the 'My Settings' set with all the extra extensions enabled.

You can see my test results in Tables 1 and 2 above. From these results, I can safely say that *Logic* gained nothing significant from adjustments to the extension set in the test system. However, since many people have reported performance and stability gains from extension-set tweaking, I can only presume that I have not successfully recreated the creaking, overweight combination of OS and applications that make up these systems.

If you suspect that you may experience better performance from careful control of the Extension set, running tests similar to those described above will confirm any potential gains. My recommendation would always be to use your computer just for music and to install just the operating system and the required applications, plus the minimum number of additional extensions and system files to support any hardware you've installed. If you're unable (or unwilling) to return the computer to a fresh state by initialising the system disk and reinstalling from scratch, running regular disk maintenance utilities such as *Norton Utilities* or Alsoft's *Disk Warrior* is essential.

The test system had plenty of memory, which is essential for the smooth running of most music and audio software, especially since the operating system used about 46MB of memory in the test system, and figures of 70MB are not uncommon with more complicated system configurations. Of course, some older computers require RAM that is now so expensive that it is not justifiable; my Power Computing clone, for example, has been relegated to email and typing duties for just this reason. For quite some time I was under the impression that it had an extension conflict that I could not isolate, which resulted in a system freeze each time *Remote Access* was launched after the machine had been on for more than 20 minutes. Enabling virtual memory (which I can now do, as it is no longer used for music applications) has cured this problem, serving once again as a gentle reminder that extension conflicts are not the only source of problems!

No amount of extension tweaking can save the author's Power Computing clone, which must now be considered antiquated beyond salvation, and is being pensioned off to email and word-processing duties.

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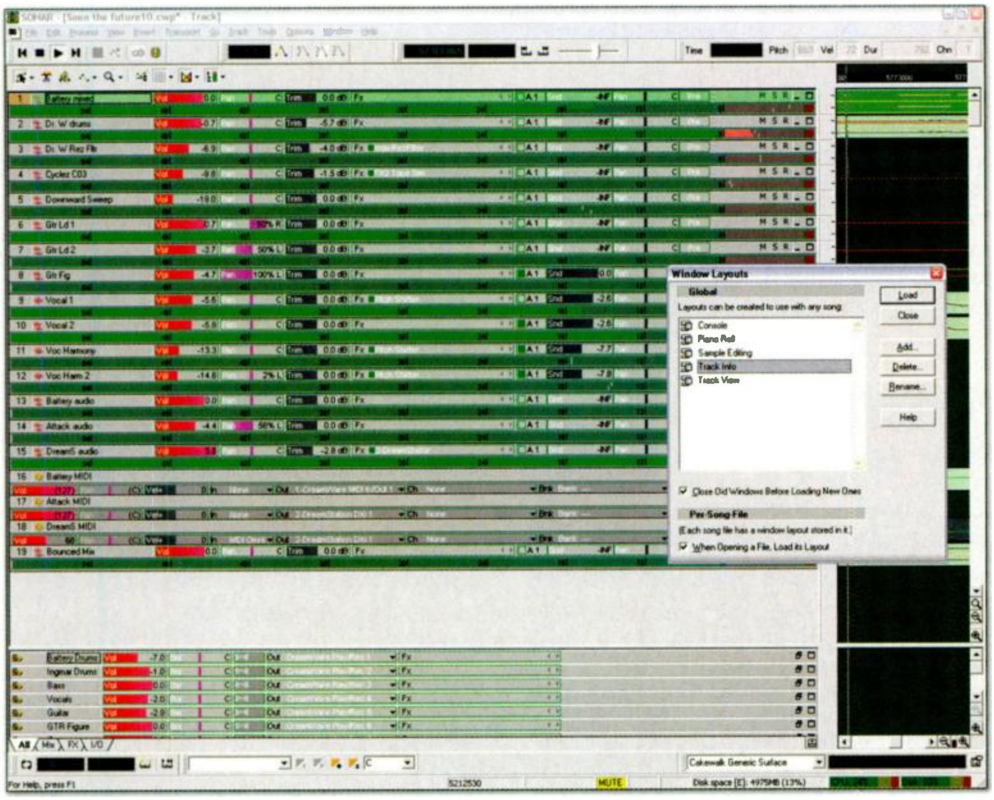


This Layout is extremely useful when you want an overview of all of a tune's parameters. For each track you can see the name and number, track type, volume, pan, effects, first aux send parameters, status (mute, solo, record), and metering.

Track down hidden sources of distortion, make your audio playback more reliable, and manage your window Layouts.

Craig Anderton

One of *Sonar*'s strongest points is that it can handle Groove Clips (*Acid*ised files that can be time/pitch-stretched). However, I recently noticed a strange problem where, even though Groove Clips were normalised to something less than 0dBFS, the track meters sometimes indicated overload. As I couldn't figure out a solution myself, I asked the Cakewalk brain trust if perhaps the meters showed clipping at a value slightly less than zero, but Cakewalk confirmed that the meter clip point is 0dBFS. Hmmm...

Eliminating Groove Clip Distortion

After putting in a lot of editing hours on my next loop library, I finally figured out what was going on. If a file plays back without distortion, is converted into a Groove Clip, and plays back at its 'native' tempo, it will not distort. However, speeding up or slowing down the song may cause slight volume increases that push the overall level past the point of overload. This is because the DSP that does the time-stretching relies a lot on crossfading, so if phase-coherent audio sections overlap, the signal peaks can add up and clip. This effect is not predictable. For example, a 120bpm Groove Clip might distort at 80bpm and 140bpm, but not 160bpm. So, when creating *Acid*ised files, my final step is now normalising them to -3dBFS.

Unfortunately, *Sonar* does not allow normalising to anything other than 0dBFS. You might think that normalising within *Sonar* then selecting 3dB Quieter from the Process menu's Audio submenu would accomplish the same result as normalising to -3dBFS. For some reason, though, that's not the case. So, I use Syntrillium's *Cool Edit Pro* v2 to call up the file and normalise it from there. Tick the Decibels Format box, tick Normalise To and enter '-3dB', leave DC Bias Adjust unticked, then click OK. So far, I have yet to create a Groove Clip that will distort, regardless of tempo, when normalised to -3dBFS although, of course, anything is possible...

There is one fine point: if you click on a Groove Clip within *Sonar* and select *Cool Edit Pro* from *Sonar*'s Tools menu, *Sonar* will not let you go there, because it assumes that wave editors can't handle Groove Clips. However, *Cool Edit Pro* v2 can process Groove Clips and retain the *Acid*ising markers. Therefore, I just open up *Cool Edit Pro* v2, and call up the loop file from there.

Managing Window Layouts

The Layout function lets you add, name, or delete particular layouts of windows and pane sizes. I first found out how useful this could be while working on a project that involved extensive use of both MIDI and digital audio, which required switching back and forth a lot between the Piano-roll view and the main Track view. So, I set up Layouts for the two views, which simplified switching between them. Any time you want to save a particular Layout, choose add from the View menu's Layouts submenu. This brings up a window where you can name the Layout. After naming it, click OK, which adds the new Layout to the list.

You probably don't want to go too nuts with Layouts, as scrolling through a long list of Layouts to find the one you want is kind of annoying, but I found I didn't need too many. In addition to the Layouts mentioned above for the Piano-roll and Track views, I created three more: ▶

Sonar Tips

One of my favourite keyboard shortcuts is the 'N' key, which turns Snap To Grid on or off in the main Track view and the MIDI views.

If you have lots of alternate takes waiting around for mixdown time, archive the muted ones by right-clicking on the track number in the track view, and selecting Archive Track. This saves CPU power.

Toolbars can dock along the bottom of the screen, not just the top. The lower part of the screen seems to be a good place for the looping and markers toolbars.

When doing MIDI, do you really need 960ppqn? You can improve MIDI timing accuracy on slower computers by going Options / Project / Clock, and selecting a lower value such as 240 or 360.

- Console view on top. I don't use the Console view very much, except to take advantage of the 'long-throw' faders for those few instances where I'm writing automation on-screen instead of using the Radikal Technologies SAC2.2 control surface. I arranged the Layout so the console's right edge extends slightly beyond the right-most side of the Track view. That way, if the Track view is on top, it's easy to click on the console's edge to bring it back to the front again.
- Sample editing. This changes the Time Ruler Format to Samples, and makes the Clips pane section very wide compared to the Tracks pane (when sample editing you usually don't need to access the aux send controls, ins and outs, and so forth).
- Track info. This is the usual Track view, but with the Tracks pane extended way to the right, with just a bit of audio showing. Calling up this Layout provides the equivalent of a Console view without having to use the console, as you can see auxes, pans, the complete track names, which effects are loaded in the FX Bins, and so on. In particular, this makes it easy to sort out the panning.

The Window Layouts function has a couple other notable options. If you tick Close Old Windows Before Loading New Ones, open windows that are not a part of the new Layout are closed before the new Layout is loaded. I find this helps keep clutter to a minimum. I also tick When Opening A File, Load Its Layout. In addition to being able to save Global Layouts, when you save a Song with a particular window Layout this Layout will be called up when you open the Song. If this box isn't ticked, then *Sonar* opens to the usual default Track view.

Finally, to make the Layout window as convenient as possible, I use the Key Bindings function (accessible from the Options menu) to assign the window to function key F2. Make sure Computer is selected as the Type Of Keys, scroll down the Key List until you see F2, then click on it. Next, scroll down the Function list until you see View Layouts. Click on it, then click on Bind. Now hitting F2 will call up the Layouts window, and once it is open you can use your computer keyboard's arrow keys to select the desired Layout, hitting Return to load. No mousing required!

Creating Crossfade Loops For Pads

Suppose you have a great sustained synth pad, like a drone or held note (or even a guitar power chord), and you want to turn it into a loop. Unless the loop start and end points share the same level and timbre, the process of looping will impart a rhythmic quality due to the change in sound when the loop repeats. Of course, this diminishes the pad effect.

Fortunately, as long as there's some audio in the original sample prior to the loop start point, there is a solution to this problem which we can borrow from the sampling world, called crossfade looping. Note that, for best results, the part you want to loop should not be normalised; there should be at least a few decibels of headroom, because what we're about to do may increase the signal level by a couple of decibels in a few places.

To begin, turn the section you want to loop into its own Clip by adding a split at the start and end (place the Now time over the start and press the 'S' key, then do the same for the end). Next, delete any audio to the right of this newly-created Clip. Copy the Clip to the right so that its beginning butts up against the end of the original. Now slip-edit the beginning of the copied Clip toward the left, which creates a crossfade with the end of the original Clip. As a result, upon reaching the end of the loop, the original Clip will include some of the audio leading up to it — the secret of creating a perfect splice.

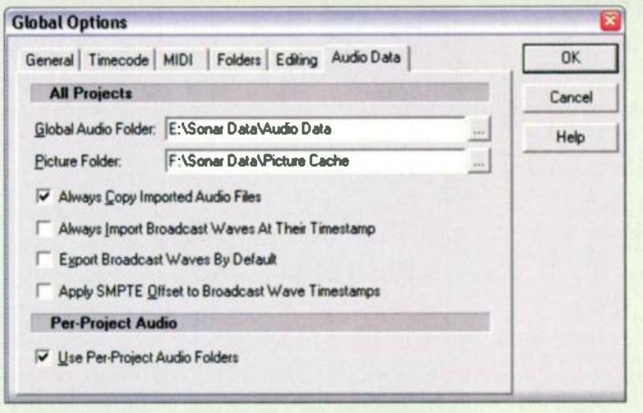
Now all you need to do is split once more at the end, so that the crossfaded section is on top of the original loop. Draw a marquee around the combination of these two Clips to select them, then choose Bounce To Clip(s) from the Edit menu. The crossfaded section will mix in with the original Clip. If you loop this Clip, the transition from end to beginning should be seamless. All that's left is to turn it into a Groove Clip — the quickest way to do this is to select the Clip, then type Ctrl+L. You may want to go into the Loop Construction view to tweak the slice points so that pitch-stretching is available over the widest possible range, but *Sonar*'s default loop points will probably do the job. SOS

Moving The Picture Cache

Sonar stores all the waveform images for your projects in a folder called Picture Cache. When you install the program, this defaults to the path C:\Cakewalk Projects\Picture Cache. But think about your C drive: it holds your operating system, as well as program files. Cakewalk defaults to putting audio data inside the same Cakewalk Projects file on your C drive, so you're pulling a lot of data off that poor, overworked drive. Granted, modern hard drives are fast, so you may not notice any problems. But if you have a lot of hard disk tracks with numerous loops that require significant graphics redraws, consider storing your audio and graphics data elsewhere.

To do this, go to the Global submenu of the Options menu, and select the Audio Data tab. This shows the path for the Global Audio and Picture Folders. Simply create new folders for this data on the desired drives, and enter the new path in the appropriate fields. For example, I have a fast E drive that's dedicated to storing data, which is where the my _Sonar_ audio files reside. The picture data, on the other hand, lives on the F drive. As a result of separating out the data, when doing hard disk recording, the hard disk tracks are pulled off disk at the fastest possible speed from a dedicated (and of course, defragmented!) drive — definitely the way to go.

For the most trouble-free transition when changing to a new file path, I suggest copying over (not moving) the existing folders to their new destinations before opening _Sonar_. Then open _Sonar_, change the file paths as described above, and close _Sonar_. Open it again, and now your projects will reference the new folder locations. If everything works as it's supposed to, you may then erase the original audio and picture folders.



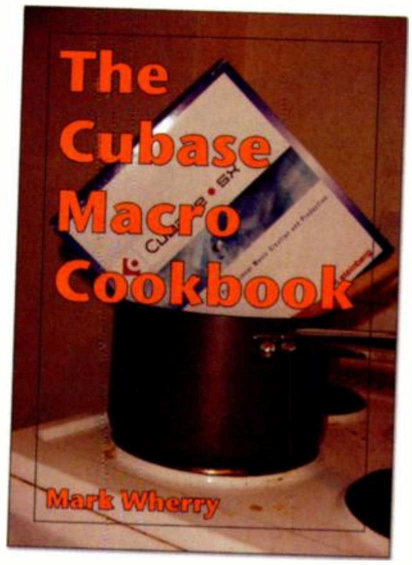
The Global Options area has a lot of ways to customise *Sonar*, and under the Audio Data tab you can set up paths for the global audio and picture folders. Note that the Folders tab lets you specify other locations, such as where Sonar should look for video files, drum maps, templates, and so forth.

Current Versions

- *Sonar/Sonar XL*: v2.2.

cubase notes

This month we take another look at *Cubase*'s Macro facility and discuss a selection of examples for adding some rather neat features to the program.



Mark Wherry

With so many new *Cubase* books appearing on the market these days, I often wonder how long it will be until we see such titles as *Cooking With Cubase*, although I admit that I'm not completely blameless in this department. Unfortunately, from a personal perspective, I'm no Nigel Slater; but to honour of one of the UK's finest TV chefs, in this month's *Cubase* Notes I offer you instead — wait for it — the *Cubase Macro Cookbook*.

For the uninitiated, *Cubase*'s Macros allow you to program a sequence of Key Commands as a single Key Command, and the Macro functionality is accessed from the Key Commands window (File / Key Commands) by clicking the Show Macros button. You can add a new Macro by clicking the New Macro button, typing a name and pressing Return. To add commands to the Macro, simply select the required Key Command in the upper part of the window and click the Add Command button. Macros can be assigned to a Key Command from the Macros Category, or triggered from the Edit / Macros submenu. For a more detailed introduction and overview of Macros, see January 2003's *Cubase* Notes (www.sospubs.co.uk/sos/jan03/articles/cubasenotes0103.asp).

Hide & Show Soufflé

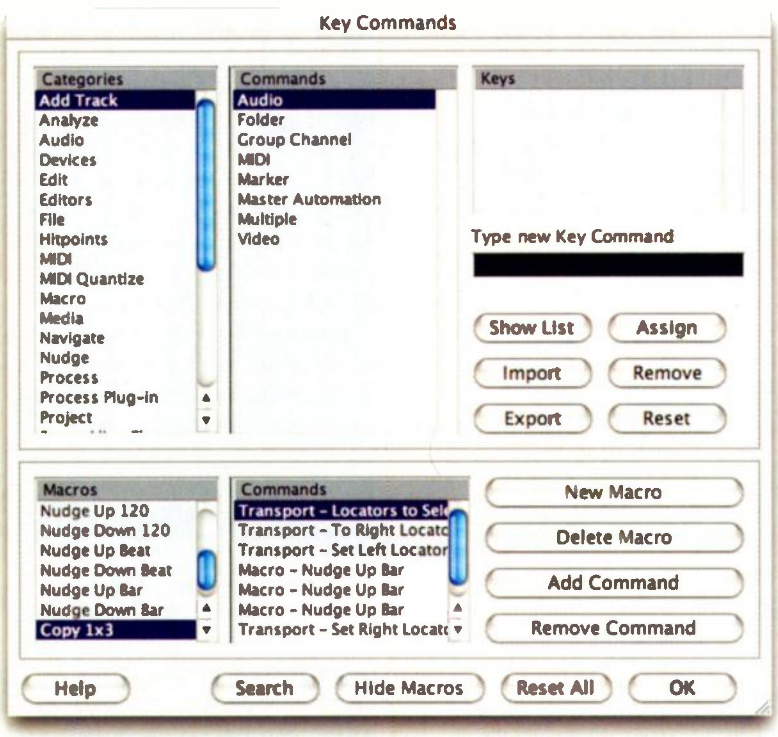
To start with, here's a simple recipe I call 'Toggle Concise Project Window'. If you look at the Editors Category in the Key Commands window, you'll notice that the Show/Hide Infoview, Inspector and Overview commands for the Project window are all available as Key Commands. So, assuming you usually enable all three of these at once (and you'll need to have all three visible or not visible for this to work properly), you can toggle all three of these commands in one Macro and flip between the two Project window views with one Key Command — Alt/Option+Shift+C, for example. Simple, but useful; and if you only use the Infoview and Inspector, for example, you could leave the Hide/Show Overview command out of the Macro.

Unfortunately, there aren't any other windows where global hide and show commands are possible: although *Cubase SX*'s advanced Score Editor features the Icon and Filter bars, which can both be toggled from the window's toolbar, for example, these particular toggle commands aren't available as Key Commands. However, you can assign the individual filters on the Filter bar to Key Commands to turn them on and off, so these could be grouped together into a Macro if required. Staying with *SX*'s Score Editor, a seemingly good idea would be to toggle the display of a selection of Symbol Palettes with a Macro, since these can be assigned to Key Commands. Unfortunately, though, the Key Commands for the Symbol Palettes only open the palettes — selecting the same Key Commands again won't close the palettes.

Quick Fried Grooves

Now for a Macro that can save you time: 'Slice Audio Loop'. We looked at the commands required to slice an audio loop in *Recycle* fashion in September 2002's *Cubase* Notes (www.sospubs.co.uk/sos/sep02/articles/cubasenotes0902.asp), but why not automate this process with a Macro? Add the following commands to a new Macro in this order: Edit-Open/Close Editor, Hitpoints-Calculate, Hitpoints-Create Audio Slices, and assign it to, say, Alt/Option+Shift+L. Now, when you have an imported Audio Event you want to play in time with the current Project's tempo, select it and use the Slice Audio Loop Macro — and nine times out of 10, the default settings will work their magic.

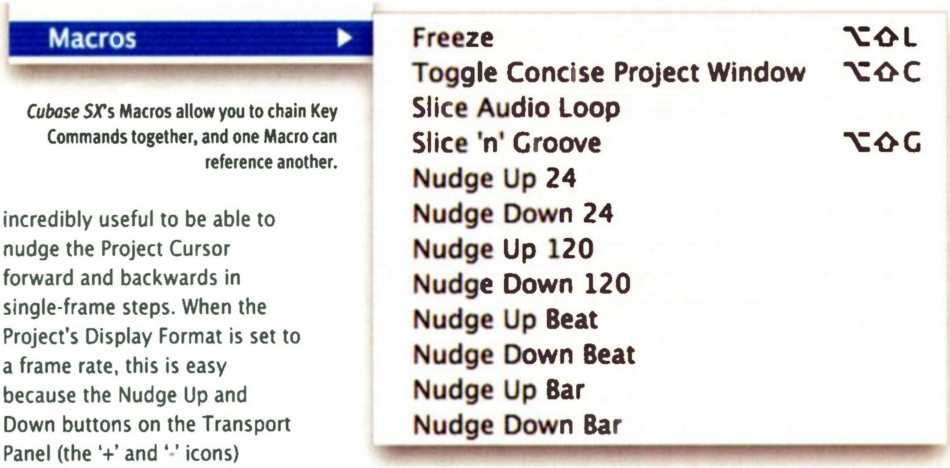
If you wanted to save even more mouse clicks, you could create another Macro with the same commands, but add Hitpoints-Create Groove Quantize before Hitpoints-Create Audio Slices — let's call it 'Slice & Groove' and assign it to Alt/Option+Shift+G. This way, a Groove Quantize preset is added to the Quantize menu at the same time you create the audio slices. I experimented with adding extra commands after the basic Slice Audio Loop Macro as well, but, unfortunately, it seems that currently no command in a Macro can be executed after Hitpoints-Create Audio Slices, even basic Transport commands.



The Key Commands window, where you define and edit *Cubase* Macros.

Project Cursor Sauté

When you're working with video in *Cubase*, it's



Cubase SX's Macros allow you to chain Key Commands together, and one Macro can reference another.

incredibly useful to be able to nudge the Project Cursor forward and backwards in single-frame steps. When the Project's Display Format is set to a frame rate, this is easy because the Nudge Up and Down buttons on the Transport Panel (the '+' and '-' icons) always nudge the Project cursor by the smallest unit in the current Display Format, which would, of course, be single frames when you're working with a frame rate. However, since we tend to write music in terms of bars and beats rather than seconds and frames, what happens when the Display Format is set to Bars+Beats instead? Since the smallest unit in Bars+Beats mode is ticks, the Nudge Up and Down buttons now nudge the Project cursor in single-tick steps, rather than frames.

Fortunately, the number of frames per second can be neatly divided into the number of ticks per second when you're working with 24- or 30-frame video, meaning that a single frame is worth 40 and 32 ticks respectively. So to nudge the Project Cursor by a single frame with video in either of these frame rates when the Display Format is set to Bars+Beats, all you have to do is press either the Nudge Up or Down button over and over again, the required number of times — great! As you can imagine, Macros can easily take the strain out of such repetition, and all you need to do is program a Macro to trigger Nudge Up or Down the required number of times. Assign these Macros to Key Commands and it's now very easy to nudge by single-frame steps when working with bars and beats. I usually give these Macros names such as 'Nudge Up 1/24/40', 'Nudge Down 1/30/32', and so on.

There are other situations where it can be useful to create a Macro with a sequence of Nudge Up or Down commands — for example, it's easy to nudge in single frames when you're working with frames with the Nudge Up and Down commands, as we've seen, but what if you wanted to Nudge Up and Down a second? As you can guess, for 24-frame video, the answer is simply to create a 'Nudge Up 24' and a 'Nudge Down 24' Macro, each containing 24 Nudge Up or Down commands respectively. While these last two Macros might seem a little simplistic, I'm recommending them because they'll come in useful for another recipe in just a moment.

Copy Creams

You'll often want to create a number of copies of an Event or Part when working with loops, and while *Cubase* offers a variety of tools to make copying objects fairly easy, wouldn't it be great to have a 'make three copies instantly' button, making a one-bar loop play over four bars, or a 'seven copies' button to cover eight bars? As we've already seen, Macros are a great way to trigger the same command over and over again; so, in theory, a 'copy three times' command should be easily created by simply creating a Macro that triggers the Duplicate command three times. If only it was that easy...

It turns out that triggering the Duplicate command three times from a Macro does exactly the same as using the Duplicate command once. I suspect the problem lies with the Macro sending the next command before the previous one has completed, which is something that could be addressed by Steinberg. My next thought was to build my own Duplicate command, by copying the current selection, moving the Project Cursor to the end of the selection (by using the Locator commands), and pasting — Edit-Copy, Transport-Locators to Selection, Transport-To Right Locator, Edit-Select None, Edit-Paste. However, this fails because the Paste command is somehow executed before the Project Cursor is moved, so the object being copied ends up being pasted to wherever the Project Cursor happens to be when you execute the Macro. Very frustrating, for sure, but at least it's a lesson learnt.

My final solution uses the Edit / Fill Loop command and is less flexible than I'd initially hoped, but hopefully it will serve to offer some other ideas for your own Macro experiments. First of all, create a new Macro called something like 'Copy 1x3' because what this Macro will do is a copy a selected one-bar object three times so it plays over four bars, although these numbers can easily be adapted. The way it works is quite simple: first set the locators around the selected object, move the Project cursor to the Right Locator (the end of the selection) and set the Left Locator to this position as well. Next, we'll move the Project Cursor forward three bars, set the Right Locator to the new position of the Project Cursor and trigger the Edit menu's Fill Loop command.

The secret to making this Macro work is how you move the Project Cursor forward by three bars, and guess what? We need to do this in single-tick steps with the aid of the Nudge Up command we created earlier, although this isn't as bad as it sounds. Because it's possible for a Macro to trigger another Macro, we can create a 'Nudge Up Beat' Macro with 20 'Nudge Up 24' commands, since there are 120 ticks in a 16th note (24 x 5 = 120) and four 16th beats in a quarter note.

So the finished Macro consists of: Transport-Locators to Selection, Transport-To Right Locator, Transport-Set Left Locator, Macro-Nudge Up Bar, Macro-Nudge Up Bar, Macro-Nudge Up Bar, Transport-Set Right Locator, Edit-Fill Loop. Activate it when you have a one-bar object selected, and you'll get three more copies after it. While this may not be the most exciting Macro you'll ever see, it hopefully illustrates a few points about what is and what's not possible, and will give you a few ideas for other Macros along the same lines. We'll present other recipes from the *Macro Cookbook* in future *Cubase* Notes, and, as always, if you come up with any interesting Macros, do email them to us at sos.feedback@soundonsound.com. SOS

Macro Tip

If you're having difficulty persuading *Cubase* to let you create Macros, it's worth remembering that *Cubase* won't let you create another Macro if the last Macro in the list (probably the one most recently created) doesn't contain any commands. This can be a problem because it's quite common for people to want to create Macros as 'placeholders' and program them later on.

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pro tools notes

The subject of clocking in digital audio systems is often a hazy area of understanding among us users, perhaps because we've been turned off by propellerheads who love arguing about it. Here's what you need to know in Pro Tools from a practical angle.

Simon Price

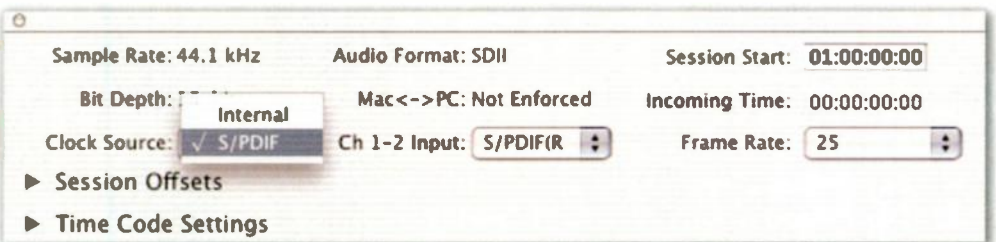
No matter what kind of Pro Tools system you use, somewhere or other in your studio there's a piece of hardware whose contribution is to mark the passage of time. This might be the timing crystal in your Mix system's 888/24 interface, or your computer's internal clock in a *Pro Tools Free* setup. Obviously, if Pro Tools is recording or playing back audio at, say, 44100 samples per second, it has to have a way of telling what a second is, or more to the point, what 1/44100th of a second is. For a single isolated audio system, such as one Pro Tools rig with one audio interface, the clock needs to be reasonably accurate and consistent so that the music plays evenly and at the right speed (and therefore the right pitch). The situation becomes complicated when there's more than one digital device involved. This is the case when transmitting digital signals between two devices (such as Pro Tools and a DAT machine), but also when you have more than one Pro Tools audio interface. The difficulty lies in the fact that inevitably there will be slight variations in the devices' clocks. At everyday human timescales, you might notice two watches varying by a few seconds from each other over a week, which at worst might make two extremely punctual people arrive at the pub a few seconds apart. However, when you're measuring time in units of 1/44100th, or even 1/192000th of a second, clock variations become apparent very quickly, and in digital audio the consequences are more severe: loud clicks in the audio signal.

The answer is very simple: any piece of digital audio gear worth its salt should have the option of switching to an external clock source so that all your gear can sing from the same hymn sheet.

Multiple Audio Interfaces

Even if you never normally have to think about clocking because your stuff's all analogue outside of Pro Tools, you may have had an encounter with it when you put your TDM system together (*LE* systems only ever have one interface). This is because when you have more than one audio interface (such as two 888/24s or 96I/Os), they must be configured to lock to the same clock source. On TDM systems other than the new HD generation, Pro Tools interfaces have two BNC connectors — 'Superclock In' and 'Superclock Out' — for sending and receiving clock signals. Superclock is a very fast clock pulse signal used by Digi instead of the more common word clock found on most digital audio gear (see box). The first interface connected to your Pro Tools PCI cards is considered the 'clock master' for the whole system, and should provide the clock to all your interfaces and PCI cards. You set this up by 'daisy-chaining' together your interfaces with Superclock connections. As well as its main connection to the computer via the PCI card, each interface should have a BNC cable running to its Superclock In port from the previous interface's Superclock Out connector.

If you have Digi's Universal Slave Driver box, or another sync box that generates Superclock (such as MOTU's MTP/AV), this takes the place of the first interface at the top of chain and becomes clock master. This is how sync boxes keep Pro Tools running in time with the sync source: the USD reads incoming timecode and varies the Superclock rate to speed up or slow down Pro Tools accordingly.



Switching Pro Tools' clock source to external digital (in this case S/PDIF) is necessary to ensure clean transfers from DAT.

Current Versions

Mac OS X
- Pro Tools HD, Mix, 24, 001, M Box: 6.0.1.

Mac OS 9
- Pro Tools HD: 5.3.1cs5.
- Pro Tools Mix, 24: 5.3.1cs10.
- Digi 001: 5.1.1.
- Digi 002: 5.3.2.
- M Box: 5.2.

Windows
- Pro Tools HD: 5.3.3.
- Pro Tools Mix, 24, Digi 001: 5.3.1.
- M Box: 5.3.3.

With the new Pro Tools HD hardware, Digi have changed the way clock is distributed throughout the system. The main difference when configuring the new Loop Sync is that the clock out connector on the last interface is cabled back to the clock input of the first device, creating a closed loop.

In The Field

Whatever kind of Pro Tools system you have, you can use your clock skills to make sure that digital connections between Pro Tools and your other gear are solid and click-free. This can be a permanent arrangement, such as a connection to a digital mixer, or a temporary change such as connecting up a hired DAT machine. Without any sync devices like a USD, any Pro Tools system (except *PT Free*) can be set to Internal or Digital sync mode. There is a pop-up toggle for this in the Session Setup window. In Digital sync mode, Pro Tools will switch to using

Quick Tips

While the audible signs of incorrectly configured clock are clicks, static, or grunge in the audio, you can often see there's a problem before any audio is running because the meters on your interfaces will randomly spike.

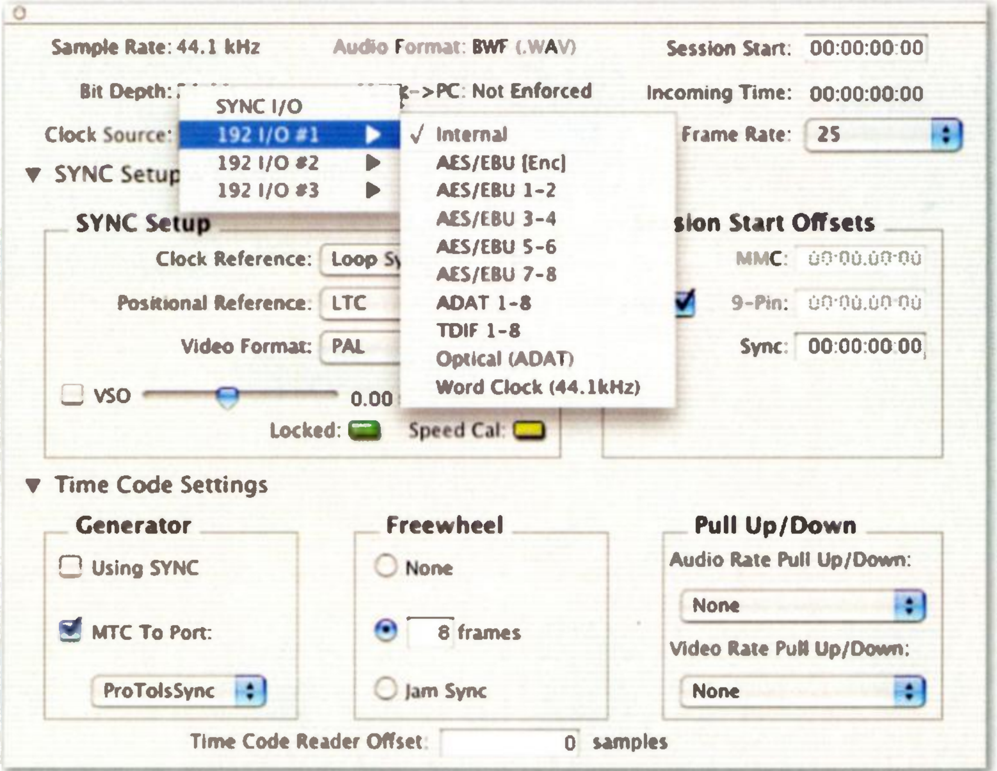
The left and right cursor keys re-centre the screen on the start and end of a selection respectively. If you centre on the end like this, the zoom keys switch to zooming in and out on this point, whereas in all other circumstances they would move the view to centre on the selection start.

Apple Core Audio Drivers for OS X are now available for download from www.digidesign.com, allowing you to use your Digi interfaces with other software in OS X, including *Reason* 2.5 (hooray!). You'll need *PT6* installed because it uses some of the components, but there'll be a download soon which gets past this requirement. A down side is that *iTunes* 4 won't work, but there's an update on the way.

whatever device is currently connected to your first digital input. It can do this because digital audio signals contain clocking information that any device on the receiving end can use. Remember, this is purely a data-rate locking system, not a sync system like timecode (see box for more on this). The basic rule of thumb is to make the source device the clock master and have the destination set to digital/external sync. If you want to record something into Pro Tools from DAT, for instance, you should plug it into digital input 1-2 on your first interface, and switch Pro Tools to Digital sync mode. On some systems you need to select whether you're using the AES, S/PDIF, or optical input type. On HD systems the Session Setup box will let you choose any of the digital inputs as the clock source, a benefit of the new Loop Sync system. Either way, your whole Pro Tools system will be telling the time exactly the same as the DAT, resulting in a clean digital audio transfer. Alternatively, you might be going the other way, with Pro Tools sending audio to a digital mixer and a DAT or other master recorder. In this case you should poke buttons on your mixer until it goes into external clock mode, or you could read the manual to save time!

Knowing this much should get you by in most circumstances, but there is another common scenario in larger studios when you have a USD or Sync I/O unit in your setup. Digi's sync boxes can be used to lock Pro Tools to just about any known type of clock signal, including the standard word clock and Superclock. As I've already hinted, they can also generate a varispeed clock for Pro Tools based on incoming timecode. Most commonly, though, you see them connected via a BNC plug labelled Video Reference (back to that in a second). All of these options are selected from within *Pro Tools'* Session Setup window under the Clock Reference pop-up. This is in contrast to the Positional Reference selector, which chooses a source of timecode when required.

Anyway, video reference is used universally in TV/Film post production, and increasingly in music studios, and is also known as 'black and burst' or sometimes 'house sync'. The idea is to take a standard timing (clock) reference from a dedicated box, and distribute it to all your equipment, so that you can forget about clocking most of the time. There may be a 'black and burst generator' for the room, or for the whole studio facility, allowing tie-lines to run digital audio between rooms. While a standard black video signal is the most common form of 'house sync', the idea could stretch to other formats. For example, some will buy a dedicated Superclock generator to run their Pro Tools interfaces, rather than daisy-chaining the built-in clock. The stability of a system's clock source has a bearing on the audio quality, which explains why arguing about which clock source to use is one of those Internet discussion forum perennials.



Pro Tools HD systems can set any of their digital inputs as the master clock.

Superclock & Loop Sync: What's The Difference?

So what's with all these clock formats, and what exactly are they? If you're sitting comfortably, I'll clip on my beard. The basic digital clock type is word clock, which is embedded in digital audio signals, but can be sent on its own via a dedicated word clock connection. This is a code within the digital stream of ones and zeros that says 'the next sample (or bunch of numbers that represents the sample) starts here', allowing the destination machine to 'slot in' the received numbers at the right time position. It's the same concept as the squiggle in a video signal that says 'the next frame starts here' so the picture can be lined up at the top of the screen.

Normal word clock runs at the same rate as the sample rate, so is adequate for transferring digital signals from place to place. Pro Tools' TDM mix environment can handle 256 digital audio streams, busses or connections, so it needs a master clock that runs at 256 times the speed of normal word clock. This is where the requirement for Superclock comes from, and explains why it needs to run at 256x. ADAT optical connections, on the other hand, send eight channels of audio down one cable, so need another format again. Pro Tools HD and TDM II feature high bit-depth samples and 512 possible streams in the mixer, meaning that Superclock is no longer adequate. Rather than make an even faster clock, which would be more prone to timing innacuracies (called 'jitter'), Digi rethought the design and came up with Loop Sync, which allows the system to self-monitor and stabilise its own clock.

One thing that clock is not is timecode, as it only indicates speed, rather than any particular time. Pro Tools differentiates between timecode and clock by calling them positional reference and clock reference respectively. This could cause confusion in versions previous to *PT6*, because the internal/digital reference mode selector is labelled 'sync mode'. It's confusing because when you're chasing timecode you will most likely be using clock from your sync interface, which means that the sync mode setting should be set to internal. Anyway, they've relabelled it in v6 so I'll shut up now!

More on *DP4* and OS X this month, including the new 'Freeze Tracks' feature...

Robin Bigwood

The new Freeze Tracks feature is a great addition to *DP*, and especially so if you own an older or more modest Mac. There can't be many *DP* users out there who haven't pushed their Mac's processor to the limit whilst working on a project. Extensive use of effects plug-ins sap processor power, resulting in an increasingly sluggish user interface, pops and clicks in audio, and even system instability. Getting around this problem has always been possible, by creating a new voice track, routing all the tracks laden with software synths or plug-ins to it via a buss, record-enabling it, making a recording pass then disabling the plug-ins. It's straightforward in principle, but a really tedious chore that can take 10 minutes to set up, and even then, it results in the loss of individual control over the tracks you've just bounced.

This is where the new Freeze Tracks feature comes in. In a single action it takes any tracks you've selected (whether they have plug-ins on them or not), creates the same number of new tracks, bounces the selected tracks to them, and then disables the audio voice assignments in the original tracks it has just 'frozen', freeing up processor power. The advantage of doing it this way is that although you don't get to tweak plug-in parameters any more, you maintain individual control over level and pan, and can of course add more plug-ins!

Using Freeze Tracks isn't difficult. You select an audio track by clicking its name or dragging over a part of its track in an editing window, hold down Control and Command/Apple and hit the 'F' key (or choose 'Freeze Selected Tracks' from the Audio menu). Your new track appears and the original is 'frozen'. If you want to return to the original track select it (*not* the bounced track), hold down Control, Shift *and* Command/Apple and hit the 'F' key again (or go to the Audio menu, hold down Shift, and select 'Unfreeze Selected Tracks').

The above works for multiple tracks, and also Aux tracks, so it'll be possible to freeze software synths in this way too. Also bear in mind that Freeze Tracks is perfect for capturing the output of external effects processors being brought into *DP* via an Aux track — freeze the Aux and you have a 'hard copy' in *DP* that you can mix, tweak and edit to your heart's content.

Using AUs In *DP4*

Although early *DP4* adopters have had to make do without the vast array of MAS format plug-ins that were available under OS 9, they're perhaps compensated to a certain extent by the promise of native support for Audio Units later in the year. Audio Units (or AUs) may well become the standard format for audio plug-ins under OS X, so it makes sense to start getting a feel for them now. But how can *DP* users do that?

The answer lies with *Rax*, an application by Granted Software (who also make the *iMIDI* utility featured in Mike Watkinson's Apple Notes column back in February). *Rax* is perhaps the first stand-alone Audio Units host, and although it's still in a relatively early stage of development, it nonetheless offers robust support of most Audio Units plug-ins, including software synths and effects. Operation is simple — after you boot it up you're asked to select an Audio Unit synth plug-in from any of the ones you've put in Library/Audio/Plug-Ins/Components (the OS X location for plug-ins, as explained in last month's Performer Notes). You can then play the synth plug-in by selecting an appropriate MIDI source from the dedicated pop-up menu. AU effects (including the handful that are part of OS X; there's a surprisingly good reverb) can then be applied to the output of the synth.

What makes *Rax* really useful is that you can play it from within *DP4*, and here's how. First of all, start up *DP* before you boot up *Rax*. Go to the Setup menu, select 'Interapplication MIDI...' and create an Output called (for example) 'DP4 to Rax 1'. Now, close Interapplication MIDI, start *Rax* and select an AU instrument. Here's the clever bit — from Rax's MIDI Source pop-up menu select 'DP4 to Rax 1', which Rax recognises because *DP4* has 'published' it to the Core MIDI part of OS X. Back in *DP*, you should now find 'DP4 to Rax 1' in the output menus of all your MIDI tracks. Record-enable one of them, and you'll be away.

Granted Software's *Rax*, a stand-alone AU host application that you can use with *DP4*.

Interapplication Audio

Whilst the combination of *DP4* and *Rax* is useful, it has a major shortcoming in that there's currently no way to get audio from *Rax* (or any other application that doesn't support Rewire) into *DP4* without it having to pass in and out of your Mac via some sort of audio interface. Before OS X, *DP* users had a solution to this problem in the form of *AudioTap*, a little MAS input plug-in which could take any audio being handled by the Mac's Sound Manager and route it into a suitably record-enabled *DP* audio track. It was great — oddball software synths, *RealPlayer*, DVD soundtracks and system sounds could all be recorded in *DP* and take advantage of *DP*'s mixing and processing architecture — but now *AudioTap* is gone for good.

However, there is some brighter news, in the form of the Linux 'audio server' *Jack*, which is currently in an advanced stage of being ported to OS X by Stephane Letz, a French software developer. *Jack* makes it possible for Interapplication audio to take place on a very flexible basis, but the downside is that audio applications have to be specially adapted in order to use its features. But before you quite rightly exclaim, 'MOTU will never do that!' there's more. A freeware

Quick Tips

dp If you're using *DP4* to play synths in *Reason* (or a VST or AU host application) you might, like me, have experienced interapplication MIDI communication ceasing when you switch out of *DP* (to adjust synth parameters, for example). In *DP3* this could be remedied by selecting 'Play In Background' in Preferences and 'FreeMIDI applications only' in FreeMIDI Setup's Preferences. To achieve the same in *DP4*, you need to go to 'MIDI Patch Thru...' in the Studio menu and select 'Patch Thru in Background'.

dp Two important *DP* keyboard shortcuts have changed in *DP4*. Heal Separations is now accessed by holding down Alt/Option and hitting the 'H' key, to avoid a conflict with OS X's 'Hide Application' function. Play Selection is now accessed by holding down Alt/Option (not Command/Apple) and hitting the spacebar. The keystroke with the Command/Apple key is now used to cycle through international keyboard layouts.

OS X framework called Application Enhancer (or 'APE') can 'patch' existing applications to give them additional features, and it's highly likely that soon any audio application using Core Audio (ie. anything that runs under OS X) will be able to be patched to allow it to be 'jackified'. Stephane Letz already reports success with patching *DSP Quattro X*, so with any luck *DP4* will soon have interapplication audio capabilities once more.

In the meantime, if you want to get a feel for what APE can do, have a look at an application which relies on it, *Audio Hijack Pro*. This is a kind of 'mini'-*DP*/*AudioTap* combination — a stand-alone audio recorder that supports AIFF and MP3 file formats along with VST and AU effects plug-ins, and can record audio from any application that uses OS X Core Audio (including *DP4*). *Audio Hijack Pro* is a pretty nifty bit of software to own even if *Jack* for *DP4* is in the pipeline, because along with letting you, amongst other things, record *iTunes* internet radio streams, you can also use it to do really quick MP3 (or AIFF) bounces of *DP4* sequences. Just set it up to record *DP4*'s output, and make sure that any external sound sources in *DP4* are being brought into the Mixing Board via Aux Tracks. Then start *Audio Hijack Pro* recording and hit play in *DP4*. It's great for collaborative projects, or for just knocking out something to listen to on your iPod.

For more info on all this, surf to http://jackit.sourceforge.net (*Jack*), www.unsanity.com/haxies/ape (*Application Enhancer*) and www.rogueamoeba.com/audiohijack (*Audio Hijack Pro*).

Preferences Files

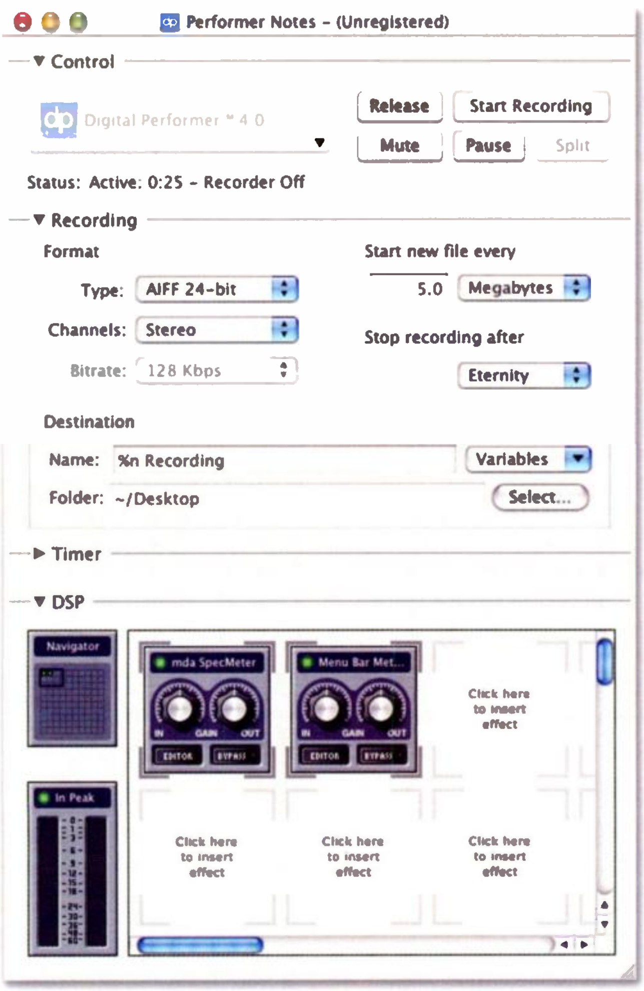
Although it has never happened to me in over a year of using OS X, it's perfectly conceivable that certain applications may end up with corrupted preferences files from time to time, particularly following a crash. Since trashing preferences was, sadly, so much part of life with *DP* and FreeMIDI under OS 9, it's probably worth knowing how to do the same under the new OS. OS X maintains preferences on an individual user basis, so if you experience problems with *DP4* that you suspect might be solved by trashing its preferences, the place to look is Users/[your username]/Library/Preferences. Here you'll find a number of files relating to MOTU and *DP4*, such as 'com.motu.DigitalPerformer.plist' and a folder named 'Digital Performer' containing files such as 'Preferences' and 'Command Bindings', very similar to the one that used to be in the Extensions folder if you used *DP* under OS 9.

First of all, the plist file (which, in OS X terminology, is a 'property list') stores information relating to *DP*'s use of OS X's navigation services — the default locations in an Open or Save dialogue box, for example. Though it's unlikely you'd ever need to, trashing it does no harm at all, since a new one gets written the next time you boot up *DP*. If you're really curious you can open plist files, which are in XML markup language format, in an application like *BBEdit* or the dedicated *Property List Editor* which is installed as part of the OS X developer tools package. It's probably best to resist the temptation to change anything you see, though!

However, the heart of *DP4*'s preferences is in the 'Digital Performer' folder, but again you can't do any harm by trashing these files. 'Preferences' seems to be *DP4*'s fundamental preferences file, whilst 'MOTU Audio System Prefs' is probably the prime suspect for audio-related gremlins. Interestingly, it appears that *DP4* is largely compatible with preferences files produced by *DP3*, to the extent that you can use an OS 9 'Command Bindings' file to carry over carefully-honed keyboard shortcuts into the new operating system. The next time you boot up *DP4* it recognises the fact that you've fed it an 'old' file, and offers to update it for you to take account of some system-wide shortcuts that might cause a conflict (see the 'Quick Tips' box for more on this).

Packages & Bundles

Last month I mentioned that *DP4*'s MAS plug-ins appear in the Finder as files with the name 'something.bundle', and said that although they *look* like folders they can't be opened. Well, that wasn't strictly true, because like Audio Units (which are called something.bundle) they're actually 'packages' whose contents can be examined by control-clicking (or right-clicking) on them and choosing 'Show Package Contents' from the pop-up menu that appears. Most of the contents of packages like these will only be of interest to programmers, but it's interesting to note that inside Contents/Resources/Presets are individual Effects Settings Clippings that correspond to presets which show up in the plug-in's pop-up menu inside *DP*. Their existence is worth bearing in mind for two reasons. First, despite what you might expect, user presets are *not* stored amongst them — they're still saved within individual project files, so if you want to build up a 'user bank' of presets you need to use *DP*'s Clippings as before. Second, you can actually delete, replace or add to them, for the ultimate in user personalisation — but don't even think about trying this unless you're completely confident about what you're doing! Nonetheless, *DP4*'s bundle files are yet another example of how things have fundamentally changed under OS X. And on that note, be sure to read next month's Performer Notes, in which I'll be looking in detail at the new XML patchlist format that *DP4* employs. SOS



Using *Audio Hijack Pro*, all sorts of system and application audio can be recorded ready for inclusion in *DP*. It can also make quick MP3 mixdowns of *DP4* sequences.

Current Versions

- MOTU *Digital Performer*: v4.0 (OS X).
- MOTU *Digital Performer*: v3.11 (OS 9).

Relive the hardware of the early synth pioneers this month, as we show you how to construct your very own matrix sequencer in the Environment.

Steve Knee

Back in the heady days of analogue synthesis, people tried to find ways of playing synths automatically, and various ideas emerged. One early design was the matrix sequencer, which had perhaps eight or 16 steps marked along the bottom, and various outputs up the side. By pressing a key in the matrix a control voltage appeared at the appropriate output when that particular step was activated. A clock signal was used to cycle round the steps, which triggered sounds depending on what buttons had been set.

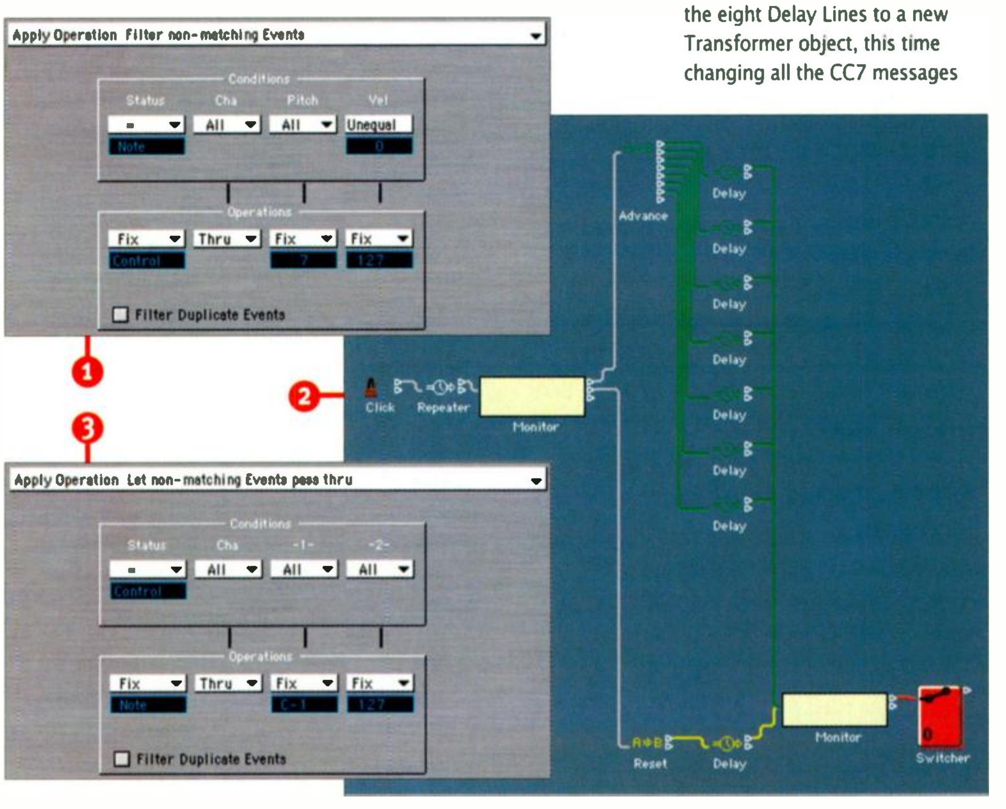
Whilst being a very simple design, it still has its devotees and has been revisited in various modern software applications, such as Propellerhead's *Reason*. However, *Logic* has no dedicated matrix sequencer and, while the Hyper Edit window can be adapted to act in this way, I prefer to use the Environment. So here's how to create a basic eight-step, two-line matrix for yourself.

Doing The Groundwork

First you need a signal to be automatically generated in the Environment during playback, and for this you need to use a MIDI Metronome Click object — with just the Bar field ticked in its Parameters it will mark the start of every bar. To get the matrix to run at a comfortable rate at 120bpm, it is necessary to go through the eight-step sequence twice. In order to create a second trigger halfway through the bar, cable the metronome straight into a Delay Line object — in its Parameters, set Repeat to one and Delay to '8 0', making sure Thru is ticked.

To route the MIDI Note On message to different parts of a matrix, a Channel Switcher object needs to be used — this object is vital to the matrix sequencer design, because it can operate under remote control. First let's concentrate on setting up the automated switching action. A MIDI Continuous Controller number seven message (CC7) with a value of 127 can be used to advance the switch, whilst the same message with a value of zero will reset it back to the start. To convert the MIDI Metronome Click's note messages to the correct Continuous Controller messages, you need to use Transformer objects.

Create two new Transformers, and cable the output of the Delay Line to them both through a Monitor object. Double-click the first Transformer to open its dialogue box and change all note events with velocity values above zero into CC7 messages with values of 127. A setting for doing this can be seen in Screen 1 — note that the Filter Non-matching Events option is selected at the top of the window. The other Transformer should be set up the same, except with the value of the resultant CC7 message at zero.



Having the right signals is one thing, but we also need to duplicate them — we currently have two per bar, yet for an eight-step sequence repeated twice we need 16 separate signals. Cable the output of the 'switch advance' Transformer into seven Delay Line objects, unticking Thru in each one's Parameters and setting the Delays from '1 0' through to '7 0'. Next, cable the 'reset switch' Transformer into another Delay Line with Thru unticked in its Parameters and Delay set to '8 0'. Finally, cable all eight Delay Lines through a Monitor object to merge them, and connect this to a new Cable Switcher. When we have some outputs for this Cable Switcher, it should happily cycle round eight steps. This setup can be seen in Screen 2.

Creating The Notes

That provides the switching control, so now we need to send a signal though the Cable Switcher. Take a second output from the Monitor object to route the eight Delay Lines to a new Transformer object, this time changing all the CC7 messages (both 'advance' and 'reset') back into Note On messages with velocity values of 127 — a setting to do this can be seen in Screen 3. Cable this Transformer to the Cable Switcher — these Note On messages will be used to activate a row of buttons which will form the steps of the matrix.

To deactivate the buttons after the end of each step, however, we need another Note On message, but with a velocity value of zero. Take another feed from the Transformer that converts the CC7 messages into Note On messages, and use yet another Transformer to change the velocity values to zero. Cable the output of this into another Delay Line object, setting the delay to be the length of time you want each matrix step's button to remain on (bearing in mind that this should be shorter than the step duration) and then connect that into the Cable Switcher as well. This setup is illustrated in

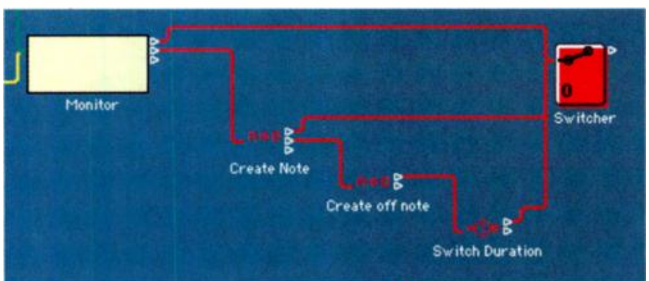
Current Versions

- Mac OS X: *Logic Audio Platinum* v6.1.0
- Mac OS 9: *Logic Audio Platinum* v6.1.0
- PC: *Logic Audio Platinum* v5.5.1

▶ Screen 4.

It's worth noting here that a strange quirk of the Transformer object means that the zero-velocity Transformer will actually output velocities of one. This is fine for switching our buttons, which will default to sending out notes of a fixed duration. To get a variable duration for the notes requires some complicated routing, which is beyond the scope of this article. However, the note durations can be easily changed in the Event List window once the data has been created.

So, let's recap. We now have a Cable Switcher which sends a Note On message with a velocity value of 127 followed by a Note On message at a velocity of zero via eight different outputs which are switched 16 times a bar. The hard work is now complete, because the signals have been created, much like the control-voltages of old. What we need to do now is decide whether or not to send each one to our instrument.



4

Building The Selection Matrix

Create a horizontal row of eight buttons by selecting Button from the Fader submenu of the New menu, and set the In field in the Parameters to match the notes being sent by the Cable Switcher. Cable eight outputs of the Cable Switcher into each of these buttons, one per button. With the Range fields in the Parameters set to one and 127, the buttons should switch on and off appropriately, providing a visual representation of which step the matrix sequencer is on. Bear in mind that *Logic* gets the MIDI data right first and updates the screen second, so the visual switching may lag slightly.

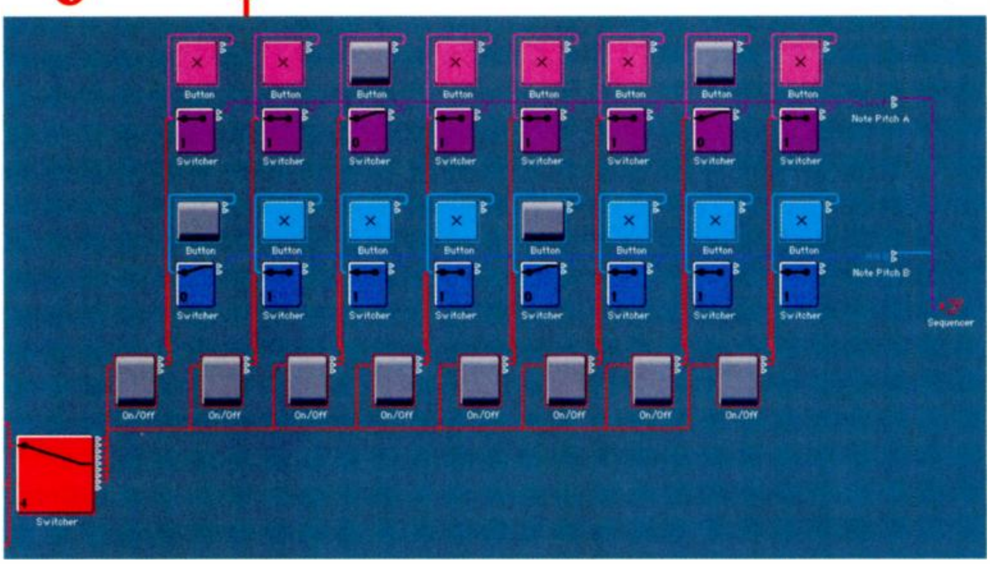
These buttons are what actually creates the data we'll send to the sequencer, so choose what kind of data you want in the Parameters box for the button — for the example here we'll use Note On messages. For each line of the matrix you want to create, just take the output of each button into its own new Cable Switcher. Control the Cable Switcher with another simple selection button — this one will decide if the Note On is sent for that step, just like the switches on an analogue matrix sequencer. Take one output from all the Cable Switchers in the row and cable them to a single Transformer to shift the Note On messages to any note number you like. The buttons will also generate notes with a velocity value of one, so use the Transformer's Filter Non-matching Events option to get rid of these.

Once the second row of buttons is constructed in the same way, all that remains to be done is to send from each row's Transformer output to a single Sequencer Input object, from where it can be routed to instruments and recorded. This last section of the setup is shown in Screen 5, but if you're still scratching your head about how it all works, you can investigate the final setup for yourself by downloading the demo Song file from the electronic version of this article at www.sound-on-sound.com.

Beyond The Matrix

The example created here is just a simple two-line affair. However, this was just to get you started. Using the data created by the main switcher, any number of rows could be created. If you want to make the screen a little tidier, move all of the unwanted objects out of the way to leave just your selection buttons. The screen can be made to look even tidier by turning off Cables in the Environment window's View menu. The parts you'll create with the matrix sequencer are unlikely to resemble anything you might play, just due to the method of programming, and may just take you off in a new musical direction. Now, where did I leave that Jean-Michel Jarre album... SOS

5



Logic Tips

The button with the chain links icon at the top left of most of *Logic*'s windows has three modes, indicated by the colour of the button: off (gray), Link (pink), and Contents Link (gold). In Link mode, the window's view will change in synchronisation with the top window. In Contents Link mode, its view will change in synchronisation with the selected object in the top window. Link mode is handy for windows that display similar objects, such as two MIDI editors. Contents Link mode is handy for synchronising MIDI editors to the Arrange window. The Sample Editor window is an exception — it only offers Link mode, which causes it to show the contents of the audio region selected in any other window. *Len Sasso*

Some of *Logic*'s plug-ins have so many parameters that the designers have chosen not to show all of them in the plug-in window's main Editor view. However, if you'd like to see them, then click the circular '001 011' button at the top of the window and they'll appear under the main Editor view. *Mike Senior*

Audio playback from both the Audio window and the Sample Editor uses the Track Audio object selected in the Cha numerical field at the left-hand side of the window. It's a good idea to set this to an Audio Track object not used in the Arrange window, so that muting and effects processing won't be applied when auditioning audio files. Note that the setting is not preserved with the song or between sessions with *Logic* — it is always reset to one. *Len Sasso*

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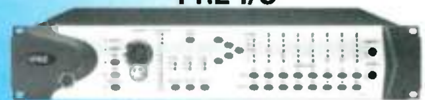

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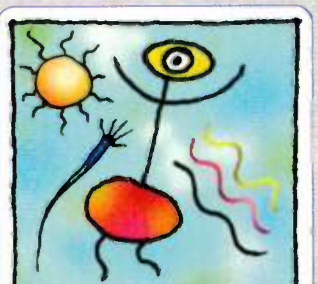

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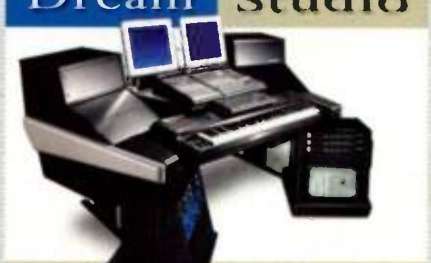

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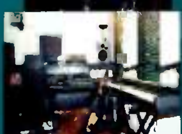

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JEN SYNX 508 analogue polysynth keyboard, five octaves, built-in effects, very rare, one knob missing, offers ☎ 020 8516 9018 (London)

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KORG TRITON RACK, immaculate condition, six months old, box and manuals fantastic synth/sampler but no longer needed, price includes UK postage, £700 ☎ Mike 01453 547111 (Glos)

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KORG Z1 MOSS synth, 13 oscillator models, 61 keys, 12 note polyphony, 256 patches, 32 multis, X-Y pad, £550 ☎ 078 5573 7591 (Colchester, Essex)

KURZWEIL K2000R version 3.87, fully loaded with all options, for sale or swaps, £750, Roland JD800 very good condition for sale or swaps, £550 ☎ 01865 776893 (Oxford)

NOVATION KS5, as new, manuals, original box and custom case, £599 ☎ Kevin 01883 331237 (Caterham, Surrey)

NOVATION NOVA II Keyboard, excellent condition, hardly used, home use only, boxed with manuals, £700 ono ☎ Steve Horsfall 077 1584 1004 (Manchester)

NOVATION SUPERNOVA 2 rack unit, boxed as new with manuals, only six months old, hardly used, £600. ☎ 079 5687 1289/020 7173 1334 (London)

ROLAND JX305 groove synth, has all of the features of the MC505 groovebox with even more sounds and a five octave keyboard, £300 ☎ Rhys 01686 670539 (Newtown, Powys)

ROLAND SH32 synth module, one month old, home use only, MIDI and audio cables included, £240 ☎ 01234 343343 (Bedford)

ROLAND U220 module, in good condition, manual, £85 ☎ Paul 078 0267 3278 (Cambridge)

YAMAHA CS1X, mint condition, manual, power source, boxed, £175 ono. ☎ Matt 01926 339469 (Leamington Spa)

YAMAHA MU100R sound module, 64-note poly plus VL (modelling) and VH (harmony) cards, 1U rack mounting, boxed as new, manuals, £270 ono ☎ 0117 9775747 (Bristol)

YAMAHA P200, highly acclaimed stage piano, superb condition, box and instructions ☎ 01986 893701 (Bungay, Suffolk)

YAMAHA TX7 rack version of the classic DX7 synth, £75 ☎ Rob 079 5837 2021 (London)

recording

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BELLARI RP533 multi-processor compressor, exciter, valve preamp, perfect for vocals and guitars, £140 ☎ Rob 079 5837 2021 (London)

DBX QUANTUM mulit-band finisher, compressor, gate, limiter, de-esser, brilliant EQ, £750 ono ☎ 077 3671 8523 (London)

DIGITECH 2120 Artist valve preamp processor, comes with Controller One floorboard providing two expression pedals, box, manual, mint condition, £380 ☎ 020 8680 9099 (South Croydon, Surrey)

FOCUSRITE PLATINUM Voicemaster, professional vocal channel strip, mint condition, boxed, manual, £200 ☎ Alex 079 4647 4132 (Bromley, Kent)

FOSTEX D160 digital multitrack, 16 tracks of uncompressed audio, 4GB hard disk, eight virtual tracks, MIDI sync, ADAT, SCSI, hardly used, complete with box and manual, £625 ☎ Dave 01384 573169 (Brierley Hill, West Midlands)

FOSTEX G16S 16-track half-inch reel-to-reel, Dolby S noise reduction, removable front control panel for use as a remote, home use only, complete with manual, £995 ☎ Paul 020 8691 0120 (London)

OBERHEIM OB3 Squared Hammond module, perfect condition, manual, £195, roland JV880 with six expansion cards, £250 ☎ Mike 079 6184 2639/079 6184 2639 (Southend)

REVOX C278, £650, PR99, £750, Studer A80VU, £950, B67VU, £850, A807, £700, A810, £750, 24 8 2 desk, £4500, Telefunken M15A, £500; A710, £400, Beyer M88, £70 ☎ 01246 275479 (Derbyshire)

ROLAND JUNO 106, very good condition, £350, Oberheim Matrix 1000, very good condition, £250, Akai S1000, 24MB with Zip drive and digital board, £170, Fostex D90, ADAT out, extra caddy, £250. ☎ 077 7913 0361 (London)

ROLAND VS840 digital multitracker, superb condition, boxed, excellent condition, excellent tool for studio work, £350 ☎ Tom 079 5616 0884 or email t_omni@hotmail.com (London)

ROLAND VS880EX, 20-bit eight-track, unused, flight case, manuals, £395, Marantz CDR770 CD recorder, boxed, leads, manuals, £185, both together for £495 ☎ 079 6184 2639 (Southend)

SOUNDCRAFT SPIRIT Studio 16 8 2, good condition, great sound, power supply, manual, included, can demo, £250 ☎ Tim 079 6827 3954 (Surrey)

STUDIOMASTER P7, 24 8 2 MIDI mutes mixer, immaculate, manual, PSU, £700 ono; Akai S950 expanded sampler, a classic, 11 outs, box, manual, offers, Alesis Monitor 2 speakers with Servo 260 amp, excellent, £450 ono for both ☎ Christian 079 3048 1370 smallworldsound@hotmail.com (Middlesex)

TASCAM 424 MkIII, boxed with PDF manual, £150; Roland PC-180A MIDI keyboard, boxed with Steinberg *Cubasis AV*, £50 ☎ Colin 079 7124 3714 or email wyrdsister666@hotmail.com (London).

TASCAM DA20MkII, very good condition, boxed, can deliver, quick sale needed hence price, £90 ☎ Jamie 079 0095 2748 (Bristol)

TASCAM DA98HR, originally purchased for pro studio in 2002, 40 hours use, mint condition, extra 30 tapes for £100 and TDIF cable for £50, otherwise £3500 ☎ Darko 077 8066 1355 (London)

drum machines

AKAI XR10 drum machine, in mint condition, great sounds, £140 ☎ 01492 572614 (Conwy)

ROLAND TR909, fully working with manual priced for quick sale, £500 ☎ Mikeb 079 0548 3513 (Bangor, Gwynedd)

ZOOM RT323, 128 drum kits, 55 bass programs, Smartmedia slot, four outs, two foot switch inputs, MIDI in/out, boxed adapter, manual, £130 ☎ Gilbert 079 4740 6799 (London)

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sequencers

ALESIS MMT8 eight-track sequencer, good condition, manual, £50. ☎ Paul 078 0267 3278 (Cambridge).

samplers

AKAI S900, mint condition, boxed, manual, 10 outputs, nice filter, sounds wicked, can post, £100. ☎ George 078 1306 6005 (Derby, Midlands).

AKAI S950 digital sampler, MIDI, 12 bit, good condition, £150. ☎ 02476 543432 (Warwickshire).

AKAI S1000 with 10MB memory, comes with CD-ROM and hard drives, sensible offers please, have to sell due to moving, offers. ☎ Richard 01869 247541 (Bicester, Oxon).

AKAI S2000, two for sale, both with 32MB, manuals and boot disk, one has an eight output board, £200 each ono or swap for Roland AX7. ☎ Mark 079 0191 3637 (Warminster, Wiltshire).

AKAI S3000XL boxed, 32 bit, Zip drive, £350. ☎ Stefan 079 7029 8544 (London).

AKAI S5000 USB card, 72MB, Zip 250, CD reader for sample CDs, manual and all leads, excellent condition, £750. ☎ 079 3913 9440 (Surrey).

EMU E4 ULTRA, eight extra outs on expansion board, 18GB internal SCSI drive, CD-ROM drive, manual, in mint condition, one careful owner, £1200. ☎ Haris 077 9868 7566 (London).

EMU E5000 sampler, mint condition, virtually unused, £350. ☎ Jonathan 01273 721551 or email jon_breeze@hotmail.com (Brighton).

EMU ESi4000 sampler, hardly used, 128MB memory, CD-ROM drive and SCSI cable included, £300 ono. ☎ 020 7738 9581 or email richardamos@aol.com (London).

EMU ESi4000 Turbo, mint condition, boxed with manual, 32MB RAM, 24-bit effects board, eight outputs, SCSI, wicked filters, can post, £250. ☎ George 078 1306 6005 (Derby, Midlands).

ROLAND VP9000, 24MB RAM, six outputs, digital, SCSI, excellent condition, 18 months old, hardly used, boxed with manual and V-Producer software, £630. ☎ Ian 077 1038 1498 (Leeds, West Yorks).

personnel

BASS PLAYER REQUIRED in London, must be experieced and able to play tight, should be a five string player into death metal, ambient dance, electronica and de-tuning, should be age 18 to 23, influences: Radiohead, Korn, Sepultura, Squarepusher. ☎ Luca 079 3262 6041 (London).

COLLABORATOR REQUIRED by a bassist, bored with same old stuff, using Steinberg *Cubase*, Native Instruments *Reaktor*, Ableton *Live* and similar, seeking collaborator for electronic/live gigs and recording original music. ☎ 01207 560981 (Newcastle Upon Tyne).

COLLABORATION WANTED, guitarist/producer/engineer with automated Pro Tools studio, looking for ambitious musicians, bands and song writers to collaborate with. ☎ Chris 01604 879303 (Northampton).

DRUMMER AND BASSIST urgently required for an album and possible gigs. ☎ Sean 0114 2398409 (Sheffield, Yorkshire).

JAZZ/METAL DRUMMER REQUIRED to form a band in London, double kick and jazz ability essential, death metal, ambient, trance/electronic, must be aged 18 to 23 and into Sepultura, Opeth and Radiohead. ☎ Luca 079 3262 6041 (London).

KEYBOARD/EFFECTS player required to form a band in London, combining death metal, ambient dance, and trance, synths, effects and musical ability are essential, must be age 18-23 and into Radiohead, Sepultura, Squarepusher and Opeth. ☎ Luca 079 3262 6041 (London).

SYNTH PLAYER required for electronica-influenced band, should have a vision for analogue sounds and atmospheric textures, must be aged 21 to 30. ☎ Jake 020 7538 2073 (London).

miscellaneous

DOEPFER REGELWERK, excellent condition, boxed, 24 faders and 24 buttons plus eight-track, 16-step analogue-style sequencer, a bit like a Kenton Control Freak but better, £350 ono. ☎ Stu 079 1927 4253 (Worcester).

KORG SRC512 Memory Card, hardly used, in original case, £30. ☎ 020 8848 6572 (Crawley, West Sussex).

ROLAND GI10 guitar MIDI interface, half-rack size, mint, box, manual, £200; Nady 201 dual diversity guitar transmitter, £50. ☎ 020 8680 9099 (South Croydon, Surrey).

computers/software

APPLE B&W G3 350MHz, 512MB RAM, 17-inch Apple B&W monitor, 9GB SCSI drive, dual SCSI Atto card, mint condition, one careful owner, £600. ☎ Haris 077 9868 7566 (London).

APPLE MAC G4 400, Emagic *Logic 4.5 Platinum*, Digidesign Digi 001, Fatar 88 keyboard, *Pro 52* soft synth, Emagic *EXS24* sampler, Yamaha NS10s, servo 170 amp, £2500 ono; £5500, can split. ☎ Martin 077 2936 8949 (West Midlands).

APPLE G4 TOWER, 400MHz, 768MB RAM, two internal hard disk drives for 40 and 60GB, running OS9 and OSX, 17-inch studio monitor, includes Korg 1212 I/O soundcard and various music software, £700. ☎ 079 5602 8644 (Sawbridgeworth, Hertfordshire).

ATARI 1040 STE with 4MB upgrade, Panasonic KXP1123 printer, manuals, leads and disks, in excellent condition, £50. ☎ Lee Archer 01942 714268 (Wigan, Lancashire).

DIGIDESIGN 888/16 interface for Pro Tools, £380. ☎ 020 8340 8425 (London).

EDIROL UA5 USB audio interface, 24 bit, 96kHz, mint condition, three month old, light home use only, perfect for small, portable laptop set ups, wide range of inputs and outputs, £180. ☎ Rosh 077 7920 9639 (Sandhurst, Berks).

FIREWIRE PC, 2.2GHz processor, 256MB memory, 60GB 7200RPM disk, CD-RW, AGP sound, three Firewire ports, Ethernet port, USB 56K modem, quiet PSU, 17-inch monitor, £425. ☎ Boris 079 7033 5616 or email boris.sedacca@btopenworld.com (Surrey).

LACIE CD player, compact stand-alone unit, SCSI, includes software, new, £65. ☎ 01492 572614 (Conwy).

MOTU 828 Firewire interface, eight analogue ins and outs, eight ADAT ins and outs, S/PDIF I/O, ADAT sync, 48V mic amps, perfect condition, £600 ono. ☎ 078 9085 3835 (London).

TERRATEC EWS88D professional PCI audio interface, three milliseconds latency, S/PDIF, ADAT, Directsound, boxed as new, £120. ☎ Alex 079 4647 4132 (Bromley, Kent).

YAMAHA CCD8-TDII TDIF interface card for 03D or 02R, perfect condition, sale due to upgrade, £180. ☎ 079 0527 5153 (London).

wanted

FENDER RHODES or Wurlitzer 200 wanted, will swap my Bentley compact piano, it is a modern upright/overstrung and has a warm, convincing tone, in excellent condition, cost £3000, photos available. ☎ 01745 833317 (Abergele, Conwy).

MIRO DC30 video capture card wanted, cash paid. ☎ Rich 01270 883779 (Stoke-On-Trent, Staffs).

ROLAND AX7 or similar guitar axe, can swap for cash or two Akai S2000s. ☎ Mark 079 0191 3637 (Warminster, Wiltshire).

ROLAND D50 wanted for parts, should be totally dead although I do want one with the pitch bend/modulation intact. ☎ 01284 762672 (Bury St Edmunds, Suffolk).

SOUNDCRAFT 2U, 19-inch rackmount 'medium' PSU. ☎ 01429 866304 (Hartlepool).

TASCAM AF8 wanted, not the AE8HR because it is the sample-rate convertion which is needed, all offers considered. ☎ Paul (daytime) 01903 286446 or email pvernon@feba.org.uk (West Sussex).

TL AUDIO 5050 preamp wanted, I can only buy it from the London area, unfortunately. ☎ 079 4727 9913 (London).

We can argue about copyright ownership until the cows come home, but the real issue is not who owns it, but who played it...

Sean Holden

In his Sounding Off in *SOS* October 2002, Chris Eccles raised the question of who can be said to own a drum loop or rhythm once it has been sampled, processed and re-used in a different context. Though I agree with much of what he said, his assertion that nobody really owns a rhythm is at the very least an over-simplification. Sure, some rhythms are so ubiquitous that nobody could be said to own them, but not everyone plays those rhythms in the same way. Some people can play a simple rhythm with an artistry that's very difficult, if not impossible to mimic. The more I hear samples of drummers and other musicians used in recordings, the more important it seems that we remember the crucial role performance plays in recording.

Recently, I was brought down to earth with a bump. After 20 years of playing rock drums I decided it would be fun to learn to play jazz. Obviously nobody could be said to own the classic 'spang-spang-a-lang' swing rhythm, but have you ever tried to play it? Or to reproduce it with a sequencer? It takes very little skill to play or program it in the basic sense of the right sequence of hits, but to make it really swing requires something that can't be written down, so much so that nobody ever even tries to. Look at a drum chart for a jazz tune and you'll see a very rough approximation of the actual part. The writer and the drummer both know it needs to swing, but that's something far too subtle to notate. And some drummers are a great deal better at playing it than others.

The same argument can be made for whatever style you care to mention. Lots of rock drummers can find their way around a kit, but very few can play it and make a sound like John Bonham. And how many of those that can will end up finding such an individual and creative voice in their own work? We can't all be geniuses, but if I have a problem with sampling it is this. Very often (as always there are notable exceptions) I have the impression that the work of some of the most talented musicians of the last century is being exploited by people unwilling to expend the same kind of time and energy required to find their own voice. You can often hear genius at work in a good sample, but that genius isn't simply going to rub off.

A really good funk drummer can make you get up and dance. Why is it, after all, that when you see a funk act appearing live they'll have a seriously hot drummer? It's certainly not because paying a pro and lugging around a drum kit (which then has to be miked up) is easier than using a drum machine or the backing track from the album. Why is it that a relatively small number of session drummers seem to have cornered a disproportionately large slice of the market? These guys can sit in the background, just keeping time, and make it feel good in an indefinable way. That takes artistry. A jazz drummer of the calibre of, say, Tony Williams could make a ride cymbal whisper in your ear, and though you can't describe what it's saying, you know that it's something very important.

If you like to work using sequenced rhythms, here is an exercise for you. Get yourself a copy of Miles Davis' *Nefertiti*, a sequencer and a decent bunch of ride-cymbal samples. See if you can reproduce a few bars of Tony's ride-cymbal playing so that it sounds identical. You may be surprised at the complexity of what he does. You might also be surprised at what a real drummer can bring to your work.

There's no doubt that in some contexts a sequenced rhythm is going to do the job fine. But there's so much more to rhythm than that. You'll have noticed that people generally choose to sample the top players because their playing has the best feel. This implicit acceptance of the complexity and skill required to produce rhythm speaks volumes. You might argue that your music is sufficiently fast, complex or revolutionary that a person won't be able to play it. But try listening to a bunch of people playing some of the more rhythmically complex pieces by Steve Reich and then ask yourself how hard it would be to sequence what they can play.

What I am essentially arguing in favour of here is musicianship. This has almost become an unfashionable concept in certain circles, but the world is full of musicians with taste who play their instrument with passion and who genuinely want to expand what it's capable of. Working with other musicians might bring its own problems; I am more aware than most of how much hard work it is to move a drum kit around. But find yourself the next Tony Williams, Billy Cobham, or John Bonham and I think you'll find you never want to use a sample again. SOS

About The Author

Sean Holden is a drummer, mandocellist, songwriter and lecturer. He plays rock with progressive goths Bunty, jazz with anyone who's up for a bit of noisy be-bop, and writes and performs modern English folk music under the name of Dodman.

If you'd like to air your views in this column, please send your ideas to: Sounding Off, *Sound On Sound*, Media House, Trafalgar Way, Bar Hill, Cambridge, CB3 8SQ, UK. Any comments on the contents of previous columns are also welcome, and should be sent to the Editor at the same address.

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