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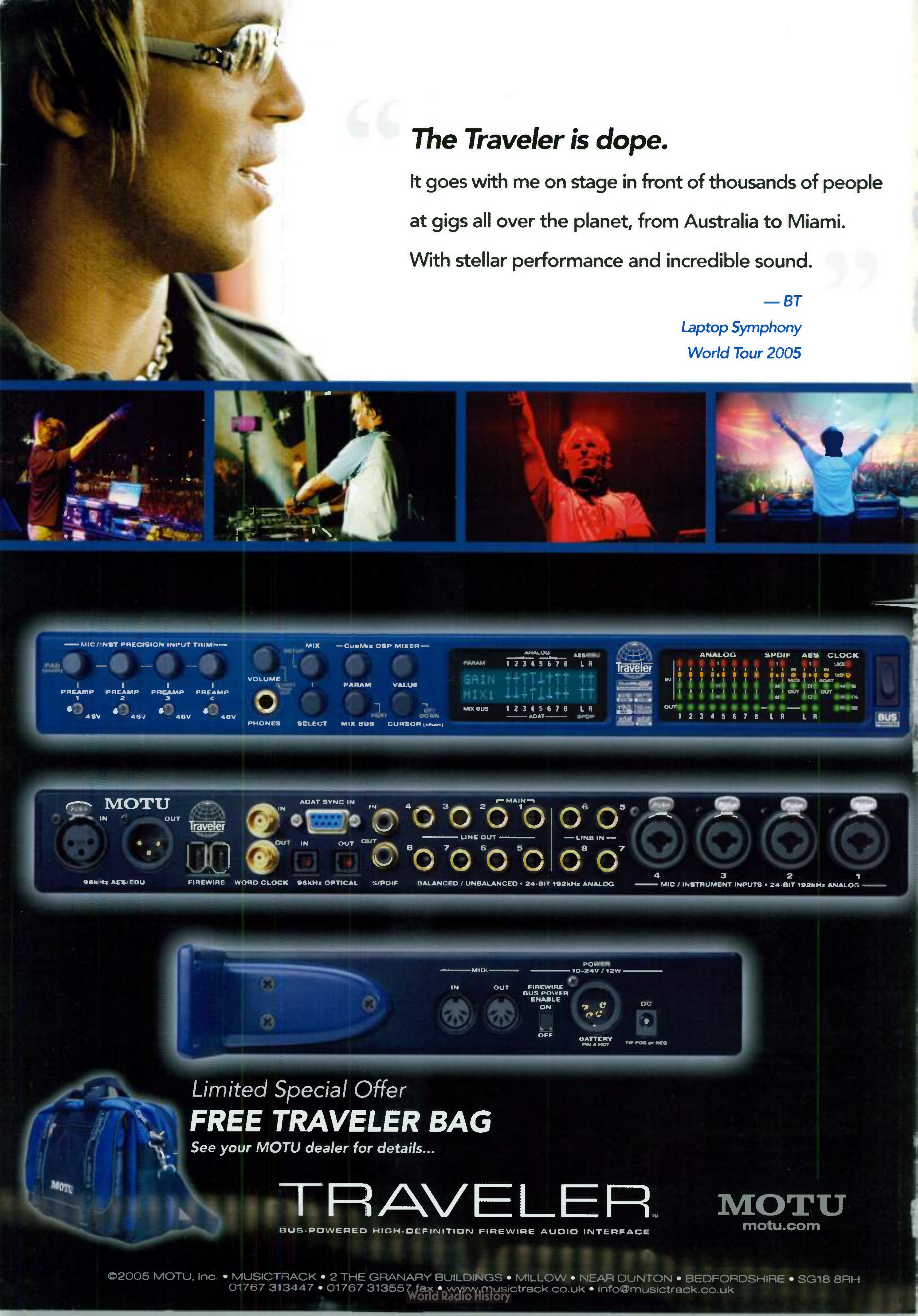
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Less Is More

Because of the distorted time field in which magazine production works, I'm writing this leader column immediately after the Christmas break, and once again the choice of TV programmes was so dismal that I decided to give the whole thing a miss and instead do a Studio SOS on my own studio. If you recall, last time I did this a huge amount of outboard gear and several kilometres of cable were pruned away, so you may wonder if there was anything left to cull! There was, and now the cupboard of eternal darkness has got so much gear in it that it could probably set up as a studio on its own!

As I hadn't used my cassette deck or either of my DAT machines within the past year these were the first to go — I can always fish them out if a client turns up with material recorded on some historic format, which is why my quarter-inch analogue two-track machine made the trip a couple of years previously. That allowed me to lose another patchbay, which in turn allowed me to lose a complete rack, thus gaining more valuable space. I also dismantled my surround system, as surround rapidly seems to be going the same way as Quad other than in movie production, but I've kept the speakers and stands nearby in case I need to set it up again in a hurry. The other thing I did was modify the mountings for my two computer monitors to lower them, as the upper shelf of my otherwise lovely Quik Lok workstation placed them so high up that I had permanent neck ache from looking at them. It is surprising just how much more pleasurable recording can be when you've sorted out the ergonomics!

Since Korg brought out the Legacy Collection, my two Wavestation SRs have been on extended leave, so now they've gone to join their friends in the cupboard, leaving my beloved Roland JV2080 as the only hardware synth in the village. This feeds directly into my audio interface, so the little mixer I was using to combine my keyboards has also gone. The result is far less knitting, far less complexity, more space and a generally better working environment. However, there's more to a studio than a table with gear on it — there's also the acoustics!

It doesn't matter how good your equipment is, you won't be able to do a good mix, let alone embark upon mastering, if you don't have adequate acoustic treatment and a decent pair of monitors. Last time I did a reshuffle I moved my studio system around by 90 degrees, and although I also moved some of the Primacoustic foam I'd fitted previously, I still felt more needed to be done. So, pausing only briefly to watch *Chicken Run* again (one of the true classics that was worth watching!), I built some large traps to go behind the monitors comprising acoustic foam on top, a Rockwool slab underneath and an air gap behind to give me better low-end absorption. My Mackie monitors have rear-facing passive radiators, and the wall behind them seems to need more attention than is the case for front-ported speakers.

A similar trap behind my listening position plus a couple of panels on the ceiling (and some self-administered Hob Nobs) completed the makeover and the improvement in both bass tightness and stereo imaging was quite noticeable, even though the room had been reasonably well behaved before. This proved to be quite timely because I needed to check out some mastering comparisons which you'll be able to read about in this issue. The difference between mastering jobs done by different engineers can be very subtle, so your monitoring has to be pretty accurate to make any sense of the results. If nothing else, this feature proves that mastering is not for the faint-hearted or the heavy-handed!

Paul White *Editor In Chief*

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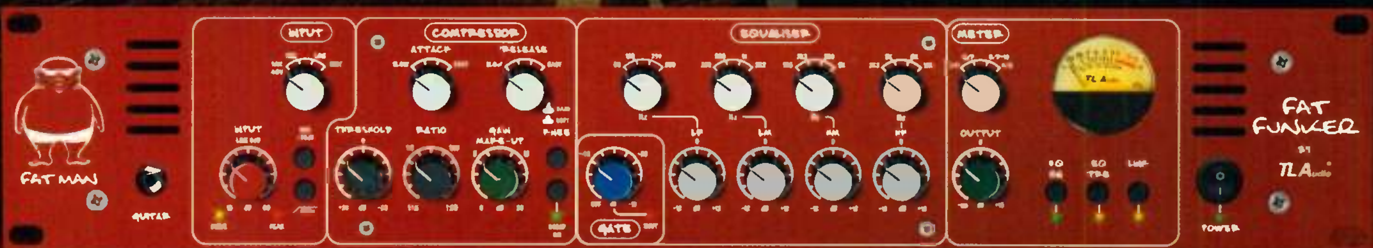

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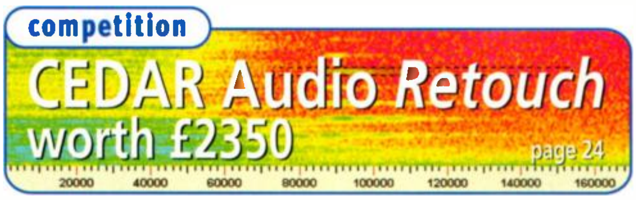

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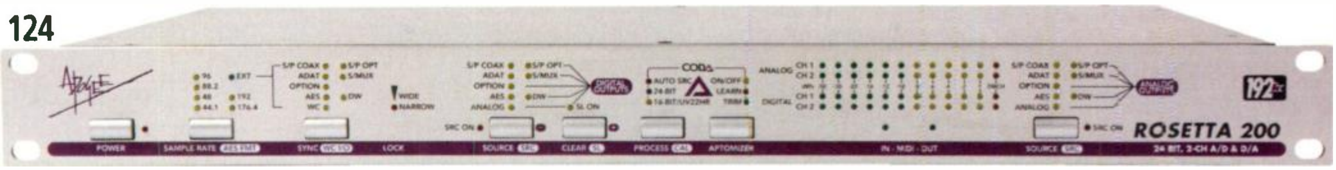

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Howard Jones

In this exclusive SOS video interview, '80s synth icon Howard Jones and collaborator Robbie Bronnimann discuss their approach to recording Howard's new electronic album, their passion for software synths, musical influences, and performing live with music technology.

Line 6 Video Interview

Company founders Marcus Ryle and Michel Doidic explain how Line 6 came to be and their product design philosophy, while the Sound Design team reveal their approach to modelling technology.

Max/MSP Programming

Is it a VST plug-in development environment, software-based Advanced Modular synthesizer or an Audio/MIDI performance environment? Find out more about Cycling '74's powerful sound creation toolkit...

Product Zone - 16 new products

- **Access Virus TI**
- **Antares Avox**
- **Creamware Minimax**
- **Korg Legacy Digital Edition**
- **M-Audio Project I/O**
- **Sonalog GypsyMIDI**
- **TC-Helicon VoicePro**
- **UA Dimension D**
- **Alesis Fusion 6HD**
- **Clavia Nord Stage 88**
- **Dave Smith PolyEvolver**
- **Korg D3200**
- **NI Guitar Rig 2**
- **Steinberg Hypersonic 2**
- **Tube-Tech MEC1/MMC1A**
- **Yamaha AW2400**

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Korg Legacy Collection DIGITAL EDITION

In this 16-minute video, record producer Steve Levine (Culture Club, The Honeyz) guides you through the virtual features and sounds of these software versions of two Korg digital synth classics — the M1 and WAVESTATION.

SOS Mix Rescue

Hear the remixes and learn ways to improve your own mixing skills.

Using Reverb

Watch the video as SOS Editor In Chief Paul White applies three different types of vocal reverb.

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Online Mastering Shootout!

We gave 3 different songs to 3 online mastering houses. Which one did the best job? Vote now...

Studio SOS

In this video workshop, Hugh Robjohns and Paul White build a DIY Vocal Booth for reader Paul Joyner.

Readers Tracks: 1.5 Hours of Unsigned Music

Use the SOS Music Player to audition this disc's compilation of unsigned music composed by *Sound On Sound* readers.

SOS Photo Zoom

Use the SOS Photo Zoom tool to get up close and personal with recent new products like the Clavia Nord Stage 88.

Recommended Computer Specification

WINDOWS
- Pentium 4 1.5GHz or higher
- 512MB RAM
- Windows XP Pro, XP Home
- DVD-ROM drive
- Sound playback facility
- Display 1024x768 or higher with Colour depth set to Highest (32-bit)
- Web Browser: Internet Explorer 6, Firefox 1.0, Netscape 7.0
- Macromedia Flash Player 7
- Apple QuickTime version 7
- Internet connection

MACINTOSH
- Apple Mac G4 or higher
- 512MB RAM
- OS X 10.3.9 or newer
- DVD-ROM drive
- Sound playback facility
- Display 1024x768 or higher with Colour depth set to Millions
- Web Browser: Safari, Internet Explorer 5.2.3 or newer, Firefox 1.0, Netscape 6.0
- Macromedia Flash Player v7
- Apple QuickTime version 7
- Internet connection

(OS 9.x not recommended but it may work on fast machines with plentiful RAM)

SOFTWARE PROBLEMS ?
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Please note: we can only provide basic advice on using the interface, and installing the software from disc. Unfortunately we cannot give in-depth help on applications provided on the disc by third-parties, or your hardware or operating system.

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Requirements

• FLASH PLAYER — This DVD-ROM uses Version 7 of the Macromedia Flash Player to present its media-rich content within your browser. If you do not have Flash Player v7 installed, you will be unable to watch video or hear audio playback.

If you have a previous version installed, you will need to upgrade to v7 (supplied on SOSDVD003).

• JAVASCRIPT — The DVD-ROM also requires that your browser has Javascript enabled.

• WEB CONNECTION — This disc is designed to link to the SOS web site to provide bonus background information to articles. Please disable popup blocking software to ensure that you see these related features (or make www.soundonsound.com a "Trusted Site" in your browser).

Roland roll back the years

It was a typically busy NAMM show for Roland, who had stacks of new gear to unveil. Following on from last year's Juno-D comes the **Juno-G**. Despite it's resemblance to the Junos of old, the new instrument is essentially a spin-off from the popular Roland Fantom X. According to Roland, the 'Juno' tag indicates that the instrument is a basic or economy model, but in the case of the Juno G, you still get a lot for your money. The Juno G features 128-voice polyphony, a 16-part sequencer and four tracks of audio recording with tape-style transport controls. An SRX slot allows for expanding the internal sound set and there's an arpeggiator, an infrared D-Beam controller and integral multi-effects. There's a USB port used for MIDI communication and for audio file data import or export as well as a PC card slot for Compact Flash and Smart Media memory cards. The Juno G also comes with Cakewalk *Sonar LE* and some patch-editing software.

Also pitched at the affordable end of the market is the new 49-key **SH201** synth, offering modelled analogue sounds and an audio input for sound processing. There are built-in effects, an arpeggiator and a D-Beam, and though the synth has plenty of controls for hands-on editing, it also shows up as a VST Instrument within your sequencer via USB (as seems to be the fashion these days) allowing the editing and saving of settings within your project or song. The SH201's audio engine provides two analogue-style oscillators complete with Roland's Supersaw waveform, a resonant filter and two LFOs. The additional Audio Filter section treats external audio. We liked the no-nonsense layout of the SH201 and the convenience of its plug-in functionality.

Rounding out Roland's new synth line-up, the **VP550** 'Vocal & Ensemble' keyboard employs vocal modelling technology and allows the user to sing into a mic while playing the keyboard to generate the sound of choirs, pop and jazz voices, male or female ensembles and vintage vocoder effects. The control panel appears very user-friendly and the demos were very impressive, with clear vocal articulation.

Also new for NAMM was the **MC808**, a sampling groovebox with motorised faders — an unusual feature for a groovebox, but very welcome — based on sampling and sequencing technology from the more costly MC909. Other than a handful of favourites from the MC909, all of the MC808's internal sounds and patterns are new. New sounds can of course be sampled and edited and WAV and AIFF files can be imported from a computer via a USB port. A Mac/PC editor is included in the package.

For aspiring electronic percussionists, Roland's **Handsonic 10** (or HPD10) electronic drum pad is a pared-down version of the original Handsonic HPD15, with 400 new drum, percussion and melodic sounds, 20 on-board effects, new metronome and rhythm-training functions and a comprehensive set of backing rhythm patterns. In all there are 64 drum/percussion kits on board. Its control surface comprises a touch- and position-sensitive playing area divided into 10 zones, and a D-Beam sensor (seemingly obligatory on all new Roland products) can be used to provide extra real-time control or even to trigger sounds. There are also sockets to add optional kick and hi-hat trigger pedals.

Also on the drumming front, Roland have launched some new drum triggers for use with acoustic kits. There are three models in the **RT10 Series** — the RT10K for kick drums, RT10S for snare, and RT10T for toms. These are smaller than their predecessors and so easier to position. All feature new trigger-sensing technology for improved response and a reduction in false re-triggering.

Roland UK +44 (0)1792 515020
www.roland.co.uk

New address for Panic Music

Panic Music Services have moved to new premises in Swavesey, Cambridgeshire. The well-established UK company provide authorised repairs and service for a wide range of brands including Novation, Allen & Heath, Akai, Roland, Tascam, Mackie and Yamaha. Their new address is 9-11 High Street, Swavesey, Cambridge, CB4 5QU.

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

Gateway Training set up new facility

Gateway Professional Development Training have just established a training centre at Phoenix Sound's new state-of-the-art recording facility, located at the world-famous Pinewood Shepperton film studios. Gateway PDT will be providing training for producers, musicians, teachers, studio owners and admin staff. The first courses are set to begin in February and March.

Gateway PDT +44 (0)1869 340878
www.gatewaypdt.com

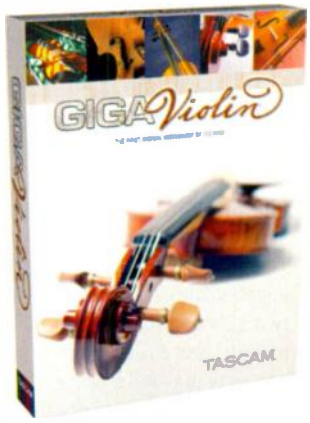
Sound On Sound at Sounds Expo

This year's Sounds Expo music technology show takes place on March the 9th, 10th and 11th. *Sound On Sound* is once again the seminar sponsor and this year's free seminar programme will be the most comprehensive so far. In addition to the regular hands-on workshops and demos run by manufacturers, there will be live sound and music business seminars, a surround-sound workshop and appearances by top

international artists sharing their expertise in music creation, performance and production. The new venue — London's Olympia 2 — affords new opportunities, including low-noise areas and a Chill Out Zone. Downstairs, away from the bustle of the main show floor, we'll be conducting our ever-popular Q&A sessions as well as seminars and live demonstrations of recording and production techniques. There will be seminars on recording guitars and vocals, understanding convolution and a real-time version of our popular new Mix Rescue column from the magazine! The Q&A panel and seminar speakers will include Editor In Chief Paul White, Technical Editor Hugh Robjohns, PC Columnist Martin Walker and other *SOS* staff members. The full programme and schedule will appear on the *Sound On Sound* and Sounds Expo web sites in advance of the show. There's no need to book for any of these events — just be there at the appropriate seminar room in time for the start. Entry to the show costs £10, but you'll get a half-price entry voucher when you register in advance at www.sounds-expo.co.uk, so register now and pay just £5 on the door. We look forward to seeing you there!

New virtual instrument wrapper from Tascam

Tascam announced a significant development for the *Gigastudio* platform at NAMM. For a couple of years, sample library manufacturers have been making the switch to virtual instruments, 'wrapping' their libraries in a software instrument front end (such as NI *Kompakt* and *Intakt*) that can then be used seamlessly within the user's sequencer of choice. Until now, there's been no such Giga-related sample-library wrapper from Tascam. That's all changed with the announcement of GVIs: **Giga Virtual Instruments**. GVIs will initially be capable of running as RTAS or VST Instruments, and, unlike *Gigastudio* itself, they will be usable on Macs as well as PCs. As with other instrument wrappers, the GVI front end will only be able to load samples from its own library, and not all *Gigastudio* editing features will be available, although it will be possible for the full version of *Gigastudio* to open and edit GVI-wrapped libraries. The first GVI will be a violin collection from Tascam themselves called ***Giga Violin***. This will take advantage of *Gigastudio*'s built-in convolution reverb, offering the user the ability to place the violin virtually in a number of real environments. It will also be possible to move the virtual mic position relative to your virtual violin. Tascam are clearly hoping more sample-library manufacturers will join them in working on libraries powered by the GVI wrapper, and prestigious orchestral library manufacturers Sonic Implants are already said to be working on something.



Tascam +44 (0)1923 438880
www.tascam.co.uk

Lexicon cater for the computer studio

Two new USB recording interfaces from Lexicon were unveiled at the NAMM show. Continuing the Greek theme established by the Omega, the new arrivals are called **Alpha** and **Lambda**. The half-rack Alpha is the more affordable of the two. It's buss-powered and has a built-in mic preamp and a high-impedance guitar input. The tabletop Lambda interface adds MIDI I/O and a second preamp. Both interfaces are packaged with *Cubase LE* and *Pantheon*, Lexicon's high-quality VST plug-in reverb. The Lexicon **MX400** also made its debut. A successor to the MX200, the hardware reverb that can be edited by computer-based software and treated as a plug-in, the four-in, four-out MX400 comprises two MX200 engines which may be used independently. It also has some additional, more complex reverb algorithms that can be accessed by using both engines working together. The MX400's four outputs also permit the use of the reverb for four-channel surround-sound mixes.

Harman Pro UK +44 (0)1707 668222
www.harmanprouk.com www.lexicon.com

Synths, software, keyboards and controllers...

New Korg gear at NAMM

Korg had a busy Winter NAMM show, unveiling a lot of new gear in all their main product areas. Top of the list for us is the **Radias** (pictured right), a new multi-synthesis product that's somewhere between a keyboard and a rack module. The Radias consists of a module that can be slotted into an optional keyboard frame, creating a distinctive-looking keyboard. The module part can be slid around on the frame, so you can place it in the middle of the keyboard, or have it off at one side, leaving a gap where a laptop could sit during a live performance, for example. The Radias is four-part multitimbral and offers sample- and wavetable-based synthesis, two-operator 'cross-modulation' (Korg's current term for FM synthesis) and also analogue modelling. All of these synthesis types are fully integrated into the two-oscillator voice structure, which allows you to mix and match different synthesis types in one patch. There are built-in multi-effects (up to nine simultaneously), a 16-band vocoder with an included headset mic, a six-pattern arpeggiator and two 32-step sequencers. And you get all of this for £999 for the basic module, and £129 on top of that for the optional keyboard frame. The Radias is due in the Spring.

There were plenty of other new synths at the Korg booth. The TR range of sample-based workstations, the replacement for the Triton LE announced in last month's *SOS*, was completed by the launch of the £1399 **TR88**, an 88-note weighted version, to accompany the TR76 and TR61. And there were further spin-offs from the Triton's Hyper Integrated sample-based synth engine in the form of the 61-note **X50** and the 25-note **Micro X**. These also offer 64MB of sounds each, plus the usual comprehensive set of built-in effects, dual arpeggiators, and assignable knobs. Common to both is a ClickPoint controller, as first seen on the D3200 workstation. But best of all is the level of computer integration offered by the two USB-equipped X-series synths. Cross-platform editor software is supplied with both, and can be operated as a VST or AU plug-in for seamless use and editing of the keyboards from within your sequencer of choice. The X50 will retail for £525, the Micro X for £469.

Talking of software, Korg's ***Legacy Collection Digital Edition*** was the recipient of a further tweak (taking it to version 1.1), whereby the software M1 reviewed in last month's *SOS* was widened to incorporate all the sounds from Korg's 1989 T-series workstations, plus the sound library from the original five sound cards in that series. The v1.1 update is free to registered users of *Legacy Collection Digital Edition* and can be downloaded at www.korguser.net.

Korg also introduced a simple-to-use eight-track digital recorder, the **D888**. Designed to look like and operate as an eight-channel mixer (with three-band EQ, effects sends and pan controls) as well as a multitrack recorder, the £549 D888 has a simple recording interface, incorporates a 40GB internal hard disk and built-in effects, and can record on up to eight tracks simultaneously, with eight virtual tracks available per recording track. It records in 16-bit/44.1kHz WAV

format, and can interface with computers via USB, allowing you to port the WAVs into your computer for you to work on them there.

Some new controllers were also unveiled. The affordable **K-series** USB MIDI keyboards are available in 25-, 49- and 61-note versions. The 25-note K25 costs just £65 and the K49 costs £99, while a price for the K61 has not yet been announced. All three feature ClickPoint controllers, a pair of assignable knobs and switches, an assignable slider and cross-platform editor software. All three K-series controllers come bundled with a cut-down version of *Legacy M1* from the *Legacy Collection Digital Edition*.

Equally affordable was the £139 **Padkontrol** (pictured above), a drum controller. In addition to its 16 velocity-sensitive pads, the Padkontrol offers an assignable X-Y controller (derived from the Kaoss Pad) which can be used to output flams and rolls. As with the K-series controllers, assignments can be edited from a computer via USB. Sounds from the likes of Toontrack (*Drums From Hell*), Propellerhead, Ableton, IK Multimedia and UVI are included.

Korg UK Brochure Line +44 (0)1908 857150
www.korg.co.uk

Ableton *Live 5* courses at Alchemea

Alchemea College of Audio Engineering are running training courses in the latest version of Ableton *Live*. Spread over six evenings or three Saturdays, the course is led by DJ and producer Danny McMillan. Class sizes are small and each student gets their own iMac G5 to work on. Alchemea also offer one-day crash courses in music technology to bring beginners up to speed, as well as short courses in *Logic*, *Pro Tools*, *Reason* and other popular pieces of software. They have recently become an Apple Authorised Training Centre and are running Apple-certified *Logic Pro* 101 courses.

Alchemea +44 (0)20 7359 3986
www.alchemea.com

Live sound facilities at Deep Blue Sound

Live sound teaching at Deep Blue Sound in Plymouth has received a boost thanks to sponsorship from mixing console manufacturer Midas and touring sound company RG Jones. Deep Blue have secured the long-term loan of two Midas consoles and have purchased a third and RG Jones have provided a 20kW PA rig. Live sound is currently a component of the BTEC National Diploma in Creative Sound Engineering & Music Technology, but Deep Blue, in partnership with Plymouth College of Further Education, are hoping to offer a dedicated live sound course later this year.

Deep Blue Sound +44 (0)1752 210801
www.deepbluesound.co.uk

Another gem from Focusrite

Focusrite have launched a 'bigger brother' to their Saffire audio interface, featuring the same high-quality Focusrite preamps, 24-bit/192kHz A-D/D-A conversion and software control features. However, unlike the compact desktop Saffire interface, the 1U rackmounting **Saffire Pro 26 I/O** can handle a maximum of 26 inputs and 26 outputs. In this version there are eight analogue inputs with Focusrite preamps, eight analogue outputs, and a further 18 channels of digital I/O (stereo S/PDIF plus 16 channels of ADAT). Two of the analogue inputs have variable impedance to match different microphone types and are equipped with insert points, and there are high-pass filters on every channel. The Saffire Pro 26 I/O can either be buss-powered or powered via an external PSU and comes with the *Saffire Control Pro* software. Akin to the *Saffire Control* software provided with the original Saffire, this allows the user to set up individual custom mixes for each analogue output and switch between tracking and playback modes. DSP within the unit allows for latency-free monitoring and onboard EQ, compression, reverb or amp modelling effects processing. These effects are also available as VST or Audio Units plug-ins within compatible DAW software packages. The UK retail price has been set at £499, which looks like good value to us.

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

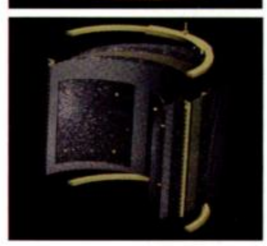
New mic, speakers and keyboards from M-Audio

At Winter NAMM M-Audio were showing their new tube mic, which aims to capture some of the magic of classics such as the AKG C12 and Neumann U47. The **Sputnik** offers cardioid, omni and figure-of-eight polar patterns, has a frequency response quoted as 20Hz to 20kHz and can handle SPLs of up to 142dB with the switchable -10dB pad. They've also introduced the **EX66** (pictured right), an active monitor with an unusual design featuring two separate low/mid-range drivers either side of a titanium-dome tweeter. It can accept both analogue and S/PDIF or AES digital inputs, and features onboard DSP to handle cabinet resonance tuning and crossover optimisation. An Acoustic Space control is said to optimise low-frequency response to match the speaker's position within your room. M-Audio unveiled plenty of new keyboards, too. The **Axiom** range (the Axiom 25 is pictured below) combines semi-weighted keys with eight drum trigger pads and eight rotary controllers, the trusty **Oxygen** range has been updated, and the new **Prokeys 88SX** is a stage piano 'so light that you can carry it under one arm'.

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

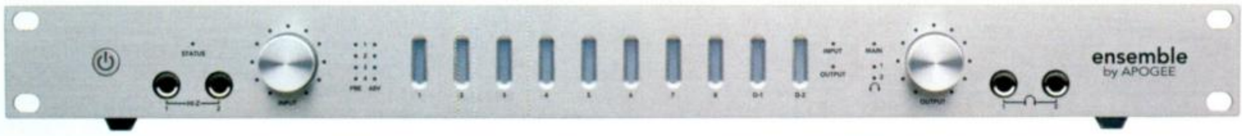
SE Electronics launch 'portable vocal booth'

Mic manufacturers SE Electronics have branched out into acoustic treatment with the new **Reflexion Filter**, a portable screen designed to reduce the amount of room ambience picked up by the mic. The screen is positioned directly behind the microphone so it can absorb and diffuse the sound from the vocalist (or instrumentalist) before it can reach the surrounding walls and surfaces and be reflected back into the mic. The Reflexion Filter is constructed from multiple layers of punched aluminium, wool, aluminium foil and aluminium foam and can be mounted on a normal mic stand. Early reports suggest that the device works very well and, at £229, it costs a good deal less than acoustically treating a whole room!



Sonic Distribution +44 (0)1582 470260.
www.sonic-distribution.com
www.seelectronics.com

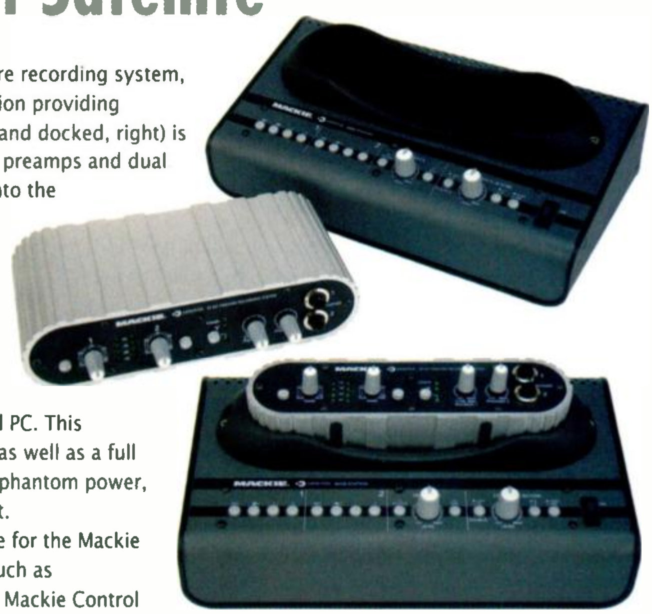
Apogee and Apple team up on new interface



There has been much speculation that Apple would at some stage introduce their own audio interface to accompany *Logic Pro*, and at NAMM, it was revealed that they have teamed up with Apogee to provide a high-end, rackmountable Firewire interface called the **Ensemble** that is 'fully integrated' into *Logic Pro*. Ensemble is a Core Audio (and therefore Mac-only) device which can handle up to 36 channels of simultaneous audio (18-in, 18-out). It offers eight channels of Apogee's A-D and D-A conversion, four digitally controlled mic preamps and extensive analogue and digital I/O. Ensemble incorporates Apogee's Soft Limit digital input limiter, UV22HR dithering and Intelliclock clocking technologies and the *Maestro* software controls mixing and routing within the unit. But the real selling point of Ensemble is that everything from mic pre gain to output level to sample- and bit-rate selection can be controlled from within *Logic Pro*. It costs £1195 and will be available at the end of February. The pricing is a little above what you'd expect to pay for a conventional interface but rather less than you'd expect for an Apogee product!

Sonic Distribution +44 (0)1582 470 260
www.sonic-distribution.com
www.apogeedigital.com

Mackie launch their own Satellite

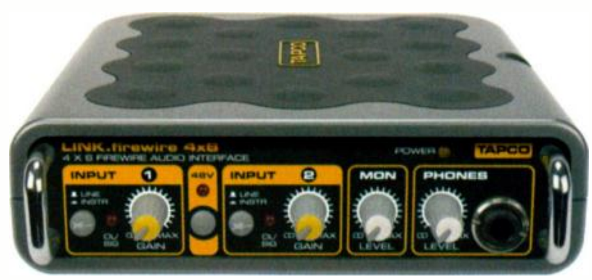


The Winter NAMM show saw Mackie launch their innovative **Satellite** Firewire recording system, which features a small portable audio interface that fits into a docking station providing monitoring controller-style features. The Satellite Pod (pictured un-docked and docked, right) is a portable 24-bit/96kHz two-channel Firewire interface incorporating two Onyx preamps and dual headphone/control room outputs with individual level controls. When docked into the Satellite Base Station, which provides AC power, additional I/O, talkback and monitor switching, the system becomes a two-input, six-output interface with a routing matrix, built-in talkback mic, comprehensive monitoring functions and surround speaker control. It's a very appealing concept — the Base Station stays permanently wired into your home studio setup and you can simply sling the Pod into your laptop bag if you need to record on the move.

Mackie's sister company Tapco also got in on the act with the 24-bit/96kHz **Link Firewire 4x6** audio interface, a four-input, six-output interface for Mac and PC. This buss-powered interface comes with Apple Core Audio, ASIO 2 and WDM drivers, as well as a full version of the aforementioned *Tracktion 2*. Its features two mic/line inputs with phantom power, balanced monitor outputs, S/PDIF I/O and a headphone output with source select.

Another exciting new product is the cross-platform ***C4 Commander*** software for the Mackie Control C4 control surface. This allows users to control hardware MIDI devices such as keyboards, sound modules, Line 6 Pods and effect processors directly from their Mackie Control C4. More than 180 profiles for a variety of instruments and effects units are included but users can also set up editors for their favourite hardware and soft synths via an intuitive drag-and-drop visual interface that maps parameters to the C4's knobs and displays.

Mackie announced the ***Tracktion* v2.1** update, which includes custom control surface creation facilities, MP3 import, MP3/OGG export and MIDI loop recording. There's also support for Frontier Design's Tranzport wireless DAW controller and various drag-and-drop file import improvements. Also new is a set of six mixing plug-ins that are free to download for registered users of the boxed version of *Tracktion 2*. The ***Mackie Mixing Tools*** bundle contains reverb and stereo panner plug-ins and mono and stereo compressor and EQ plug-ins. Both are available now, while Mackie's other show announcements aren't due until the spring.



Mackie UK +44 (0)1268 571212 www.mackie.com

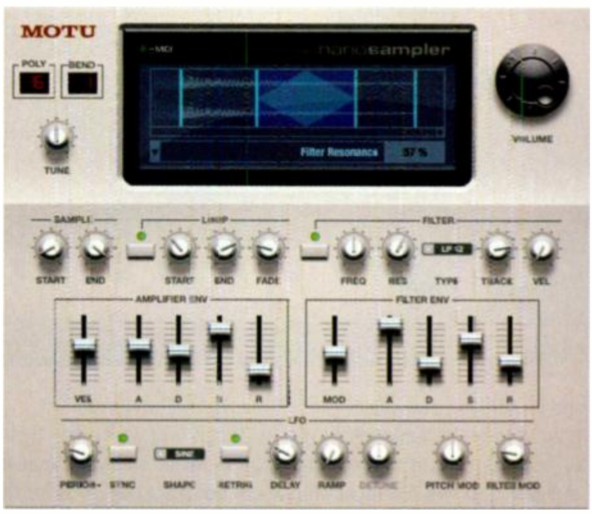
Hardware Hammond module from Creamware

Following on from their Minimax and Pro 12 ASB, Creamware unveiled a third member of their Authentic Sound Box range at NAMM. The **B4000 ASB** is a Hammond B3 module with a faithful complement of nine real drawbars. It's based on the *B3000* plug-in for Creamware's SCOPE DSP platform and claims to model many detailed aspects of the original electromechanical Hammond B3, right down to the effects of worn tonewheels, a feature which is fortunately user-adjustable! The B4000 also offers a built-in Leslie rotary speaker emulation, with separate control of and access to the bass and treble 'rotors'. Like the other instruments in the ASB range, the hardware module can be controlled via USB from a stand-alone Mac or PC Editor, which offers you access to more parameters than can be edited from the B4000's front panel. It'll cost £649 and will be available by the end of March.

Sonic8 +44 (0)8701 657456
www.sonic8.com www.creamware.de

Digital Performer 5
launched at Winter NAMM

MOTU used the NAMM show to unveil a major update to *Digital Performer*. ***DP5*** features no fewer than six new virtual instruments, including several types of synthesizer, a drum machine and a CPU-efficient 'mini-sampler'. Then there's the Meter Bridge, a new window which allows the user to monitor the levels of all signal paths within the *DP* mixing environment in one place. Tracks can now be organised into folders and sub-folders and then collapsed or expanded, and four new audio editing tools have been added to the tool bar. Version 5 also has some new features relating to music-for-picture projects, always one of *DP*'s strong suits. Professional visual cues can be added to movie files and a 'visual click' can be superimposed on Quicktime movies. Other new features include click and count-off refinements, the ability to add non-destructive level automation to individual soundbites and the ability to play MIDI notes from the computer keyboard. Pricing for version 5 was not available when we went to press, but previous experience suggests that it's likely to be the same as the current version — £399.



Meanwhile, MOTU have released ***Ethno Instrument***, a sample-based virtual instrument for Mac and PC. Built on the UVI Engine which also powers MOTU's *Mach Five* sampler, *Ethno Instrument* is equipped with 4GB of sampled sounds and 4GB of loops and phrases from around the world, and synth-style envelope, filter and modulation controls. It can operate as a stand-alone application or as a VST, Direct X, MAS, Audio Units or RTAS plug-in. It costs £249.

Musictrack +44 (0)1767 313447 www.musictrack.co.uk www.motu.com

New plug-ins from Waves

Waves have replaced the ageing *X-Noise* with a new noise-reduction plug-in based on 'a brand-new algorithm that reconceptualises the way noise is treated'. ***Z-Noise*** is claimed to offer unprecedented transient preservation and low-frequency resolution, and unlike many rival products, doesn't need to be shown a 'noise only' sample in order to work at its best. It also has an adaptive mode allowing it to track a changing noise signal over time. The other big news from Waves is the ***SSL 4000 Collection***, a plug-in bundle developed under license from Solid State Logic. The three plug-ins replicate the G-series stereo buss compressor, the E-series channel strip, which combines EQ and dynamics, and the G-series channel equaliser (pictured here). Waves say their engineers spent over a year analysing SSL hardware and that the plug-ins sound 'virtually identical'.

Sonic Distribution +44 (0)1582 470 260
www.sonic-distribution.com

Open Labs
produce smaller keyboard PC

Open Labs' Neko keyboard-based computer caused a stir when it was announced at NAMM a couple of years ago, and our December 2004 review rated it as a capable PC-based recording platform. But it was also rather heavy and quite expensive, and with no direct distribution in the UK, the cost of shipping a system over from the US was also significant. However, at this year's NAMM, Open Labs announced the considerably smaller and much lighter **Miko**. Retailing in the States for $1999 (about £1120 at the time of going to press), it should cost much less to ship. Other than being smaller, Miko is based on the same AMD 64-bit processing architecture as the Neko (systems are partly user-configurable, so you can order a dual-processor system if you want), and it runs Windows XP, concealed within a 'shell' OS that allows you to access just the features that you need to run music software, manage I/O and access plug-ins. Other than that, the Miko is a Windows PC at heart, and will run most plug-ins or software designed for that platform, and will, claim Open Labs, record or play back up to 64 tracks of 24-bit/96kHz audio simultaneously. Externally, in addition to the computer keyboard, 37-note musical keyboard, built-in 15-inch colour touchscreen and dual fader/knob panels, the Miko includes a licensed Presonus four-in, six-out audio section with two phantom powered mic/line preamps, two line inputs and a dual-channel headphone out. Outputs are on six balanced jacks and a digital Toslink connector. We'll be taking a proper look at it later this year.

Open Labs +1 512 444 6222 **www.openlabs.com**


"There'll be no collection
necessary for the review sample.
It's staying here."
Phil Ward– Sound On Sound

Arturia model Prophet synths

Arturia added another 'V' software emulation to their range of virtual instruments at this year's NAMM show. ***Prophet V*** models (you guessed it) Sequential Circuits' Prophet 5 five-note polyphonic subtractive synth and also includes an emulation of Sequential's later Prophet VS vector synth, the forerunner of the Korg Wavestation. The Prophet 5 emulation is based on both Rev 2 and the more stable Rev 3 versions of the original instrument, and offers all 54 parameters and the 40 presets of the original synth, as well as a new mono mode and plenty of completely new presets. The VS emulation includes all the original 96 digital waveforms, which can be assigned to up to four oscillators and blended and crossfaded using the MIDI-controllable joystick and the five-point envelopes of the original synth. Both emulations can be run simultaneously, and built-in chorus and stereo delay effects can be added. Unlike the original synths, *Prophet V*'s polyphony can be set from two to 32 notes, host processor power permitting. *Prophet V* is available in stand-alone, VST, Direct X, RTAS and Audio Units versions and costs £139.99.

Arbiter Music Technology +44 (0)208 207 7880.

www.arbitermt.co.uk www.arturia.com

New gear from Presonus

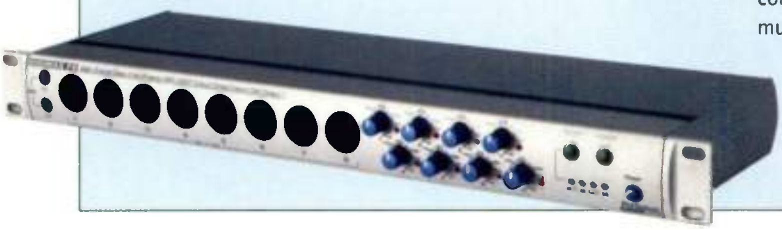
Following in the footsteps of the Firepod interface, Presonus' new **Firestudio** is an 18-in, 18-out Firewire interface equipped with eight mic preamps and S/PDIF, ADAT, word clock and MIDI I/O. But what sets the 1U rackmounting Firestudio apart from the many other Firewire interfaces out there is the included Monitor Station Remote, or MSR, a handy little box that connects via Cat 5 cable and provides a range of monitor controller features. It has two headphone amps, a talkback mic, a level control and Mute, Mono and Dim buttons, and users can switch between tracking, stereo mixing and surround mixing modes.

Pictured below, looking almost identical to the Firestudio, the new **Digimax FS** is in fact an eight-channel Class-A mic preamp with 24-bit/96kHz ADAT digital output, plus balanced analogue direct outs and balanced insert points on every channel. It's primarily aimed at anyone wanting to add eight high-quality preamps to their computer audio interface or digital mixer via ADAT. The sample rate can be set at 44.1, 48, 88.2 or 96 kHz and the Digimax FS will happily sync to external devices via word clock.

Finally, Presonus have launched the **Faderport**, an unconventional USB MIDI controller that provides one long-throw motorised fader, a pan knob, transport controls and a variety of other buttons (Mute, Solo, Record and so on). The compact desktop unit is intended as an easy way to add fader automation to single or grouped channels, as well as providing quick access to a variety of common controls, although claiming that it offers 'mouse-free music production' is over-stating the case a little! The Faderport is compatible with Windows and Mac OS and supports all the major DAW packages.

Hand in Hand Distribution +44 (0)1579 326155.

www.presonus.com www.handinhand.uk.net



Boss rinse and repeat

The Boss Loop Station concept has just become even more sophisticated with the addition of the new **RC50** seven-footswitch floor unit. It positively dwarfs the diminutive RC20XL looping pedal we reviewed back in September 2005 (www.soundonsound.com/sos/sep05/articles/bossrc20xl.htm), and allows three different stereo loop tracks to be manipulated at the same time. It also supports multiple overdubs, so phrases can be stacked. The RC50 can be locked to MIDI Clock and offers Loop Quantize and Loop Sync functions so that all recorded phrases can start together. Tempo Sync allows the tempo of multiple phrases to be matched during playback. There are 376 play-along sampled-drum guide patterns in a variety of time signatures and up to four optional footswitches can be added for extra control.

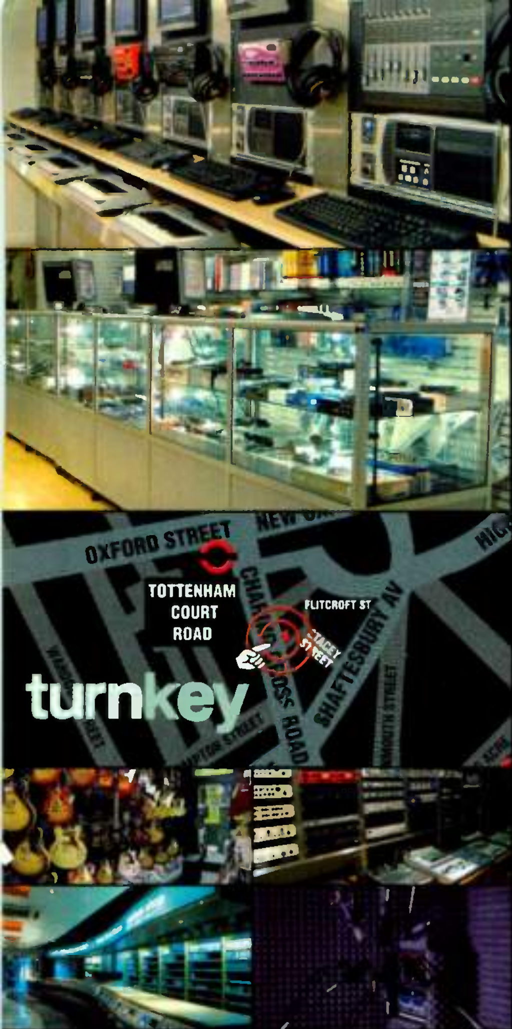
For those wanting a straightforward means of recording, the Boss **BR600** is a cost-effective, stand-alone, Compact Flash-based multitracker that provides eight simultaneous playback tracks, each with eight virtual tracks. Digital effects are built in, including COSM guitar and bass-amp models, chorus, delay and reverb, plus EQ for every channel. Pitch correction is also included, as is a drum machine offering almost 300 drum and percussion patterns. Beats may also be programmed from the velocity-sensitive pads. For portable recording, the unit can operate on six 'AA' batteries and there's a stereo mic built in.

Roland UK +44 (0)1792 515020 www.roland.co.uk

Novation expand controller keyboard range

The Novation Remote 25SL controller keyboard, which featured on the cover of last month's *SOS*, was praised for its innovative Automap automatic control-mapping software, its variety of controllers and its handy LCD displays (www.soundonsound.com/sos/feb06/articles/novationremote25sl.htm). Now Novation have produced 37- and 61-note versions of the Remote SL. The new **Remote 37SL** and **Remote 61SL** (the latter pictured below) have all the same features as the original, with the bonus of a few extra octaves to play with. They're good value too — the Remote 37SL costs £369, while the Remote 61SL costs £419.

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



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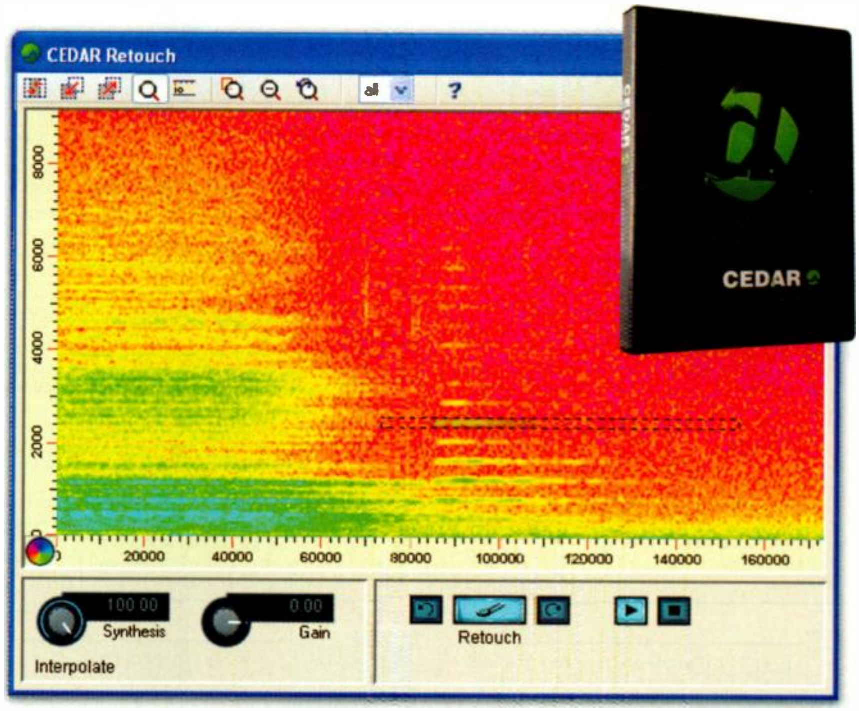
Cedar Audio *Retouch*

Worth £2350



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Audio restoration systems have traditionally been dedicated — and limited — to removing certain types of noise, most commonly crackles, pops, clicks and scratches. Cedar Audio's *Retouch* is different. It does allow you to identify and remove all of these, but it's also capable of eliminating a huge variety of other sounds, from coughs and record scuffs to church bells, squeaky chairs, banging doors and even car horns.

Whilst it's possible to remove some kinds of unwanted noise using EQ or compression, these techniques can damage 'good' signal and cause undesirable side-effects such as high-frequency drop-outs. *Retouch* operates on both the spectral and temporal content of the audio, and offers a graphical view to help the user to accurately identify and specify unwanted sounds. Once this is done, the offending audio can be replaced with sound derived from the surrounding signal — leaving all other audio untouched and intact.

Since its initial release in 2002, *Retouch* has quickly become a favourite with professional audio engineers and has been used to fix problems with unwanted noise on all kinds of projects, from archive recordings of the Goon Show to the latest Harry Potter movie. *Retouch* has helped to save countless projects — if you're working with audio from a live performance, or restoring an old recording, it can be invaluable.

Cedar Audio opened the world's first commercial audio restoration studio in Cambridge in 1989. Since then they have gone on to develop some ground-breaking technology and have been involved in more high-profile restoration projects than it is possible to list here. This month, they have generously agreed to give away a copy of *Retouch* for *Pro Tools* to *Sound On Sound* readers. *Retouch* v3 is available for *Pro Tools HD* and *LE* v6.9 onwards on PC and is worth £2350.

If you would like a chance to win this fantastic prize, simply fill out the entry form at the bottom of this page and post it to the address on the coupon. Alternatively, you can enter via the electronic form on the *Sound On Sound* web site. Please make sure you answer all the questions and complete the tie-breaker. We also require your full address, including your postcode and your daytime telephone number. The closing date for entries is 30th April, 2006. SOS

questions

When was Cedar's studio first opened?
a. 1979 ☐
b. 1989 ☐
c. 1999 ☐
d. 2005 ☐

Which of these sounds can *Retouch* not remove?
a. The sound of a banging door ☐
b. The sound of church bells ☐
c. The sound of a car horn ☐
d. The sound of an explosion in a vocal booth ☐

The filmic adventures of which fictional wizard were assisted by *Retouch*?
a. Gandalf ☐
b. Harry Potter ☐
c. Rincewind ☐
d. Merlin ☐

Cedar Audio *Retouch* tie-breaker

***Retouch* is very clever, but imagine if it could be applied to actual events. You could edit out the six weeks you spent in that dodgy goth band and replace it with a summer holiday from when you were 11. If you could edit out any part of your musical life, what would it be and why?**

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

Post your completed entry to: Cedar Audio March 2006, *Sound On Sound*, Media House, Trafalgar Way, Bar Hill, Cambridge CB3 8SQ, England.

the small print

1. Only one entry per person is permitted. 2. Employees of SOS Publications Ltd, Cedar Audio Ltd 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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* 80 note polyphony based on average patch load. Oscillator/filter model selection can result in slightly less or even more.

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Visit the *SOS* Forums via www.soundonsound.com

A designer responds...

I write to you concerning Gordon Reid's review of our 200e Electric Music Box, presented in your December 2005 and January 2006 issues. Seldom do I comment on reviews, but I wish to submit some corrections and clarifications.

The first and foremost issue is the build quality. Your reviewer attacks me on four points, all of which I shall defend. We design instruments that far outlast others. All our components are subjected to in-depth scrutiny, particularly with regard to longevity (mean time to failure) before being considered for application. The "wobbly" pots in our system do not reflect imminent failure — the wobbliness stems from the support scheme used by the bearings. The failure rate is zero over millions of hours of testing.

Secondly, your reviewer attacks my choice of interconnects. These standards were established in 1963. I have seen zero banana jack failures and zero signal connector failures in this period, and continue to use the same parts built by the same reputable manufacturers.

The third reflection on build quality is the failure of a plastic latch, used to stabilise the system while hand carrying it or moving it in or out of its protective case. This was the extent of its intended application — it was not meant to keep the system locked in its closed position while transporting it in the rear seat of an automobile or on public transportation. A very sturdy carrying case is provided with the system, although I understand this was not used when transporting the *SOS* review system — use it, and your system will be properly protected during transport. Would you transport a violin without its case?

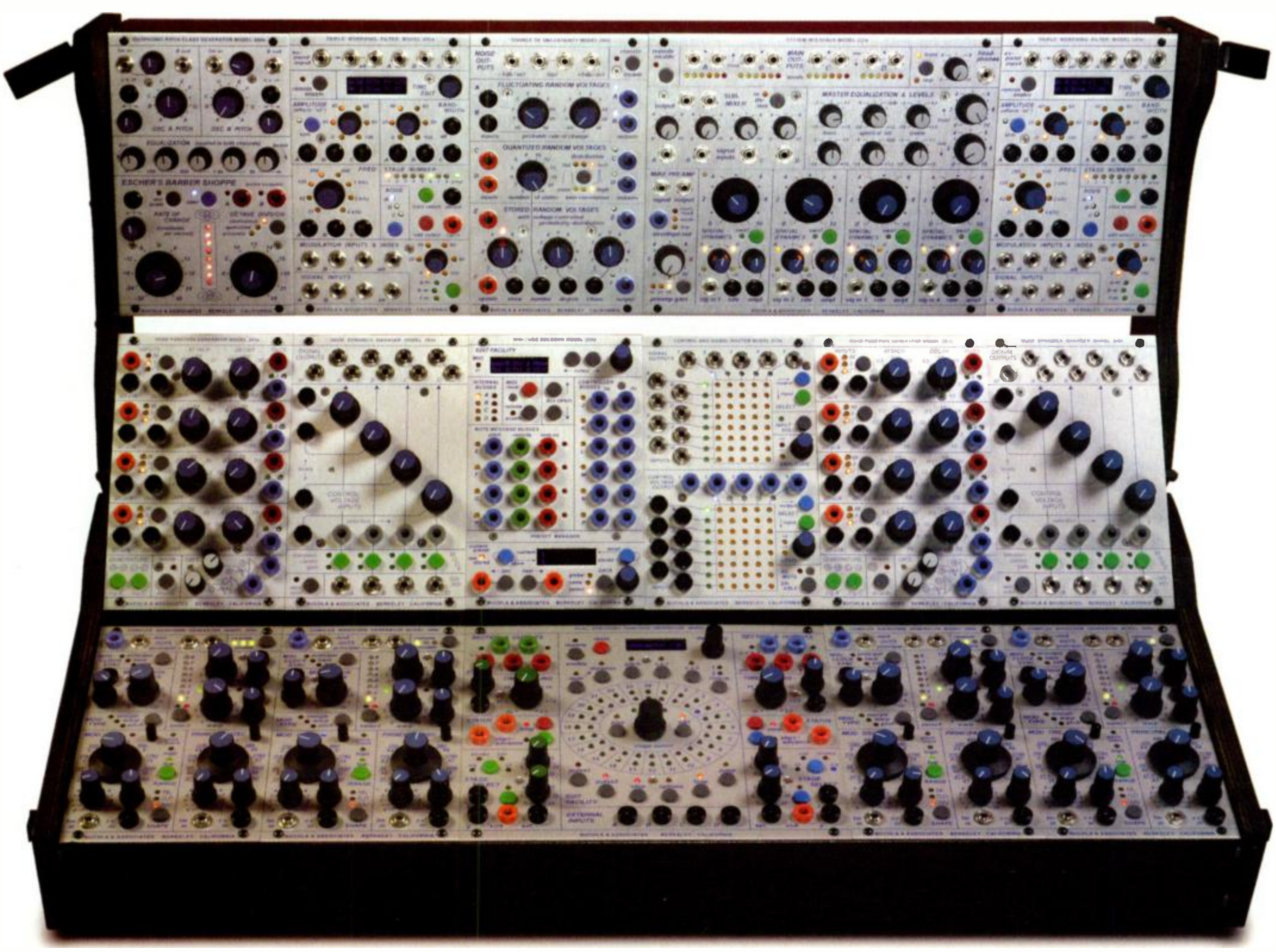
Lastly, your reviewer takes me to task for my choice of finishing materials. Aesthetics influence this decision, but let me point out that the nine-ply, void-free, marine-grade plywood used on the 200e and denigrated by Gordon is definitely not cheap — it's sourced from Finland and Russia. Perhaps you'll find solace in investigating the inner makings of tennis racquets, skis, pianos, and furniture, all of which have used laminated wood for vastly superior strength and stability.

Gordon also suggests that our 200e systems are not readily compatible with our older 200-series models. I wish to clarify this. Firstly, there's the issue of physical compatibility. All 200 modules (including the newer 'e' models) are built with panels that follow the 4.25 x 7-inch standard that we introduced in 1963. However, the Model 212 Dodeca module and the Model 275 Reverberator are too deep to fit into 200e cabinets.

Electrical compatibility is also important, and our last power supply revision was in 1970. The 200e provides highly regulated supplies at plus and minus 15 Volts (not the 12 Volts of the external power supply), a fact easily confirmed with a little probing of the innards. Only two older modules, again the 212 and 275, require an additional voltage (24 Volts).

Signal, control voltage and pulse levels are another aspect of compatibility. Once again, our last revision was in 1970 — all 200 modules, including 200e versions, have followed suit. My conclusion is that the 200e series modules maintain their intended compatibility with their predecessors, with the aforementioned rare exceptions.

Now to the issue of expense. I am aware that our system is not the cheapest on the street. But "a staggeringly huge price tag"? I'm reminded of the hi-fi gear introduced by ▶



Electrovoice in the late '60s. About one-eighth the size of conventional equipment, the stuff was solidly built, definitely up to snuff in performance and similar in price. But it failed to make an impact — the public seems to feel that superior stuff has to occupy superior volume. With the acceptance of car stereo, small of necessity, the public's taste has loosened up a bit, but high end hi-fi equipment still comes in big, mostly empty, cabinets.

Price tags on gear with performance characteristics similar to that of the 200e, but housed in multiple and definitely non-portable racks, is equally expensive, yet causes no 'sticker shock'. This is a principle I am well aware of, but choose to completely ignore when designing new products. I would have hoped that your esteemed reviewer would see beyond the 'bigger is better' principle. Actually, maybe he did, when toward the end of the second part of the review, he remarked, "is the 200e really that expensive? If you carefully consider what it might cost to purchase a modular synth with similar features from elsewhere... the Buchla can almost seem cheap". Maybe, in the course of his research, a radical change of mind occurred.

Gordon also suggested that the prices quoted for the 201e6 and 102e18 cabinets, $700 and $1400 respectively, were the prices for "the empty cabinets, not for the cabinets filled with modules as in the *SOS* review system". At those prices, the cabinets are not empty, but, in addition to the accompanying external 12 Volt supply Gordon noted, contain highly regulated, completely isolated power supplies, as well as low impedance power distribution rails and sockets. They also house internal data busses and card slots, and the elaborate, but invisible, interconnection wiring that establishes electrical connection between boats.

In the opening paragraph of the Overview, in the first part of the review, Gordon reminds us of a few things that the 200-series is *not*. As for what it *is*, the 200e is internally a hybrid synth, analogue in the signal paths, and digital in the control sections; an easy separation to implement and maintain as a result of signal versus control-voltage distinction. As far as I'm concerned, the system is generally analogue. The controls are almost all functionally dedicated and interconnections are analogue; the 200e simply behaves as an analogue synth.

Hopefully, I leave your kind and tolerant but inquisitive and deserving readers with a slightly better sense of what we're all about. Reliability is of the highest concern to us, and we intend to continue making instruments of unimpeachable quality. We do our best to build fine instruments at fair prices, but we promise to avoid the cheap. Fame, fortune and financial success are of secondary importance, and we'll happily flail about in "the backwaters of the music industry" for a long time to come.

May you enjoy the world's many instruments.

Don Buchla, Buchla & Associates

Managing Editor Matt Bell replies: Thanks for your comments and clarifications, particularly with respect to the precise nature of the compatibility between the older 200-series modules and the 200e system. In fairness, we didn't suggest that it was impossible to use 200-series modules in the newer casings, merely, as stated on the Buchla web site, that there were some compatibility issues. Clearly, however, these are minor and don't affect the majority of 200-series modules.

As regards the build quality, our concerns were founded on the feel of the controls based on many years of experience of audio and recording gear. To a man, the entire editorial staff of *Sound On Sound* had feelings similar to Gordon's about the pots and connectors when first confronted with the 200e during its time at our offices, and, having all had experience of 'homebrew' kit synths with plywood casings, we felt that the plywood created an impression unsuited to the cost of the system. In fairness, though, we can confirm that there were no component failures of the review system while the 200e was in our possession, with the exception of the plastic locking clip — but then, we had not been supplied with the carrying case Don Buchla mentions, and had to transport it without the case's protection, which may have further weakened the clip.

Our comments on the price of the system were twofold, and may seem contradictory, but we regard them as complementary. We can confidently say that $20,000 dollars is the kind of outlay that nearly all *SOS* readers would have to seriously consider before making. In that sense, the *absolute* cost of the system is indeed high. But it's only fair to also consider the *relative* cost, as Gordon did towards the end of the two-part review, by comparing the 200e, as far as possible, to a comparable system offering comparable features. As Gordon stated, such comparisons involving the 200e are not straightforward, as there aren't many other synths quite like it, but by doing this as far as possible, he concluded that the Buchla could indeed seem cheap by comparison.

The concern about the price of the 'empty' cabinets stems from a misunderstanding. We were concerned that some readers might have felt the 700 and 1400-dollar price tags related to the boats filled with modules, as in the review 200e system, whereas of course, these prices are for the cabinets before they are filled with modules. The use of 'empty' here was intended to signify an unloaded boat, not that this price bought you a completely empty wooden shell devoid of power supplies. I hope that this is now clear.

Finally, it's important to state that in publishing the extensive 200e review, we wanted not just to review that system, but also to reintroduce Buchla and his considerable achievements to the world. We felt, as we stated at the start of part one, that such redress was long overdue, particularly amongst our readership in the UK and Europe, where Don Buchla's name is much less well-known than in the United States. Far from being designed to slight Don and his work, the article was intended to be a celebration of his achievements, and an attempt to familiarise our readers with his unique approach to synth design.

As part of this process of reintroducing Buchla and his products, we wanted it to be very clear from reading the review that although the 200e might bear a resemblence to other patchable or modular synths, there was a very different design ethos at work here, and that readers approaching the system with expectations that it would behave like other modulars would have to rethink their attitudes. Hence the extensive comments about what the 200e was *not*, the extensive coverage given to the individual modules' capabilities, and the publication of the unconventional waveform traces in part two. The idea was that by reading to the end of part two, the average *SOS* reader might gain, as Gordon did while writing the review, a much closer understanding of Buchla's designs. We'd be interested to hear whether you feel we succeeded in our aims! SOS

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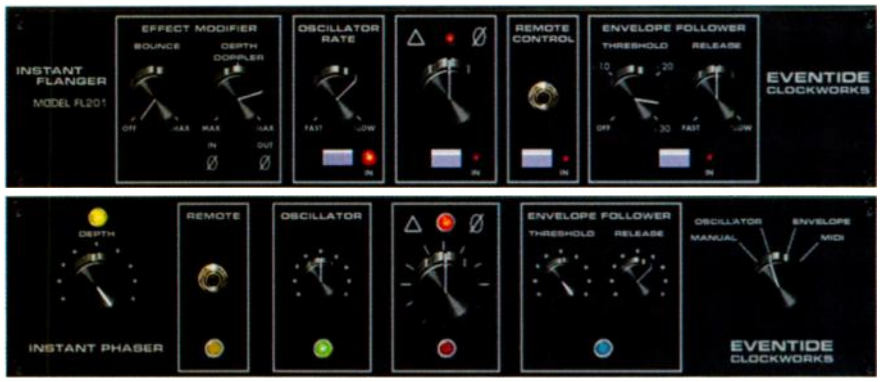


Eventide have re-created their revolutionary Instant Flanger and Instant Phaser units as software plug-ins.

Q What's the difference between phasing and flanging?

I have several phasing and flanging plug-ins, and the same effects on my Korg Triton. While not identical, the two effects seem quite similar. What's the difference between the two?
Neil Morley

***SOS* contributor Steve Howell replies:**
These two types of effect are indeed related in many respects, and they are both created by delaying the signal by small amounts and then mixing this with the dry signal.

If you delay a simple sine wave and superimpose it on the original waveform, when the phase of the two sine waves is close, the sound is reinforced in some places. But when the two are exactly 180 degrees out of phase and exactly equal in amplitude, this results in total cancellation. It's simple maths — add +5 to -5 and you're left with nothing.

With more complex waveforms, superimposing the same signal and delaying it slightly creates what is known as a 'comb' filter — the frequency response of the filter has various peaks and troughs of amplitude ('teeth') across the harmonic spectrum. Some frequencies are reinforced whilst others are cancelled due to phase differences across the frequency spectrum, and when the delay time is changed (typically with an LFO), the teeth move to create the characteristic 'swooshing' effect common to phasers and flangers. However, whilst essentially based on the same principles and producing a similar effect, the two effects are achieved differently.

With the phaser effect, the signal passes through all-pass filters which have a non-linear frequency phase response. This results in phase differences in the output signal that depend on the input signal frequency — for example, the phase of a low frequency might be shifted by one quarter of a wavelength, whilst a higher frequency will be phase-shifted by a different amount. In other words, various frequencies in the original signal are delayed by different amounts, causing peaks and troughs in the output signal which are not necessarily harmonically related.

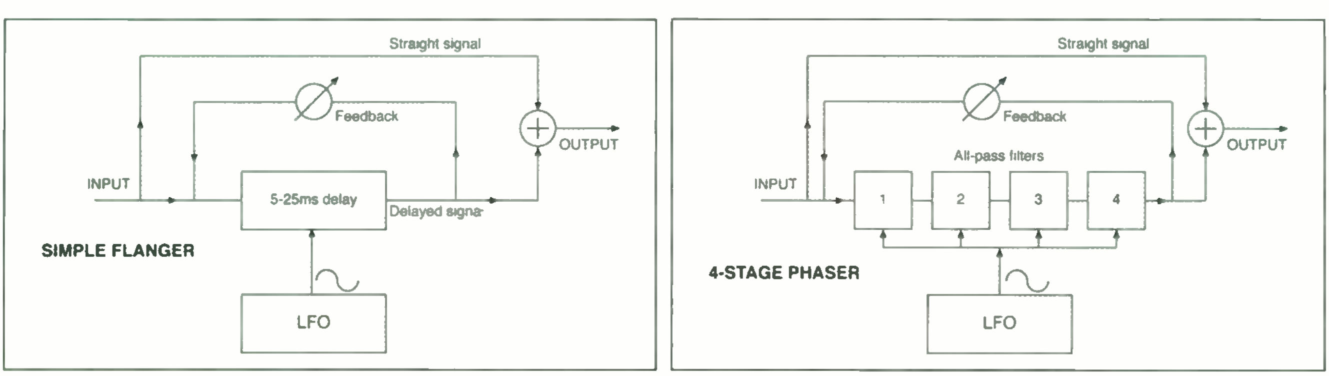
Flanging, on the other hand, uses a delay applied equally to the entire signal which is similar in principle to phasing except that the delay (and hence phase shift) is uniform across the entire sound. The result is a comb filter with peaks and troughs that are in a linear harmonic series.

Typically, the comb filter created by flanging's uniform delay will have many uniformly spaced 'teeth' whilst a phaser (depending on the design of the circuitry) will have fewer teeth — maybe even just one — and/or they will be unevenly spaced and the spacing will depend on the whim of the designer, using tried and trusted scientific principles such as 'does it sound good?'

Flanging is easily created using a short delay (around 10ms) that is modulated by an LFO. Phasing, however, needs further consideration. Because its frequency response has fewer 'teeth', having just one all-pass filter gives a subdued phase-shifting sound. But by adding extra 'stages' (in other words, extra all-pass filters), you can add extra degrees of phase shift, which is why you might see some phasers with an option to select 90, 180, 270 and 360 degrees of phase shift — each one switches in another filter, thereby potentially introducing more 'teeth'.

The term 'flanging' comes from the original technique of using two synchronised tape machines playing back identical audio — during playback, the flange (or rim or outer edge) of one of the tape machine's reels would be obstructed in some way — slight pressure applied with the operator's finger to the reel, for example — so that one tape machine was delayed ever so slightly for a brief moment and then, as the 'obstructed' tape machine gradually got back in sync with the other, you'd hear 'that sound'.

Flanging and phasing remained very definitely in the Heath Robinson domain for many years, using two manually obstructed tape machines, until the company Eventide used analogue electronics to produce the same effects in their Instant Flanger and Instant Phaser units. Despite advances in modern DSP technology, these units still stand the test of time as perhaps the ultimate effects of their type (although that is not to dismiss their modern successors, all of which can produce excellent results) but the Eventides uniquely allowed the effect to 'go through the null', the point where the effect is phase/time-shifted such that it creates a dramatic (but temporary) total audio phase cancellation. During the '70s, many manufacturers released some fantastic phasing and flanging products. Of note (in my ▶



These simplified block diagrams show a flanger circuit, which applies delay equally to the entire signal, and a four-stage phaser, which uses four all-pass filters to delay different frequencies in the original signal by different amounts.

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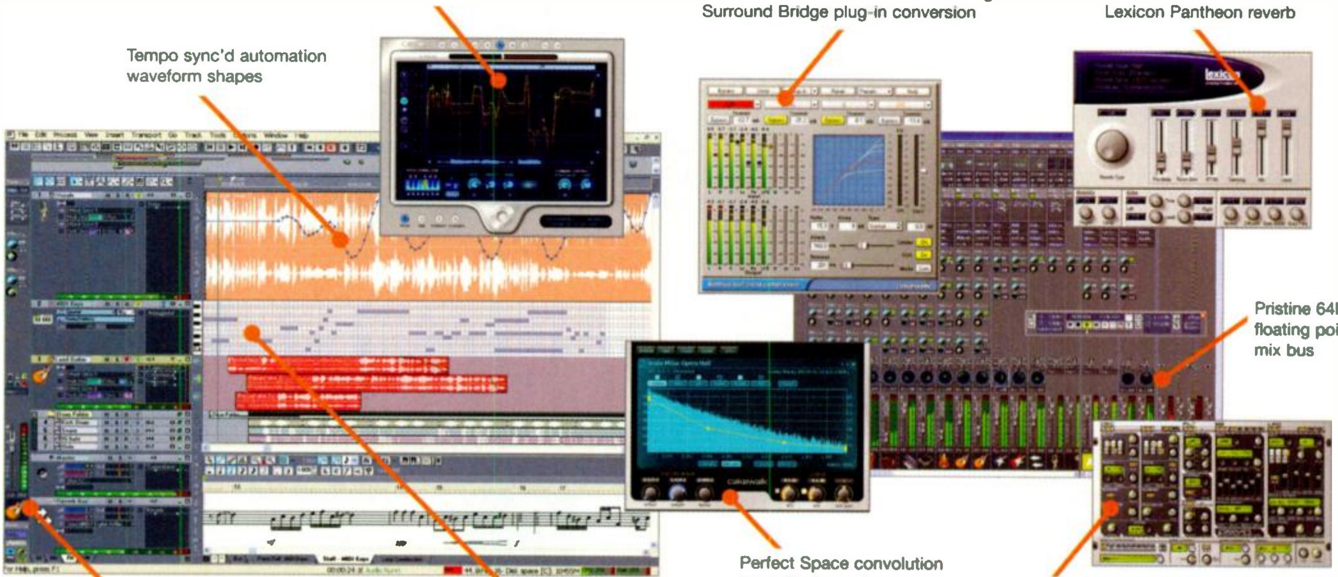

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SONAR 5 Studio Edition	unlimited	unlimited	500kHz	64-bit FP	•	•	•	•	•	•	—	—	•	5 (2) Roland synths groove sampler SoundFont sampler, analog modeling	•	•	•	•	•	•	•
SONAR 5 Producer Edition	unlimited	unlimited	500kHz	64-bit FP	•	•	•	•	•	•	•	•	•	8· (2)rgc.audio synths, REX synth, (2) Roland synths, groove sampler SoundFont sampler, analog modeling	•	•	•	•	•	•	•
Cubase SL 3.1	128	5	96kHz	32-bit FP	•	—	—	—	•	•	—	—	—	3 dual-oscillator virtual bass, drum machine	•	—	•	•	—	—	—
Cubase SX 3.1	256	8	96kHz	32-bit FP	•	—	—	—	•	•	—	—	—	5. dual-oscillator virtual bass, drum machine, surround pad, mono analog	•	—	•	•	—	—	—
Pro Tools LE 6.9 (and M-Powered)	Digi002.18 Mbox:2	5	96kHz	32-bit FP	—	—	—	—	—	—	—	—	—	4. Reason Adapted 3, Ableton Live Lite 4, SampleTank LE, Fxpansion BFD lite	—	—	•	•	—	•	•

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Q&A

opinion) are the Electro Harmonix pedals — the Electric Mistress flanger and the Big Stone and Small Stone phasers — which were cheap and cheerful but offered fantastic sounds. Roland also produced some gorgeous-sounding pedals and rack units during this time, as did MXR, Mutron, Oberheim and others.

Flanging and phasing are, of course, most obvious and startling on harmonically rich sound sources but can be equally effective (if used more subtly) on mellower instruments — it was a popular effect on the Fender Rhodes in the '70s, and Japanese synth-meister Isao Tomita would regularly run mellow sounds through his Eventides to great effect. When using either effect, the parameters are almost identical. Typically flangers and phasers will offer LFO speed and depth controls (and maybe a choice of LFO waveform, although this is typically a triangle or sine wave), as well as a feedback control that enhances or heightens the effect. But whereas a flanger effect will sometimes offer a base delay time (typically 5-20ms), a phaser might offer a base phase angle as well as the aforementioned 'phase degree' switch. In either case, the modulating LFO will 'rotate' around these base settings to provide a variety of different effects.

Today, there are many variations on the basic effects. We have stereo effects where the left and right channels' modulation is inverted giving rise to a subtle panning effect. We also have effects where multiple delay lines and/or all-pass filters are used in parallel (and modulated at different phase angles of the modulating LFO waveform) to create rich, evolving textures, and we also have 'crossover' effects where the feeback from one channel is fed to the other (and combinations of the above) to create all manner of swirling textures.

Q What's the best way to set up instruments in *Logic*?

I understand that if you have Mac OS 10.4 'Tiger' you can add and set up external MIDI instruments using the Audio MIDI Setup utility. I understand that it's a much simpler way of routing instruments and that *Logic* can see this setup, thereby avoiding having to use *Logic's* Enviroment page. Can you tell me the procedure for doing this?

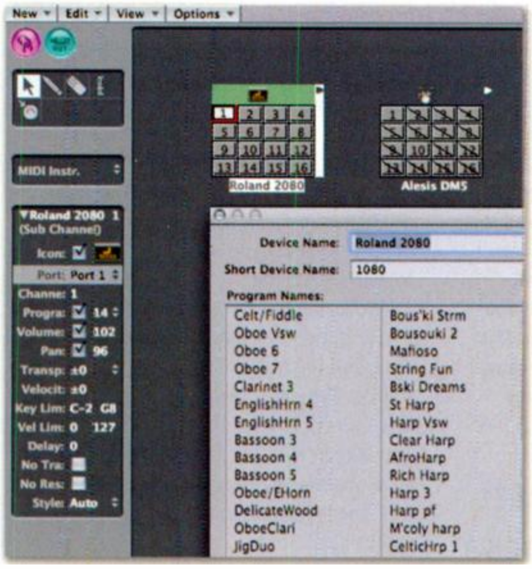
Morni Otadaferua

Editor In Chief Paul White replies: While you can do some neat 'drag to connect' things in OS X's Audio MIDI Setup window (which is found in the Utilities folder in Applications), adding MIDI instruments to *Logic* is still best done in the sequencer's Environment window. You can set up and test MIDI ports in Audio MIDI Setup (AMS) just by scanning your connected interfaces, and all available MIDI ports show up in *Logic*. But there are lots of things you can't do in AMS (or at least I can't find them!), such as setting up Multi Instrument object Parameters, MIDI Bank Change message protocols and patch names, so I'll explain how to set up a MIDI instrument in the Environment window.

First, if you're not on the MIDI Instruments Layer already, select it from the menu just beneath the Toolbox. If you can't see the Toolbox, use the View menu to turn the Parameters view back on. Once you're in this page, go to the New menu and select New

Multi Instrument. This will create a new Multi Instrument object with 16 squares corresponding to the 16 possible MIDI channels, though it doesn't matter if your instrument has fewer parts than this, or even if it isn't multitimbral at all. Initially the squares will have diagonal lines through them, so you need to click on the ones you wish to use with your instrument to make them active. For example, if you have an eight-part multitimbral synth that is set to operate on MIDI channels one to eight, then click on boxes one to eight to get rid of the diagonal lines. Now you can click on the Multi Instrument name at the bottom of the icon and use the Text tool from the Toolbox to open up a name dialogue box. Give your object a name relating to the physical synth you have plugged in.

With the Multi Instrument object still selected, go to the Parameters box at the left of the window and select the physical MIDI port to which your synth is connected. It's also worth selecting each of the squares you've unticked and then ticking Program, Volume, and Pan in the parameter box so that your connected instrument can receive these MIDI messages from *Logic*. Why these are off by default I'll never know! If you select your Multi Instrument by clicking at the top of the object, the parameter box on the left should show All in the MIDI Channel information line. If you select by clicking at the bottom of the object, the MIDI channel of whichever numbered square is currently selected will show up in this line.



It's best to set up hardware MIDI instruments in *Logic*'s Environment page.

You can also change the graphical icon assigned to your Instrument by clicking and holding the mouse pointer over the existing icon in the Parameter box. There are plenty to choose from, and selecting a suitable icon for each device can help you keep track of what's going on in the Arrange window. The icon box needs to be ticked if it isn't already. A message will then ask if you'd like this icon to apply to all the MIDI channels of your instrument. Normally this is fine, but it's up to you.

Finally, if you double-click on one of the numbered squares you've switched on, a patch window will open up filled with General MIDI patch names. In this window you can choose the MIDI Bank Change message required by your instrument from the Bank Message menu, and you can also type in new patch names to ▶

World Radio History

match your actual synth. This is pretty tedious, but a quick Google search will almost certainly locate a ready-made Environment object for your particular synth, in which case just cut and paste its Multi Instrument object into your Environment before you complete all the steps that come after creating your Instrument object. Needless to say, you should do all this in your default or template song so you only have to do it once. To add the new Multi Instrument object to an older song that you're resuming work on, simply copy and paste it from your default song and then make sure it is assigned to the correct MIDI port in the Parameters box.

Q How do I re-create the sound of those old string synths?

I am trying to re-create the sound of old string synths like the ARP Solina or Elka Rhapsody on my Novation K-Station but I just can't do it. I have tried all the tricks to create a fat, chorus/ensemble sound — detuning oscillators and so on — but it always sounds like a polysynth, not that string synth sound. Any suggestions?
Neil Davies

***SOS* contributor Steve Howell replies:** The way in which the early string synths made that distinctive sound is very different to the polysynths that came after them (including the excellent K-Station).

Old string synths use what's known as a 'divide-down' oscillator, derived from electronic organ technology, as their sound source. This generates a single sawtooth-like wave which, if heard 'raw', sounds very thin. This is then fed into a simple attack/release envelope shaper. Typically, the attack control was labelled 'Swell' and release was called, confusingly, 'Sustain'. But herein lies a problem — although the divide-down oscillator was totally polyphonic, the simple envelope shaper was usually monophonic, which created certain problems. On some string synths, chords would swell in nicely, but playing another note whilst sustaining the initial chord would re-trigger the envelope and the whole thing would fade in again — the original chord and the new note. Other synths worked such that new notes played whilst sustaining others had no envelope shaping. In either case, some care had to be taken with regard to playing technique. The problem was overcome in the Roland RS202 string synth, which had separate envelope articulation for each note, but I digress...

The secret to a string synth's lush and swirly sound is that this thin, envelope-shaped waveform passes through a chorus unit. However, this was not your typical chorus unit, such as you'll find in the Novation K-Station (or other synths, analogue or digital). Instead of a single modulated delay line, a string synth's chorus unit typically uses three delay lines modulated at different rates and depths by independent LFOs. This was intended to re-create the natural phenomenon in real string sections where no player's vibrato is the same, creating a rich ensemble sound. The outputs from these chorus units were then summed to produce that thick 'string synth' sound that is quite unlike the sound of two (or three) detuned polysynth oscillators.

However, it is possible to create something

Unlike the first string synths, which had monophonic envelope shapers, the Roland RS202 offered separate envelope articulation for each note.

It's possible to imitate the string-synth chorus effect by using an LFO to modulate the tuning of the Novation K-Station's oscillators.

that's close to the classic string synth sound, and modern virtual analogue synths such as the K-Station are often better suited to this than older analogue synths. This is because they have rock-solid tuning, which is a good thing in itself but also better emulates the divide-down oscillators used in the originals, and they generally have better modulation facilities to re-create the ensemble sound. And because the K-Station has three oscillators, it's possible to get quite close to the sound of the three chorus units that string synths like the ARP employed. The same principles apply to other synths though, and I have achieved the same thing on two-oscillator synths.

First select one of the K-Station's 'blank' patches (the 'default' sound — single sawtooth OSC 1, filter wide open), bring up OSC 2 and OSC 3 on the mixer and select a sawtooth wave on all three oscillators (again, the default selection in the 'blank' patches). Tune all three to unison with no detune (Detune value of 0 — again, the default setting). Set LFO 1's speed to around 70, and for OSC 1, set LFO 1 Depth to +5, and for OSC 2, set LFO Depth to -5. All of these values can be adjusted to taste later on. Leave OSC 3 untouched.

That should give you something that resembles the basic detuned, 'chorusy' tone of a string synth. It's not entirely authentic but set carefully, it's pretty close. The trick here is not detuning the oscillators as you might expect but recreating the effect of the three different chorus units by independently modulating the pitches of the ▶

different oscillators separately. On the K-Station, because you can only control the oscillators using LFO 1, the fact that one oscillator's modulation is inverted is intended to fake the effect of separate LFOs.

After that, of course, you can do what you want with the sound. Set a slow Amp Env (slow attack and long release, full-level sustain) to emulate a string synth's typical envelope shape and maybe use the K-Station's filter to refine tone. You can add delay and reverb if you want and maybe a hint of chorus to smudge everything, maybe even phasing to re-create the classic Jean-Michel Jarre string synth sound. Experiment with EQ as well — some old string synths were a bit 'honky', so try boosting mid frequencies slightly!. You should also experiment with filter sweeps which will allow many of the sounds possible on the ARP Omni, a curious string synth/polysynth hybrid. For experimentation, try selecting a pulse wave on one of the oscillators and maybe even some PWM controlled by LFO to add further separate animation. You might also like to drop OSC 3 by an octave to create the sound of a lower footage being switched in.

Q How should I build a vocal booth?

I'm having a small vocal booth built within an open-plan office to record voiceovers for games and DVDs. The main reasons for building a booth are to keep the sound of the air conditioning and the PC out of the mic, and to reduce the background noise of office staff charging past on their way to buy sandwiches! I think the higher fequencies will be OK, but lower frequencies, where male speech is pitched, travel well through this building.

I'm told that the studio in our last office had a floor floated on sand. It had a two-door atrium to cut out noise but it was small, with a tiny rectangular booth and a larger listening room about 8 x 10 feet, rectangular but with an undulating ceiling. The walls were plasterboard on a wood frame, hung with fabric stuffed with Rockwool. The same builders are available, but if anything we have less space here in the new office — the total available area is 18 x 10 feet. But we do have a floor that lifts so we can make a floating floor flush with the rest of the office.

So, what shape should a tiny booth be? Should the mic face into a corner, and is Rockwool behind fabric enough acoustic treatment or would some corrugated shaped foam on the walls help? We're making it free-standing to please the landlord. Should we just float it on neoprene to isolate it from the surrounding floor? As for the listening room, will foam provide better diffusion than Rockwool? Does the shape matter in such a small room or will the lower frequencies go right through the walls unheard? Or should we not have a booth at all, and instead create one larger room and isolate the sound of the PC using a cabinet? The benefits of booths seem to be debatable...
Jon Chan

Editor In Chief Paul White replies: I think that if your main business is voice recording, then a vocal booth would be desirable, and as voices have relatively little low-frequency content, you shouldn't get too many mode-related problems in your main control room as long as you put in some basic acoustic treatment. Without a booth you may have problems isolating the vocal from unwanted sounds, but you're also right to be suspicious of booths as badly made ones can sound dreadful.

Vocal booths need to be very well treated to avoid them sounding boxy, so you need to allow for at least six inches of acoustic treatment on the inner walls and ceiling. A plasterboard and studding construction with a double-glazed door is fine in most applications of this type and you can float the walls of the box on neoprene to add some isolation from the floor. The inside floor can be created by using high-density Rockwool slab and then resting chipboard directly on it with a felt or rubber buffer between the edges of the floor and the booth walls. Two layers of chipboard, glued and screwed, work better than one. Filling the space between the inner and outer plasterboard skins with Rockwool insulation will help improve sound isolation, as will using two or more layers of plasterboard on the outside of the booth. I'd be reluctant to make the finished internal size of the floor of the booth less than around 4 x 5 feet though it needn't be rectangular. There's no easy or cheap way to supply a constant, quiet air flow to such a booth while maintaining isolation, so opening the door regularly between takes is probably the most pragmatic solution.


When building a vocal booth, foam can be used to tame high-frequency reflections, while reintroducing some reflective surfaces (in this case old CDs) can counteract an excessively dry sound.

A practical way to treat the inside is to use more high-density Rockwool (a minimum of 30mm thick) spaced away from the plasterboard walls by at least a couple of inches but ideally four inches. You can do this using wooden battens for support as high-density Rockwool is quite rigid. This air gap will reduce the boxiness that you may have experienced with your original design. Put fabric or visually attractive acoustic foam on top of this to contain any loose fibres, then use a perforated MDF sheet (the type used for covering radiators) to cover some of the side walls at head height as this will allow some high end to be bounced back into the booth. Without some high-frequency reflections, the sound may be too dry, but you can always add more HF reflectivity later by putting up more perforated sheet if necessary. During our Studio SOS visits, we've previously used old CDs taped to the walls at head height to improvise some high-frequency reflection. If you have room to hang a heavy curtain or rug behind the speaking position this will also help kill reflections. Normally the artist will use a cardioid mic and be facing the glass in the door so no significant reflections will reach the mic. It doesn't really matter whether there's a flat surface or a corner behind the vocalist as long as the rear and side surfaces are mainly absorbent.

For the main room, you'll need to place absorbers at the 'mirror points' at the very least. Again Rockwool slab spaced off the wall and suitably covered works well as does sticking acoustic foam to Rockwool or using thick foam spaced off the wall on blocks. In a room that size you'll probably find that a square metre of absorber above your listening position and on the walls at either side will make a big difference. You may also need absorbers on the wall behind you, especially if it is fairly close, otherwise shelves and furniture may scatter the sound adequately. If the walls are solid, you may also need some bass trapping, which can be accomplished by building more Rockwool slab traps across the corners or between the walls and ceiling. The triangular space behind these can then be stuffed with Rockwool insulation to improve performance but even without this you should get good results. If the budget allows, Real Traps Mini Traps are good in this application and save a lot of DIY work.

If you search our web site, you'll find a lot of practical articles on acoustics and soundproofing that may be helpful to you. SOS

Steve Howell

Dave Smith PEK

Four-voice Analogue & Digital Synth

The Evolver began as a mono desktop unit, then became polyphonic, and now there's a polyphonic keyboard version. It's almost as if Dave Smith's latest synth is steadily evolving back into his earlier Prophet 5...! We check it out.

As has been explained in these pages before, Dave Smith's *resumé* is quite a read. A major innovator in the hi-tech music world since the mid-'70s, he designed the first truly programmable synth, Sequential Circuits' Prophet 5, as well as the monophonic Pro 1, Prophet 10, the gorgeous T8, and many others. He is credited as one of the fathers of MIDI, having helped to design and promote this now-ubiquitous interface. He was also the creator of Vector synthesis in the Prophet VS (later to blossom into the Korg Wavestation) and was developer of the world's first proper software synth, Seer Systems' *Reality*. Having just *one* of those achievements on my CV would make me very happy!

Evolving Evolvers

But despite having developed software synths, Dave is today back to making hardware products in the form of his Evolver range. First off the line was the Evolver, a monophonic synth in tabletop form reviewed in *SOS* February 2003 (see www.soundonsound.com/sos/feb03/articles/davesmithevolver.asp). This was followed by the Poly Evolver rack (see *SOS* December '04, or www.soundonsound.com/sos/dec04/articles/dspolyevolver.htm). The Evolver uses a 'matrix' method of programming, where you select a line of parameters to edit on the matrix with the buttons at the side and then adjust that line's parameters with the eight encoders at the top of the matrix — it's a capable, but fiddly interface. The Poly Evolver (arguably) took a backwards step with a 16x2 LCD and an 'up/down' page system (though you can use a mono Evolver to program it, or a PC/Mac software editor). That's the Evolver story so far, but now everything has changed with the arrival of the Poly Evolver Keyboard (or PEK), which seeks to combine the best of both previous versions and improve on them, being polyphonic, fitted with a keyboard, and, as you can see above, covered in many more controls than graced the front panel of the original Evolver. There are rotary encoders, illuminated switches and blue LEDs — it really is a classy thing to look at. There are even wooden end cheeks, and although I know they make no contribution to the sonic quality of the instrument, they do add *something* to the 'experience' of owning and playing it.

What sets the Poly Evolver Keyboard apart is that panel — a knob or switch for (almost) all the functions makes this powerful synth considerably easier to use than the other Evolvers, and most of the everyday functions are typically just a knob or switch away with other, less commonly used functions being available on menus shown on the 2x16 LCD (but even those use encoders to select or set their values).

The panel is very well laid out. The main business area (oscillators, filter, envelopes, effects and output parameters) is right in front of you, directly above the keyboard and follows a logical progression from left to right. Furthermore, good labelling makes it perfectly clear how the signal flows through the instrument, especially with regard to showing the various feedback paths that occur in each voice.

Other functions, such as Envelope 3 and the four LFOs, are found at the top left of the panel and the four step sequencers are to be found at the top right. Dominating the centre of the panel is the programming area, where the 512 sounds (four banks of 128 sounds each) can be selected using the plus or minus buttons, two data encoders or the large keypad. The instrument's three different modes of operation (Combo, Program and Global) are also selected here (see the box overleaf for more on this).

At the rear, the PEK sports a multitude of audio outputs. There is a headphone output, a pair of main stereo outputs which carry a mix of all the voices, and there are also individual (stereo) outputs for each of the voices. Inserting a cable into these (balanced) quarter-inch jacks removes that signal from the main stereo outputs, as it should in my opinion. There is also a stereo audio input allowing external signals to be processed through the PEKs filters and effects. Naturally, it has MIDI, plus a 'special' MIDI/DIN connector for daisy-chaining other Evolvers to expand polyphony. As the PEK is more biased towards performance than its counterparts, there are two variable pedal inputs (assignable) and a sustain pedal input.

Disappointingly, power is supplied by an external PSU — on a keyboard of this nature (and price), I was expecting a sturdy mains cable and built-in power supply. In fairness, it's not a wall-wart type — the PSU is connected to the mains with a cable of a decent length — but it was still a surprise when I removed the PEK from its packaging. Disappointing as well was the omission of a digital audio output, which I would have thought would make the PEK more flexible in a modern studio.

The final item on the rear panel is another for the 'eye-candy' department: four bright blue LEDs that pulsate according to the rates

The PEK's lightweight velocity- and aftertouch-sensitive keyboard might not be to everyone's taste, but it's perfectly adequate for the synth sounds on offer. The PEK also sports two transparent wheels to the left of the keyboard, which glow a beautiful blue; in subdued lighting, the PEK is a thing of beauty (see the final page of this article). Even my wife, who is usually quite unimpressed with the various bits of gear that pass through our doors, was very taken with the PEK's cosmetic design, calling it 'the ultimate big boy's toy'! And she was absolutely right...

Overview

I won't spend too much time on the synthesis functions available to you (although I may allude to them by way of my description of the panel) as these have been ably described in Paul Nagle's previous reviews of the Evolver and Poly Evolver, but to summarise, Dave Smith's products are a potent marriage of digital and true analogue technologies in one instrument. Each voice has two digital 'vector' oscillators and two analogue oscillators per voice which are fed to an analogue low-pass filter (the digital oscillators by way of a digital-to-analogue converter) and an analogue voltage-controlled amplifier. From here, the signal comes back through an A-D converter to be further integrated with the digital audio processors.

This means that you have the best of both worlds: software LFOs and envelopes, four step sequencers, effects, sophisticated modulation control, precise control of parameters, and loads of memories to store sounds thanks to the digital side of the synth, plus the warmth and character of analogue. The PEK's voice structure is unchanged from that of the other Evolvers, so for more details, head for those older *SOS* reviews.

SOUND ON SOUND

Dave Smith Instruments PEK
£1800

pros
- Sounds great, with unique and innovative sounds.
- The knobular front panel makes the complex synth engine quite manageable, even easy to use!
- Looks fantastic in almost every respect.

cons
- External PSU (no matter how many times we see 'em, we still don't like 'em...).
- Four-voice polyphony does seem a bit limiting these days.

summary
The Poly Evolver Keyboard isn't cheap, and it is only four-voice polyphonic. But it is one hell of a synthesizer in the true sense of the word, and it's delivered in a very desirable no-compromise package.

of the four LFOs. These must have been included for visual purposes, because they serve no practical purpose whatsoever otherwise — but they do look very nice! I guess it's only a matter of time before synth nerds in the audience will be guessing which patch is being used based on the respective rates of those LEDs...

Control Centre

At first glance, the PEK's control panel looks pretty straightforward (and indeed, for the most part, it is) but I was surprised to find continuous encoders for all the controls. I am in two minds about the use of these in this context. Of course, they allow a control movement to take immediate effect from the parameter's stored value, but as you are adjusting them, you have no idea when you are reaching the control's 'end stops' (so to speak) and I found myself referring to the LCD (which, like Novation's K-Station, shows the value of the selected parameter's value) far too often for such a knobby synth.

Furthermore, some of the parameters have quite a large range, and so some of the encoders require a few turns to cover that range. The Filter Cutoff control, for example, requires a few turns to create a full manual filter sweep, and the Resonance, LFO Rate, and Filter Env Amount controls, as well as others, seem to require more turns than we're perhaps used to.

In defence of this method, however, apart from immediate 'take-up' of a parameter's stored setting, it also allows a finer degree of accuracy when setting values, and that has to be applauded. I guess it just takes a bit of getting used to after using conventional 'pots' with physical end-stops at seven and five o'clock. It would have been good, perhaps, if the encoders were 'velocity sensitive' — in other words, if quick movements allow coarse settings whilst slower movement allow more precise settings. Also curious is that the encoders caps have pointers, which are largely meaningless on totally continous controls. On a more positive note, 'zipper noise' (ie. stepped digitised value changes) from the controls was minimal to the point of being inaudible.

Reservations about the encoders aside (which might well not bother anyone else — it's something of a personal gripe), there is generally a dedicated control or switch for each function, which is great. There are a few compromises — there's only one set of controls for the the four oscillators, the LFOs and the step sequencers, for example — but illuminated switches allow you to select the 'module' you want to tweak very easily. But these compromises are understandable, as it would be unfeasible to provide dedicated controls for all of these.

In most other respects, this is very much a 'no-compromise' design — Dave Smith could easily have tucked a few of the dedicated functions away on a menu-based programming system, and the PEK could probably survive without some of the 'eye-candy' like, for example, the master octave LED indicators, the pulsating blue LEDs for each of the four LFOs (especially those duplicated on the rear panel!), and the large dedicated program number display. The PEK could also have shared common controls for the three envelopes, and instead of having dedicated switches to select oscillators, LFOs and sequence banks, it could have used one (frustrating) button to cycle through each of those. But Mr Smith has obviously put function and usability before form (and issues of cost) to create a knobby user interface that is simple but comprehensive and puts most functions right in front of you. I am not denying that there are few areas (notably modulation) where you have to figure things out (and here, a larger LCD might have have helped) but overall, this is generally a gratifying synth to use despite the programming depth and complexity available to you.

Around the back, all seems complex, but it's all quite logical. Each PEK voice gets its own stereo output, so in addition to the headphone jack and main stereo outs, there are a further eight jacks (two per individual voice), plus a stereo pair for the audio inputs, then the power connector, connectors for sustain pedal and two continuous footswitches, the *de rigeur* MIDI trio (plus an extra connector for daisy-chaining other Evolvers and Poly Evolvers), and finally, those completely gratuitous (but very impressive-looking) blue LEDs!

King Of The Mode

The PEK has three main operating modes: Program, Combo and Global. Global is the most utilitarian of these, dealing (as it usually does) with overall tuning, MIDI settings, sequencer clock, external audio input gain, LCD contrast and the like.

Of the other two options, Program mode is the simplest: you play or sequence a single four-voice sound, and have 512 sounds to choose from. You can also stack all four voices in a Unison mode for truly dangerous sounds, and there are various playback-triggering modes.

Combo mode combines several Programs to provide simple four-part multitimbrality. Many of the pre-programmed Combo memories are complete musical pieces in themselves that utilise the four sequencers playing each part separately. Others stack Programs or create keyboard splits. Some are keyboard-triggered sequences with separate Parts running in different octaves, at different clock divisions and at different sequence lengths. Others are 'free-running' quadra-timbral sequences over which there is no control other than by moving the front-panel knobs relating to the currently selected part — transposition of these from the keyboard is not possible, which is a shame. It's also worth pointing out the four-voice limitation here — each of the four parts is, of course, strictly monophonic.

Combo mode is by far the most entertaining, as it can be used to realise complete compositions. However, the limited voicing means that such arrangements must be limited to four parts. That said, the presets show off what's possible and are quite inspirational (over-use of distortion notwithstanding), ranging from the 'innocence' of early multitracked synth work in the style of Tangerine Dream and Kraftwerk to examples which owe more to four-on-the-floor techno. There is no denying that the Combo mode throws up many interesting musical ideas that might not be apparent when working with a more conventional multitrack sequencer.

Sounding Off

How does it sound? Fabulous for the most part! I am a bit at odds with some of the presets, many of which attempt to sound 'modern' simply by slapping on distortion rather too often for my liking, but this is easily rectified by, well, turning down the Distortion

Software Vs Hardware — The Long View

Sometimes it feels as if software synths are gradually displacing their hardware forerunners, and might one day cause them to disappear from studios altogether. However, there are still many people who prefer hardware, and it would seem that Dave Smith is one of them. Despite having developed the world's first software synth (Seer Systems' *Reality*), he has reservations about software instruments.

Interestingly, his objection is not so much one of sound quality, but more to do with the long-term 'value' of software. As he told *Mac Music* in an interview in 2003, although there's a multitude of software synths available today, with seemingly every college student having designed one, he believes that software synths are a bad business model, requiring constant (and therefore costly) updates and testing under different operating systems with different hosts to ensure that they continue working: "a constantly moving target", as he called them. As a result, his belief is that the majority of these swiftly designed software instruments won't still be working in 10 years' time, because the sheer amount of work required to keep them up to date will weed out all but the most successful of them. This is apparently why he prefers to design hardware products — because he feels that they are intrinsically more reliable and will still be of value in the decades to come.

Whether or not that's true, it certainly holds for the reliability of his PEK — I had not a single problem during the review period. And such is the depth of the PEK's programming potential, I feel certain that this synth will still be a source of musical inspiration 10 years from now.

knob. Many of the presets show off the instrument's broad capabilities, from genuine (not modelled) analogue warmth through to biting digital textures. The unique Feedback section, in particular, takes you into territory not normally found on other synths and the panel legending shows clearly how this interacts with the sound. I am also especially pleased to see that Dave Smith has not 'compromised' the digital oscillators by refining them to remove aliasing, and even the manual proudly proclaims that like the Prophet VS, they sound 'quite trashy at higher frequencies', a 'feature' which only serves to add more character to sounds.

Such is the depth of the PEK's synthesis capabilities that many of the presets explore territories inaccessible to other synths, notably long and (yes...) evolving sci-fi textures that could have come straight off the soundtrack for *Forbidden Planet* or from the depths of the BBC's Radiophonic Workshop in the late '60s and early '70s, when they had just taken delivery of their EMS VCS3s and AKSs. Many presets reward you with abstract, atmospheric soundscapes that slowly develop over time. At the other extreme, however, the PEK is equally capable of underpinning a track with strong basses and lush pads, or dominating it with strident lead lines, spikey arpeggio textures and more.

In short, this is a powerful, capable synth that has a place in almost any genre of music, and with a distinctive character that sets it apart from other synths on the market. There's only one criticism really; in this day and age, it's hard to overlook the fact that it is four-voice polyphonic, a spec last seen on keyboards in the days before Dave Smith had even designed the Prophet 5. There's no way to increase this polyphony with, say, voice expansion cards, although you can daisy-chain multiple Poly Evolvers to achieve this.

Whether this is an issue for you is up to you to decide. Personally, I found it to be a restriction, limiting even the use of an octave bass and a simple triad (a total of five notes) for pads and the like, and more ambitious endeavours are out of the question without note stealing becoming obvious. I would have loved eight voices to play with, but I suppose this would have added yet more to this product's already fairly substantial price.

But to whinge about polyphony is, to some extent, to miss the overall point of the PEK, in much the same way that it is churlish to complain that the Minimoog Voyager is monophonic. The PEK is a well-built and classy-looking instrument which is equally at home in the studio or on stage. The hands-on, knobby control panel makes programming it much more manageable than the company's other Evolvers but (thankfully) there is still that beguiling element of serendipity, whereby a twist of a control can yield an unexpected result. If you like hardware and you like keyboards, check out the PEK — it's a deep, unique-sounding synth with vast sonic potential. I think you could explore it for years to come and still not exhaust its store of surprises. SOS

It'll look great on stage... the PEK in reduced lighting conditions.

information

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M-Audio Microtrack 24/96

Digital Stereo Recorder

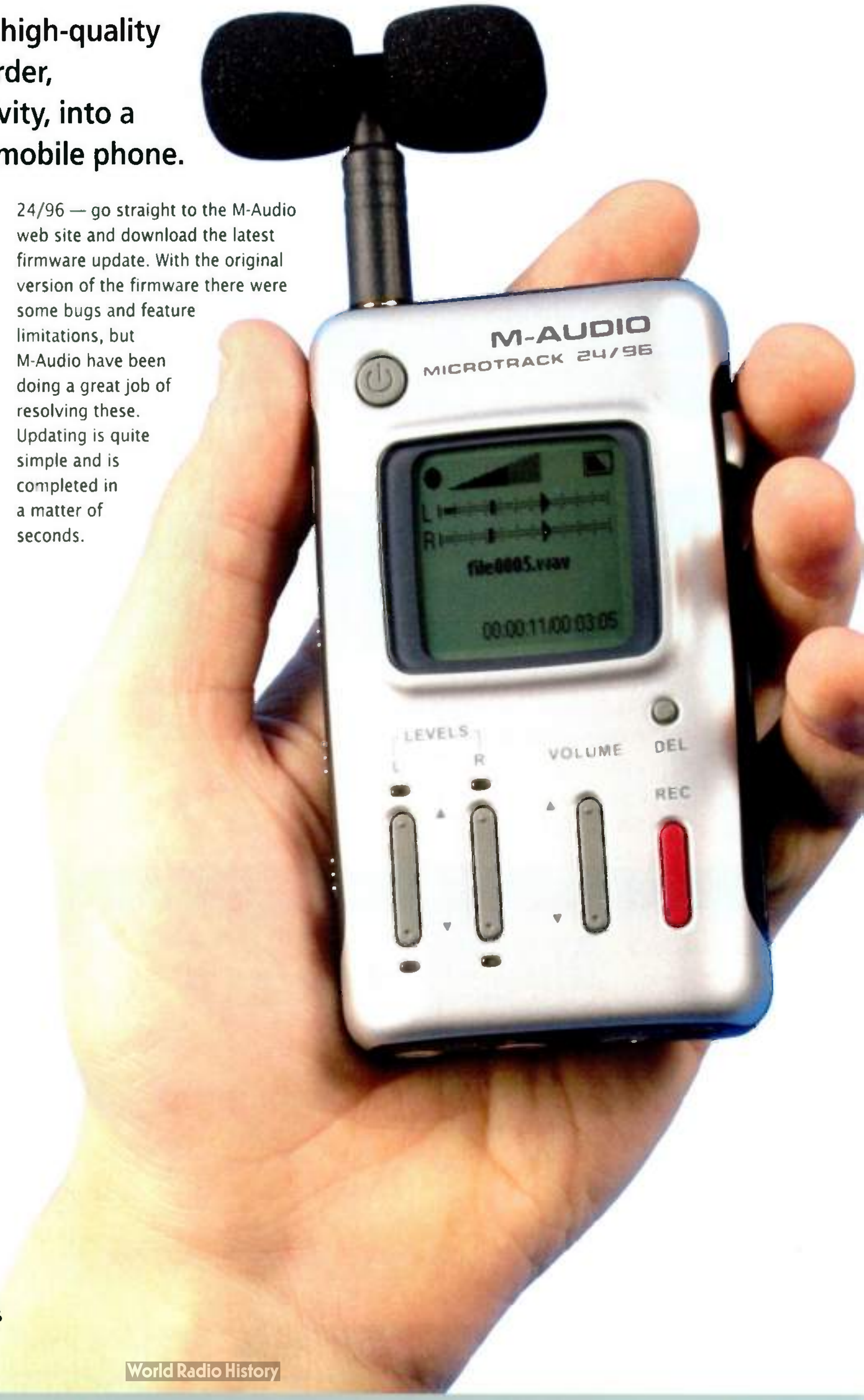
The new Microtrack packs a high-quality 24-bit/96kHz two-track recorder, complete with USB connectivity, into a box not much larger than a mobile phone.

Matthew Moore

Despite being not much larger than a mobile phone and weighing in at just under 150g, the new Microtrack 24/96 is anything but light on features. Audio connections include a stereo mini-jack mic socket, two quarter-inch TRS mic/line inputs, and a coaxial S/PDIF input. There is phantom power available on the quarter-inch sockets and 5V power on the eighth-inch socket to feed a bundled stereo 'T' mic. On the output side, there are unbalanced phono sockets and a headphone mini-jack.

The Microtrack 24/96 can record stereo WAV files at 16-bit or 24-bit resolution (and 44.1kHz, 48kHz, 88.2kHz, or 96kHz sample rates), or MP3 files at 96-320kbps, and these recordings are stored onto a removable Compact Flash card inside the casing. Power comes from an included power supply or from an internal long-life lithium-ion battery, and the specifications state recording times of approximately four to five hours on a single charge, reducing to about three hours when phantom power is engaged. You can charge the battery either via the power supply or by connecting the Microtrack 24/96 to a PC or Mac via USB. M-Audio have obviously taken note of Roger Thomas's Sounding Off column back in *SOS* December 2005, as the Microtrack also comes with a nice black bag with a drawstring!

There is one thing you should definitely do before anything else with the Microtrack 24/96 — go straight to the M-Audio web site and download the latest firmware update. With the original version of the firmware there were some bugs and feature limitations, but M-Audio have been doing a great job of resolving these. Updating is quite simple and is completed in a matter of seconds.



Test Spec

- Microtrack 24/96 firmware v1.2 beta.
- 2.8GHz Pentium 4 PC running Windows XP SP1.
- 1GHz Apple Mac G4 running Mac OS v10.3.9.

You will also want to charge the unit fully to extend the battery life, probably using the supplied PSU. This is of the ubiquitous wall-wart variety with two pins, although sliding the unit apart exposes a figure-of-eight socket which allows you to connect a suitable UK mains lead.

The supplied 64MB Compact Flash card was already formatted on my unit, but if you purchase another, larger card (pretty essential if you want to do any serious amount of high-quality recording) you will need to format this — again a pretty simple job. Note that not all Compact Flash cards are compatible with the Microtrack, and the M-Audio web site has a list of approved ones.

Using The Microtrack 24/96

The Microtrack 24/96 is nothing if not versatile: during the review period I made two different location recordings with it, used it as a sketch pad for song-writing, transferred non-audio files from a PC to a Mac, took it on the train as an MP3 player, and even made the tea with it. (Ok, so I lied about the tea.) However, despite its flexibility of use, I was struck from the outset by how easy it was to use.

Once you have booted the machine, you simply select which input you wish to use and which file type, press Record, set your recording levels, and start recording. It's worth noting that pressing the Record button actually starts the machine rolling, so if you want to pause whilst you set levels, you will need to press the little scroll wheel on the right-hand side of the machine, pressing it again to take you out of pause mode. One of the joys of this unit is that you can press record at any time, no matter where you are in the operating system's menu structure, making it a very quick process to start recording. In addition, I could boot up the Microtrack 24/96, plug in the included mic, set a level, and be recording within 40 seconds — doing the same with *Cubase*, including setting up a mic and stand, took nearly one minute and 45 seconds. This could be the difference between catching the inspiration for that number-one single, or losing it.

The Microtrack gives each file/recording a sequential number, and at present there is no way to name these, or indeed to reset the numbers, without doing a default reset of the unit. A pen and paper are most useful for keeping track of your recordings. Who can remember in a month's time whether file 0094 was really the best take? Once you have recorded, you can either listen back via the unit or transfer files to a computer. In the latter case, if you connect the Microtrack 24/96 up to a computer via the USB 2 cable, the recorder's Compact Flash card appears on the computer's desktop as an external drive, and files can be transferred quickly to your hard drive.

SOUND ON SOUND

M-Audio Microtrack 24/96 **£349**

pros
- Can record 16-bit and 24-bit WAV files at sample rates up to 96kHz. Also records MP3 files at a variety of different resolutions.
- Easy to use.
- Very portable.
- Lots of applications.

cons
- Preamps are a little noisy.
- Phantom power not the full 48V.
- Firmware still needs some revision.

summary
Despite a few teething troubles, this is an extremely versatile and portable little stereo recorder which is very easy to use and can deliver professional results.

Listening back to recordings direct from the Microtrack's headphone jack was hampered because the fast-forward, rewind, and scrub facilities were a little erratic. The unit would scrub forward for maybe 10 seconds and then the display would freeze and it would take a few seconds to work out where it was. Several attempts on the same file resulted in arrival at a number of different places! Sometimes it would be further into the track as expected, at other times it would be in the next track or possibly back at the beginning of the selected track. This proved quite annoying when scrubbing through long tracks — finding a moment 20 minutes in, for example, proved almost impossible. In each case I ended up transferring the files to my PC and auditioning them there. This is definitely something M-Audio need to resolve.

I was initially somewhat disappointed with the preamps on the Microtrack, as they were a little noisy for my liking. The mini-jack socket and included mic were particular culprits. However, for instant capture of ideas and for documentary purposes I don't see this as a problem. The balanced preamps are better (but still not as quiet as those of my M-Audio 1814, for example), although I don't feel that super-quiet preamps can be expected for a product at this UK price. I recorded a solo piano recital with a stereo mic pair, and found the resulting file needed a little noise reduction to make it usable. However, a pair of Rode NT5s and 27 players from the London Philharmonic orchestra (with a much larger sound than a solo piano!) proved a very nice combination. The detail was good and I was set up in about a quarter of the time it normally takes me to set up my laptop and interface. I also felt confident that I could just leave the Microtrack to get on with it's job whilst I worried about the front-of-house sound — not something I can always confidently do with my laptop.

Whilst on the subject of connecting microphones, it's worth mentioning that, although the unit can indeed provide phantom power, it does so at 30V rather than the usual 48V, and this may preclude the use of certain microphones. M-Audio's web site provides a list of microphones which will work with this lower supply, although this is certainly not exhaustive. For example, neither the Rode NT4 nor the Rode NT5 are included, but the NT5s certainly make an excellent partner for this device — I would imagine the NT4 would also. However, it is fairly easy to find out the phantom-power needs of a mic from the manufacturer or from www.microphone-data.com.

Mighty Micro

I feel that Compact Flash recorders will become more and more popular as time goes on — already the BBC are using them to record the Proms! Although the cards are little pricey at the moment (a 2GB card offering 3.5 hours of stereo 16-bit/44.1kHz will cost about £80-100 at the moment), ▶

M-AUDIO MICROTRACK 24/96

The Microtrack 24/96 records direct to a Compact Flash card installed via a slot in its side.

prices are dropping all the time and I imagine cards with sizes comparable to hard drives will be available in the not-too-distant future.

I've tried the Microtrack in a number of different roles, and found it incredibly easy to use and simple to set up. If I were looking to do more critical classical recording with it, I'd probably pair the unit up with an Apogee Mini-Me (or similar mobile preamp) via an S/PDIF cable, and still have a stereo setup that's smaller and less hassle than my laptop. make the unit more complicated to operate. At present you can't monitor the S/PDIF input, which might be a bit of a problem if you are using external preamps. Although you can often monitor from such units directly, I prefer to do it via the recording device, as you never know for certain that signals are getting to it correctly. 'Well, it sounded alright at the preamps' might not wash on a professional session! M-Audio have informed me that they are looking into a solution to this problem, and in the short

Although I haven't made many ventures into the 96kHz arena yet, having the facility to do so is a real plus point for me. The ability to use the unit as a memory stick for non-audio file transfer is also very handy, and it makes an excellent backup device as part of a location recording setup. As I mentioned earlier, I shall certainly be turning to the Microtrack 24/96 to capture inspiration when writing, as it is much quicker to set up than either my desktop or laptop computers in this respect. It reminds me of the good old days with a Dictaphone!

I do feel that there is room for development with the unit. I would like to see a little more detailed metering, with figures rather than just lines. Limiters on the inputs would also be quite useful. The ability to name files might make life easier, but might also time I've been using this unit they have been beavering away updating the firmware, including new features and improving the way the unit works. Who knows, they might eventually include the tea-making facility!

At this UK price, it's hard to find anything with exactly the same features, and if you are looking for a small, convenient recorder that does any or all of the jobs above I would thoroughly recommend the Microtrack 24/96. SOS

information

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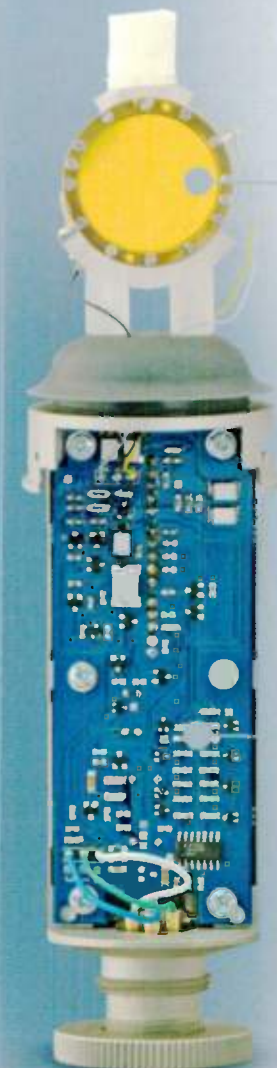

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Fostex NX6A

Several unusual new technologies have been combined in this new product from Fostex to create an accurate mixing tool.

Hugh Robjohns

This intriguing new product from Fostex is priced within a few pence of the manufacturer's similar PM1 model, reviewed in *SOS* November 2003, which seems a little odd from the marketing point of view. However, the new speaker boasts several interesting features (all denoted by snazzy acronyms) designed to give it the sonic edge.

First, however, let's look at the basics. This is an active two-way design built into a compact 11-litre reflex cabinet with a pair of ports on the front baffle. The actual dimensions are 350 x 230 x 334mm (hwd) and each speaker weighs about 13kg. The driver complement consists of a 6.5-inch magnetically shielded doped-paper woofer and a 0.75-inch metal-dome tweeter with a magnesium diaphragm. The substantial discrete solid-state amplifier chassis on the rear panel provides a nominal 60W for the woofer, and another 40W for the tweeter.

The tweeter is protected behind a wire grille and is set into the front baffle at the back of a shallow waveguide which ensures time-alignment with the woofer and smooth high-frequency dispersion. The very unusual woofer design is what first catches the eye, though, looking like it is intended to juice soft citrus fruits! In fact, this is Fostex's HR (Hyper Radial) cone design, which is claimed to be extremely rigid while also being lightweight — the ideal combination required to provide very low bass distortion figures. Equally important is the way the cone is attached to the driver chassis, and we have another innovative design with its own acronym here too. This is the UDR Tangential Edge, which is said to eliminate unnecessary resonances. UDR stands for 'Up/Down Roll' and describes the very odd-looking twisted effect of the surround.

We're not done with the acronyms yet — there is another hidden away inside the cabinet. Standing waves inside rectangular speaker cabinets have always been a problem, and various techniques have been devised to deal with them. In the case of the NX6A, Fostex have combined long-fibre wadding with what they call HP Sound Reflectors. These appear to be multi-faceted inserts — one on the rear wall and one at the base of the cabinet — which are intended to scatter the sound internally, preventing standing waves (and hence resonances) from forming. The complete speaker is claimed to enjoy a frequency response of 55Hz-38kHz at the -6dB points, and boasts a maximum sound pressure level of 104dB/W at one metre.

The rear panel is equipped with three

Photos: Mike Cameron

Active Nearfield Monitors

SOUND ON SOUND

Fostex NX6A £699

pros

- Attractive UK price.
- Unique woofer design.
- Tight and detailed sound.
- Good stereo imaging.

cons

- Slightly forward character can become fatiguing.
- Lean bass slightly misleading.

summary

A well-built and well-priced active monitor. The slightly forward character and lean bass won't appeal to all tastes, but you do get a detailed sound stage which facilitates accurate mixing.

rotary controls: Volume, Lo EQ, and Tweeter Level, and two input connectors — a balanced XLR (+4dBu nominal sensitivity) and an unbalanced quarter-inch socket (-10dBV). The latter takes priority if both are connected. The Volume control turns the signal all the way down to silence, which is useful. The Tweeter Level control has a ±3dB range and simply adjusts the gain of the appropriate amplifier. The crossover frequency is unusually high, at about 7kHz, so this control really only affects the brightness of the top end. The Lo EQ is a peaking equaliser centred on about 55Hz with a ±3dB range again, and is intended to provide some compensation for boundary effects when placing the speaker close to rear walls or corners. The only other rear-panel facilities are the IEC mains inlet and rear-panel power switch. An LED on the front baffle illuminates when the speakers are powered.

Listening Tests

I set the NX6As up as usual alongside my reference three-way PMC IB1s — not a fair comparison, I grant, but one which provides a useful benchmark reference point for me. My initial reaction was of a fairly forward character in the mid-range, with a very tightly controlled, but rather dry, bass and a well extended and neutral top end. The response changes fairly smoothly off-axis, but it is clear that these speakers have a fairly narrow dispersion. At 30 degrees off axis the response was almost 10dB down by 6kHz, so aiming the speaker directly at the listening position is important.

On a range of familiar listening material, the NX6As gave a very stable and precise stereo image, with plenty of mid-range detail and a punchy-sounding bass. Unlike many ported speakers, these monitors don't overdo the bottom end at all — they appear over-damped if anything, which gives the subjective impression that there isn't a lot of weight or depth. However, as on the PM1s, it is easy to separate out bass instruments in the mix and judge their balance with accuracy, which is how monitor speakers are supposed to work.

The only thing that troubled me slightly was the forward mid-range — something it shares with the PM1s — and looking at the published frequency response this appears to be caused by a broad peak extending between about 1kHz and 3kHz, reaching about +3dB relative to the rest of the response. In fact, below 1kHz, the response extends as an almost flat line down to 60Hz before falling away smoothly at about 18dB/octave. Above 3kHz the response is equally flat up to about 9kHz, above which it falls in a series of broad peaks and dips, but is still only about 10dB down at 40kHz.

Returning to that mid-range peak, it is something of a double-edged sword. On the one hand it provides an impression of lots of detail and ambience, and of being able to look into a mix. On the other hand, it has a slightly 'shouty' quality that some may find fatiguing after long mixing sessions.

In a typical home-studio environment the NX6As are capable of playing very loud without obvious compression effects until silly volumes are reached. The spectral balance also remains consistent once you get above a certain threshold — as with most reflex speakers, the bass seems to disappear when the volume is turned well down. Given the UK price, the NX6As are impressive and well-built, visually distinctive, and capable of a fine, revealing sound stage that helps rather than hinders the mix process. SOS

information

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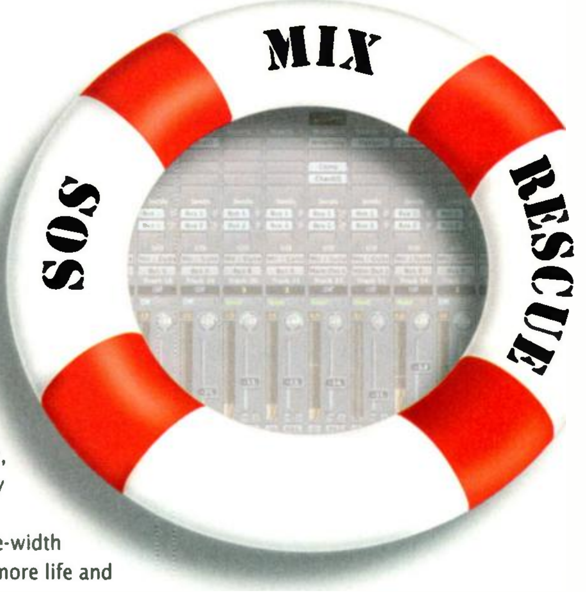
Mix Rescue



Kick drum and lead vocal present tough processing challenges in this month's remix.

Paul White

The song receiving the Mix Rescue treatment this month is 'Time', performed by the band Panic Number and recorded in a home environment. In many ways a conventional pop/rock type of mix, it is centred on drums, bass, and guitar, with a strong lead-vocal part and dense backing vocals during the chorus. The instrumentation and style are somewhat reminiscent of early Genesis, but with a little less prog-rock pomp.

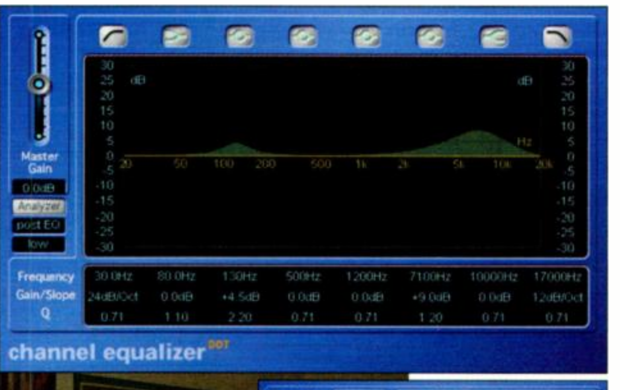
Where it becomes more complicated is that multiple layered guitar parts have been used, along with multiple layered backing vocals. Layering in this way is a well-proven textural device, but it makes mixing a little trickier, as it's very easy for the whole mix to get clogged up with texture, leaving no room for the song to breath. An interesting element of the backing-vocal section is a four-part choral effect (augmented by a straight four-part backing vocal), which I enhanced using a little subtle processing. There is also a dense, pulsing synth part that worked well musically, but sounded a little static to my ears, so I used a simple flanger plug-in to create a kind of pulse-width modulation that gave the part more life and definition.

Sound Source Problems

After scrutinising the original tracks, I also identified several 'sound source' problems such as some vocal pitching issues, dull vocal tonality (which also changed in the middle of the song), and a certain amount of low mid-range room tone on the vocal track that caused the vocals to sound somewhat congested. According to Andy Cope, who handled the recording, they hung a duvet behind the mic to help deaden the room acoustics, but they might have done better hanging it behind the singer, as this is where most of the problematic reflections come from. The drums were recorded cleanly over several tracks, but the kick drum had no depth of tone — a problem I've often found when tackling remix jobs — so once again I had to resort to processing trickery in order to give it the necessary weight.

As usual, I tackled the entire project in Apple's *Logic*, as it's a sequencer I'm used to working with, but I utilised plug-ins from a number of different sources, and the vast majority of the techniques explored here could be replicated using any decent sequencer and plug-in package. I didn't use any pitch-fixing technology here, as the usual 'quick fix' plug-ins didn't seem to make a big subjective improvement to the lead-vocal sound. A few hours' work with something like Celemony *Melodyne*, could probably have sorted out the pitching quite seamlessly, but I think the band would be better off re-recording the lead vocal from scratch, as they should be able to get a much better sound.

To make handling the mix easier, I subgrouped the drums, backing vocals, and guitar parts to their own stereo busses. While there's little point in making



The drummer's recorded snare sound was not as crisp as it could have been, so Apple *Logic*'s EQ and gating plug-ins were used to tighten it up.

On the original recordings, the kick drum sounded very lightweight, and quite heavy processing was needed to sort this out. PSP Audioware's *Vintage Warmer* gave the sound extra density, while *Logic*'s *SubBass* plug-in added in some low-end weight. The *Noise Gate* was also used to reduce the level of spill on the kick-drum track.

over-critical sound adjustments to isolated tracks (because they always sound different when playing together), it can be valuable to try to fix some of the more obvious problems one track at a time, and then to fine-tune your settings in the context of the final mix. You can also set up sub-balances of things like the various backing-vocal parts, as these may be harder to hear when they're part of a busy mix.

To my ears, the band's original mix was guitar heavy and a little cloudy sounding, but it certainly had plenty of energy, and the general quality of playing was good. The distorted guitar parts all had similar sounds, and three of these had been doubled up to produce a richer effect, so balancing all seven guitar lines turned out to be a bit of a juggling act.

Remedial Work On The Kick Drum

I usually start out a mix by setting up a solid foundation of drums and bass, but before I could do that here I needed to sort out that lightweight kick-drum sound. Having tried EQ and compression, I resorted to using *Logic*'s *SubBass* plug-in to generate some low end. This seems to be becoming a habit, but it's not something I've needed to do before embarking upon the Mix Rescue series. The trick is to add just enough sub-bass to fill out the sound without exposing the trickery, and in this case I ended up with what felt like a very solid and satisfying kick sound. In addition to the sub-bass generator, I also gated the kick to reduce the amount of spill by around 12dB, and then used PSP Audioware's *Vintage Warmer* plug-in to add a bit of analogue-style saturation to give the sound more density. This wasn't overdone and worked well once the rest of the kit was added in.

The rest of the kit was recorded cleanly with separate close mics and stereo overheads, though the toms seemed to be having trouble cutting through the busy mix, so I used *Logic*'s *Exciter* plug-in to give them a bit more edge, but nothing too drastic. Other than using some high-end 'air' EQ on the overheads, and rolling out some low end, there was only gentle polishing required. I used some gating and EQ to make the snare drum sound tighter and crisper, but again I didn't do anything too drastic to the original drum sounds other than the kick drum. I did, however, apply further processing to the whole drum submix using the Noveltech *Character* plug-in from the TC Powercore arsenal. I opted to use that in combination with a lesser amount of overall EQ, as that seemed to give the most solid and lively kit sound, which cut through the mix without sounding too hard or edgy. If I knew exactly what this plug-in did, I'd tell you, but although I know what effect it produces and I can handle the controls, I've no idea what's going on inside the algorithm! I consider it to be a 'more of everything' effect.

Something else I've also discovered about 'home recorded' acoustic drums, including those recorded in my own small studio, is that they have no real sense of space, so in addition to the usual ploy of adding a little reverb to the snare and perhaps the toms, I also like to put the whole drum track through a well-chosen small-room ambience treatment, in this case a convolution-based drum-room ambience courtesy of *Logic Space Designer*'s Garage preset. Adding just enough ambience to take the dry edge off the sound makes the drums sound more believable and lively without cluttering the sound. A pro-studio recording of this type of material would probably see the drums set up in a nice-sounding live room, so that's the effect I was going for. For 'conventional' reverb, I set up a short, bright plate using the Universal Audio UAD1's *Plate 140* plug-in on send one, and also added this to the main- and backing-vocal parts.

Bass & Guitar Processing

There was nothing wrong with the bass-guitar part, but I gave it a little more focus by putting it through *Logic*'s *Bass Amp* plug-in and then added some compression using *Logic*'s own compressor, just to keep the level even. That made it sound a bit on the forward side when soloed, but in the mix it sounded fine. Soloing the drums and bass at this point revealed a much more cohesive-sounding foundation for the track.

I like to get the mix to the point where it sounds more or less believable with all the faders left static, after which I can use ▶

Rescued This Month...

The band Panic Number are (clockwise from bottom left) Andy Howard, Alex Moon, Chris Marsland, and Andy Cope. Based in Somerset, they are already playing live dates, and are currently putting together a first album. Influences include Elbow, Oasis, Muse, Soulwax, Björk, Radiohead, Placebo, Jeff Buckley, Vex Red, Doves, Supergrass, Starsailor, Incubus, the Beatles, the Smiths, Queen, and Coldplay.

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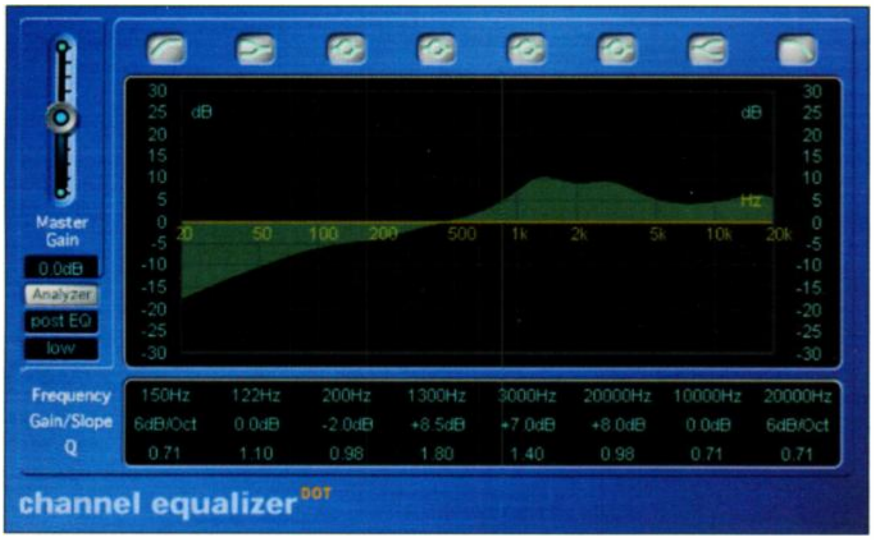


The lead-vocal part caused the most problems during the remix, and radical EQ was required to deal with the original's clouded sound.

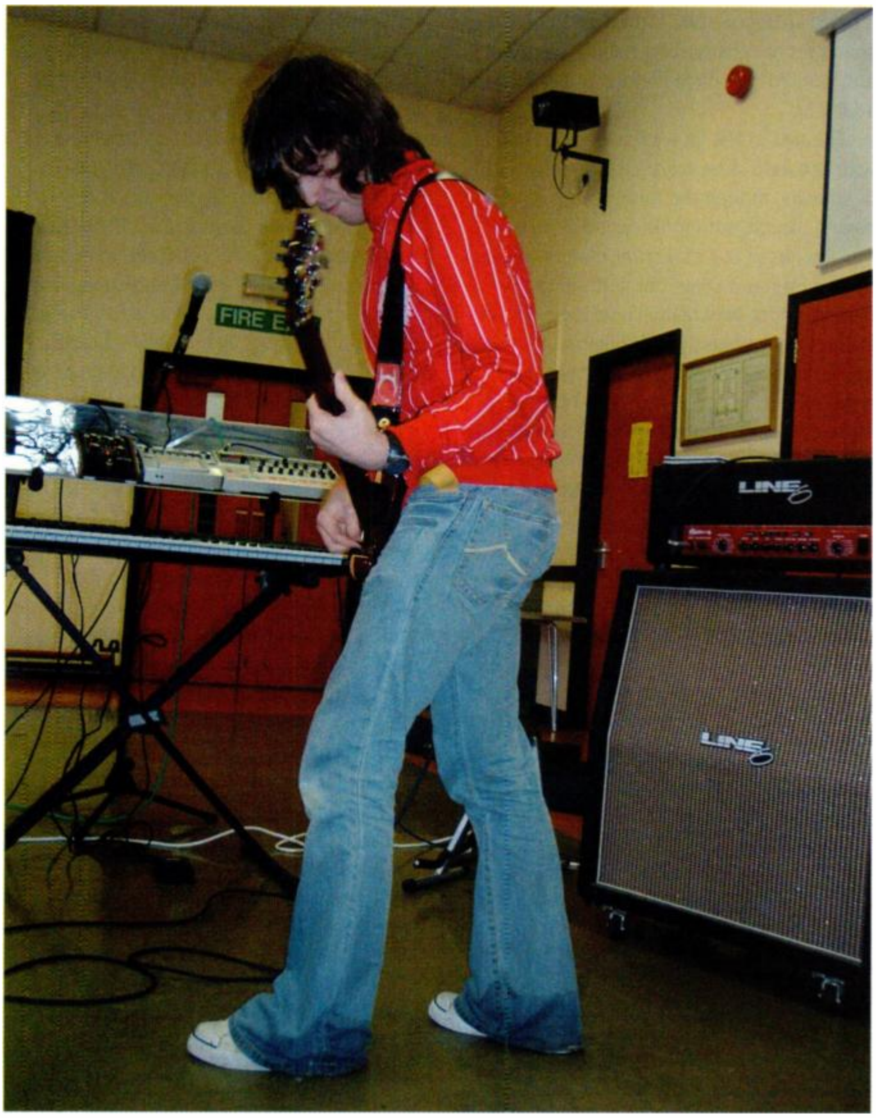
automation to bring out or suppress certain elements. In this case the guitar parts needed the most attention, as they all employed a similar overdriven tone and were pretty 'full on' a lot of the time. I rather liked the 'wall of sound' effect this produced, but various parts needed to be dropped or lifted to keep the guitar out of the way of the vocals and also to reinforce key parts in the song where the guitar riff is important to the song dynamics. The song opens with a layered guitar part to which a phaser effect had previously been applied. I panned these tracks to create a wider image, but otherwise left them pretty much alone, using just a little high and low EQ cut to focus the sound. I ended up automating the vocal level too, as it tended to sound over-loud in the quieter sections of the song.

The backing vocals comprised two parts, each of four layers, where one part was fairly choral in nature, with all four voices singing 'aah'. To enhance the choral effect, I used the UAD1's *Dimension D* plug-in to add a subtle chorus-like sheen, but otherwise used only a hint of reverb. The *Dimension D* processing created a very believable ensemble sound to the layered 'aah' parts, lending a nice vintage feel, which I thought suited the song well.

Salvaging The Lead Vocal

Although I was worried about handling all those guitars, it was the lead-vocal part that eventually gave me the most trouble, and I had to resort to extremely radical EQ to get it sounding closer to the way I felt it should have been in the first place. Even then I wasn't entirely happy with it, but without re-recording it I could have spent hours working on it and might never have managed to get it sounding better than I did. The vocal had a rather congested sound, as though the singer was recovering from a cold, and there was also a sense of small-room boxiness to the whole vocal track. More absorbent material in the recording room, behind the singer, would probably have helped.

To help find a good vocal EQ strategy, I set up *Logic*'s *Match EQ* plug-in and used a solo recording of a singer with a similar-sounding voice to give me my target audio spectrum. Applying this to the vocal in question showed that a lot of low end needed to be cut, and a significant mid-range boost needed to be applied to arrive at a similar spectrum. Rather than use *Match EQ* to do this, I replicated the general shape of the correction curve using *Logic*'s *Channel EQ*, and found this produced a very workable result with only a little further tweaking. However, as the vocals seemed to get even duller in the second half of the



A large number of different guitar parts were recorded through the band's Line 6 Flextone II modelling amp. However, in order to create a 'wall of sound' effect, many of these parts were quite similar, and therefore needed careful balancing and automation during the mixdown.

song (possibly because the vocal track had been compiled from several takes), I had to automate the EQ boost and level to compensate.

Once I'd fixed the vocal EQ and added some compression, it sat nicely in the mix and felt much more confident than before. The UAD1 *Plate 140* reverb seemed appropriate for the '70s style of the song, but I also snuck in a little TC Electronic *Classicverb* ambience to give the vocals a stronger sense of place, and to help thin down that small-room boxiness.

Three different reverbs were used in the final mix. A *Space Designer* drum-room convolution patch was applied to the drums submix to give more of a live sound, while Universal Audio *Plate 140* and TC Electronic *Classicverb* plug-ins provided more traditional plate and ambience treatments which were used more generally across the whole mix.

Finishing Touches

Other than using some overall EQ on the backing-vocal submix to add 'air' in the 10kHz region and to roll off the low end below 200Hz, I used the *Character* plug-in once again to add definition to the guitar submix, but combined this with high-frequency cut to stop the overall guitar sound from being too edgy and to keep it out of the way of the cymbals. That done, I used *Vintage Warmer* once again on the overall mix to add some gentle compression and the faintest hint of a 'smile' EQ curve. I also used its in-built limiter to keep a check on the signal peaks, allowing me to keep the average level fairly high. The big knob in the middle of the plug-in's interface adjusts the amount of 'analogue' drive, and has to be used very sparingly if the effect isn't to become too obvious, so I used just the barest hint.

Overall I was pleased with the general feel of the mix, which had a nice, homogeneous '70s sound to it, rather than being ultra-separate and clinical. As usual, I kept the vocals, bass, and kick drum in the centre of the mix and then spread out the other parts to inject some sense of stereo width. With doubled-up guitar and backing-vocal parts this is easy to do by

Hear The Differences For Yourself!

Judge the changes to 'Time' for yourself by checking out the following audio examples I made during the session — they can be found at www.soundonsound.com/sos/mar06:

- /audio/OriginalMix.mp3

This is the Original Mix as sent in by the band.

- /audio/OriginalAahs.mp3

Original four-part backing vocals.

- /audio/ProcessedAahs.mp3

Backing vocals with reverb and a *Dimension D* chorus effect.

- /audio/OriginalKick.mp3

Original kick-drum track.

- /audio/ProcessedKick.mp3

Kick-drum track processed using gating, sub-bass generation, and analogue emulation.

- /audio/OriginalDrums.mp3

Drum mix using unprocessed original tracks.

- /audio/ProcessedDrums.mp3

New drum mix with individual and global processing applied as described in the article.

- /audio/OriginalSynth.mp3

Original synth track.

- /audio/ProcessedSynth.mp3

Synth processed using *Logic*'s flanger to create a pulse-width-modulation, chorus-like effect.

- /audio/OriginalVocal.mp3

Original vocal with no processing.

- /audio/ProcessedVocal.mp3

Final vocal processed with EQ and compression, and with plate reverb and room ambience added.

- /audio/Version1Remix.mp3

Remixed version of the track.

- /audio/Version2Remix.mp3

Remixed version of the track with an additional flanging effect applied to the lead vocal.

Classic Tracks

Bob Marley & The Wailers
'I Shot The Sheriff'

Artist: Bob Marley & The Wailers
Track: 'I Shot The Sheriff'
Label: Island
Released: 1973
Producers: Chris Blackwell, The Wailers
Engineers: Phill Brown, Tony Platt
Studios: Tuff Gong, Basing Street

Bob Marley and the Wailers were the first Jamaican musicians to achieve world stardom. Tracked in Kingston and finished in London by Island engineers Phill Brown and Tony Platt, their breakthrough album was a truly international recording.

Richard Buskin

Starting as a tape-op at Olympic Studios, London, in November 1967, Phill Brown was initially trained by such industry notables as Keith Grant, Glyn Johns and Eddie Kramer while working with artists like the Rolling Stones, the Small Faces, Traffic and Jimi Hendrix. Not a bad start. In 1970, after having built Toronto Sound, Canada's first 16-track studio, with his brother Terry, Brown then became a house engineer at the newly opened Island Records facility on Basing Street in Central London, where he initially worked with outside clients and stayed until going freelance in 1976. By then his credits included Harry Nilsson, Jeff Beck, Led Zeppelin, Robert Palmer and one Robert Nesta Marley, who as a member of the Wailers had first worked alongside Brown on the band's second Island release, the 1973 album *Burnin'*.

Released just six months after the Wailers' major-label debut *Catch A Fire*, the Burnin' album combined some old material originally produced by Lee 'Scratch' Perry back in 1968 with new compositions such as 'Get Up, Stand Up', 'Burnin' And Lootin' and 'I Shot The Sheriff'. These melded the funky reggae beat with lyrics about spirituality and the Jamaican struggle with poverty and violence. Accordingly, the record, featuring lead vocals by Marley, Peter Tosh and Bunny Livingston, was essentially a call to arms, and 'I Shot The Sheriff' became its best-known song. As written and sung by Marley, the main protagonist admits that he did 'plant a seed' and gun down head-hunting Sheriff John Brown in self-defence, but pleads innocent to all charges of killing the deputy.

While Eric Clapton's relatively pale 1974 cover of the song would rise to the top of the US charts, the original did at least earn Marley worldwide fame and help his name precede

Phill Brown at work in the early '70s.

Photo courtesy Phill Brown

the group's when Tosh and Livingston subsequently departed to pursue solo careers. Indeed, by far the most powerful version of 'I Shot The Sheriff' appeared on Bob Marley & the Wailers' superb *Live!* album in 1975, which Phill Brown captured at the Lyceum Ballroom in London, but it was the one recorded inside Basing Street's Studio Two that helped pave the way for Marley to transport his name, his music and his message to all corners of the globe.

Inside Basing Street

"In Studio Two, from '70 to '73, we had the first Helios desk, originally built for Olympic by Dick Swettenham," Brown recalls. "In fact, Island was all Helios, as was the mobile studio, the Stones' mobile and the Who's Ramport. The original Helios in Studio Two was 24 in and eight out, whereas when they brought in the next model in late '73, early '74, it was 32 in and 16 out. The 24-input desk was what we used for the *Burnin'* album, and it was brilliant. Its basic EQ was fantastic, fixed to 10k for the top end and 50Hz for the bass end, with four or five mid-range frequencies, and it had a beautiful sound. What's more, it had immediate mic amps — you plugged a mic in and there it was in front of the speakers instead of back somewhere in the distance.

"Helios, Cadac and Trident were all very similar during that era — just beautiful desks — yet they didn't have a lot of effects: two echo sends, I think, and maybe two foldbacks. That's all we had, so mixing was always about plugging in particular sounds from particular tracks. We didn't have aux sends where you could send every channel to all kind of things, and we therefore had to think ahead about what we were going to do. The Helios was a lovely desk for two people to operate in a mix environment on 16-track. Once it got to 24-track, things got hairy, but 16 was perfect.

"Along with the in-board EQ, I used the Urei 1176 for compression on the vocals or any other recordings that we did — they were pretty much the only compressors we had apart from some Helios ones — and EMT echo plates as well as the old Eventide digital delays. That was pretty much our classic setup during that era. A lot of what we did was down to miking technique and getting sounds in the room that we liked or we wanted. Rather than change it later, you had to be pretty close to what you wanted to have on the record.

"We used largely the same mics from session to session. I'd stick [*AKG*] D20s on guitar amps — I still do — and use 87s for backing vocals, 67s for lead vocals, D20s or FET 47s on amped keyboards... I guess we had our favourites, especially 87s and 67s during that era. Then again, I also know that

Photo: Ian Dickson / Redferns

Tony Platt and Richard Digby Smith — who worked at Island as an engineer — both fancied [*Shure*] 57s and the old [*Electrovoice*] RE20s. They weren't the conventional mics, but they were getting great results. And although I had my favorite mics for certain things, if you saw the mics used by the different engineers on the Bob Marley sessions, they weren't all the same."

The monitors in Studio Two were 15-inch Tannoy Red drivers inside Lockwood cabinets suspended from the ceiling, while the tape machines were a 3M 24-track, 3M 16-track and an Ampex eight-track that was used to make transfers.

"The desk, tape machines and speakers were all housed inside a pretty small control room — maybe 15 foot square, 20 foot square — where the engineer would sit right up against the wall with the desk in front of him. It would be very un-hip today. And the studio itself would hold, at a push, 15 musicians if it was a string session, or otherwise a five- or six-piece band. It was an easy room to work, whereas upstairs [*Studio One*] could accommodate a 60 piece line-up and was a much harder beast to manage. It needed either acoustic intruments, including strings, or a heavy rock band to make it work. On the other hand, the Studio Two control room was fantastically accurate and it became a very serious mixing environment before the computer-aided era."

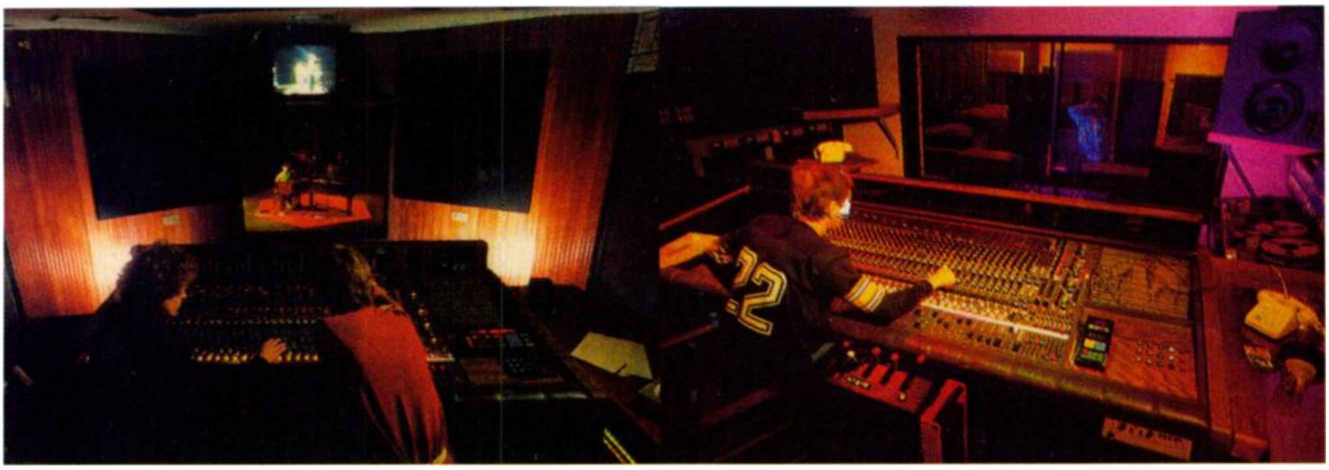
Across The Atlantic

Prior to the sessions, the Wailers recorded their songs — including new versions of numbers such as 'Put It On', 'Small Axe' and 'Duppy Conqueror' that had previously been produced by 'Scratch' Perry — at Marley's own Tuff Gong eight-track facility in Kingston, Jamaica. The tapes were then transported to Basing Street and transferred to 16-track in preparation for overdubs of extra guitars, keyboards and vocals, as well as the mix. This was the procedure that had already been utilised for the *Catch A Fire* album.

"The tapes that they brought over actually contained seven tracks of recordings because track six didn't work," says Brown. "Everything was mono: two tracks of drums — the bass drum and a kit track — a track of bass guitar, a track of Hammond organ, a track of guitar, two tracks of vocals... very basic, but it was decent quality. We'd add maybe an extra guitar, an extra keyboard and try new vocals, but we didn't change very much in terms of the original vibe. We'd listen to the tracks as we transferred them and try to get the sounds to sit in that type of space rather than bring them to a completely different kind of thing. What was done where didn't really come into it, although I for one was definitely trying to make the record more Kingston than Notting Hill Gate."

Phill Brown was, thanks to some prior work with Trojan and Jimmy Cliff, "kind of aware of reggae" when he first became involved with the *Burnin'* sessions. However, by his own admission, he also wasn't a huge fan of the genre and he didn't know a lot about it. For him, it was simply a case of editing and overdubbing the material that he was being presented with.

"In sonic terms I was given a huge rein," he says. "For instance, when I was setting up the mix for the classic *Live!* album, Blackwell would just say 'Give me more audience,' and I was then very much left to do what I'd do. That entire record was mixed in six hours. It was a different era. If I was mixing it today there'd be a lot more people saying 'More bass, more this, more that...' Back then, everything was done really fast and Blackwell seemed to trust the people he had around him. The final call was always Chris's decision, but in those days it wasn't so much about the technology. It was much more important to capture the atmosphere, the energy and the vibe of the track. Especially with reggae and Bob Marley songs — we could have made them very schmaltzy by smoothing them out, but that's not what it was about, and a lot of the records nowadays are very smooth by ▶



Basing Street Studios as it appeared in the late '70s: by this time, Studio One (left) and Studio Two (right) had both been refitted with MCI 500-series desks.

comparison.

"I'd heard some of the Wailers' earlier material, which was more roots reggae — the slower, kind of sleazier stuff — and we did actually speed things up. When we mixed, we'd speed up the 24-track a little bit to make the music more accessible. That was really a Chris Blackwell decision, setting it up for a Western audience; an English, European and eventually American audience. Real reggae was much slower in a sense, more roots sleaze, and we were just making it a little bit more commercial. That's why the overdubs were done."

Editing & Overdubs

In the case of 'I Shot The Sheriff', there were three versions on the eight-track that were all copied to 16-track.

"I remember Rabbit [*keyboard player John Bundrick*] came in and played either electric piano or a Wurlitzer kind of overdub part that was DI'd," Brown recalls. "Everything was very immediate. If you listen to those records, there's not a lot of room sound, as such; just great in-your-face sounds. If it was acoustic piano, we would mic it up with a couple of [*Neumann*] 87s and record it fairly good quality. However, most of the overdubs back then were done with Wurlis, Clavinets, electric pianos — stuff that was much more DI'd, amped up and a little more aggressive. And when the session guys came in we would have them play across three, four, five tracks that we had ideas for rather than work on one track at a time and finish it.

"So, in terms of hard sessions — sessions being three hours — we would probably copy everything from eight-track to 16-track in a three-hour session and then we would bring in musicians for a three-hour session. Maybe somebody like Rabbit would be there for two of those sessions — be there for six hours or of those sessions — be there for six hours or an entire evening — but basically the musicians would come in and we would run the songs by them and see what they thought. Chris would point out the direction in which we might want to go, and we would just record and then off they would go after they'd done their three hours or six hours or whatever. You see, there were specific ideas and plans, but they weren't really pinned down. It was more a case of 'OK, let's get this guy in. He'll be good for this, and while he's here why don't you try him on that?'

"There would have been a pass or two of the keyboards on 'I Shot The Sheriff', and we may then have tried out the odd extra guitar part before moving on to the vocals. All of the songs arrived with vocals already on there, but *Burnin'* was the last time that the whole band was present in London — after that, some of them didn't travel and the line-up changed — and so we recorded a track of backing vocals as well as a new lead vocal by Bob Marley. A [*Neumann*] 67 or 87 on vocals was pretty much the norm of the day, so that's what I used for Bob, and everyone else went through an 87. Again, it was a case of

Bringing Reggae To The World

Few people did more to bring Jamaican music to a wider public than Island records boss Chris Blackwell, who co-produced most of Bob Marley's '70s output along with lots of other reggae classics. "Blackwell was one of those guys who was very good at putting like-minded people together in the studio, and he was always there with encouragement: 'Yeah, this is good. Let's move on to this,' or 'Oh yeah, that's feeling great. Why not try this?'" explains Phill Brown. "However, he wasn't producing the record in the sense of a modern pop production where somebody would be there to direct every moment. He came and went. And when he went, we knew what we all had to do, so if we were in the middle of vocals and Chris popped out for half an hour, we'd just carry on and Bob, the musicians and I would make decisions like 'Yeah, that feels good. Let's go with that verse.'

Photo: GEMS / Redferns

"Chris was really coming at things from both a commercial and a musical angle. I've known him a long time and I guess his greatest love is Jamaican music and that whole approach. I mean, he's given a lot of people a huge number of chances — he's been one of the great mavericks of the music business. And with all of us back then, he'd allow us to do our thing. He was making commerical decisions, yes, in terms of speeding things up or asking for more audience in the live mix, and they were well judged, but these were all based on a music that he really loved.

"Okay, so the real die-hard reggae enthusiasts of that era may have been really pissed off because certain tracks were sped up — there was an article in the *NME* where the guy figured out that two versions of the same piece of music were 13 seconds different in length and seemed to be in a different pitch. However, it wasn't the Bob Dylan as Judas thing, and at the time it appeared that the Wailers were willing partners in all this. My memory is of everyone being there during the overdubs — I'm not sure that was the case when we were mixing. That was more down to just Chris and I. But I don't recall anyone complaining. It was later on that you'd hear through the grapevine that maybe one of the guys wasn't pleased. At the time it wasn't a problem, it was like 'Let's tweak that up by a few percent.'"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just capturing the moment and capturing the vibe of the guys rather than the finesse of a modern recording.

"The small room in Studio Two was very live, so we had a couple of screens up with an 87 just to make it a little bit more cosy. The control room was on the other side of a window, so visually there was very easy contact, and we just recorded with the meters to the 16-track machine. Of course, at the time of those sessions the Wailers weren't as big as they would become, but my main memory and image of that era is of these guys, these quite hard characters, who had a real attitude. Obviously, they were kind of street guys — not sweet locals, but really cool characters who were writing music that really meant something, and when it came to the vocals Bob got out there and just kind of did what he needed to do.

"He sang, and it was very easy for him to do stuff, so we didn't do a lot of takes. In fact, we never did in those days. There were probably a couple of passes, a couple of versions and one would be chosen. It was not a long process. Everything was happening quite fast. Bob very much knew what he was after, and so did everyone else because he was not so much the leader of the band in terms of telling them what to do. They were all aware of what they were doing and he knew exactly what he needed to do.

"Comping did take place, but this was before people got heavily into that kind of thing. I mean, it was done during the early '70s, but you were more likely to have had two takes of vocal and switch between them in the mix according to what you wanted. We hadn't really got to the point of doing three, four, five, six vocals and mixing down. In fact,

Photos courtesy SARM

Basing Street Studios is now SARM West.

we were always concerned about frequencies and sound, and so we never wanted to mix down too much because of the loss in quality. This would later result in better tape machines becoming quite common, but back then, if there were two vocals by Bob Marley, it was quite likely that they'd get switched in the mix rather than bounced beforehand."

Following the keyboard and vocal overdubs, and prior to Aston Barrett recording more lead guitar, some 16-track, two-inch editing had to take place. After all, there were three versions of 'I Shot The Sheriff' on tape and Blackwell wanted to use the first part of one version spliced together with the second part of another version. Thereafter, the aforementioned guitar was overdubbed and the song was mixed along with some other tracks.

At The Mix

"We would maybe mix three songs a day," Brown remarks. "Again, it was a fairly fast turnaround, and Tony might well have mixed the songs again the following week. It was that kind of approach. Chris would leave me to set something up and people would be floating around all the time; dancing and spliffing and partying. Chris would keep checking in, and once he heard something that he thought was what he wanted, he'd say 'OK'. If something needed to be guided into shape, he'd sit down with me and say 'Let's try a bit more of this, try that,' but most of the time people were floating about, listening and having a good time, and if something felt really good we'd put it down. It was much more relaxed than it probably would be today.

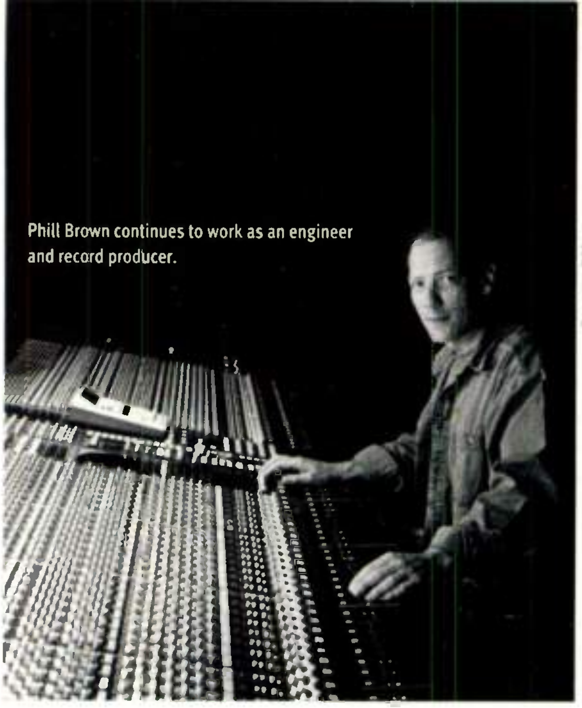
"Since the tracks had been honestly recorded, with a pretty clean, straight sound on the tape machine, the effects that we used — and they were very limited during that era — were just an EMT echo plate, maybe a bit of spring reverb, some Urei compression, perhaps a little bit of delay and ADT on the backing vocals, all done in the mixing environment. Looking back, it's amazing that we made those records with such limited gear. It really was very basic, and yet it all sounded great even if manual mixing brought its own quirks to the table.

"We had to be creative, and if we wanted to have something that was really pretty wild — something that would be easy today to dial up on some machine — we had to work out how to do it. If we wanted the sound of somebody in a church, we basically had to go there and record them. We couldn't fake it. And in the studio we fed a lot of stuff back through Leslie speakers and distorted stuff through amps. We had to move things on. In fact, we could distort the desk very easily — that was a great device that we used at times. You could get a great fuzz guitar out of the Helios desk by overdriving the line, which you can't really do on modern desks because they just take too much input. These desks didn't, so there were kind of quirks that you could work with."

The overdubs for the *Burnin'* album were completed within three or four days, and a

Photo Brian Jahn

Bob Marley's studio, Tuff Gong, is still an active and popular recording venue in Kingston.



Phill Brown continues to work as an engineer and record producer.

similar amount of time was then spent on the mix.

"There was the odd spin, the odd effect or a bit of phasing to push home a particular point, but by and large the Wailers records consisted of all solid sounds," asserts Phill Brown, whose other engineering and/or production credits include Paul Kossoff, Steve Winwood, Talk Talk, Dido and Santana. "There was nothing trippy or psychedelic or anything like what 'Scratch' Perry was doing with dub reggae. It was like recording a rock band. There were five different instruments, a handful of tracks, and there it was: solid, really cool, it had the vibe, and there was no trickery.

"You can bring up any of those tracks and the attitude of the playing is fantastic. We were just adding a few extra bits to that and giving it a little bit of gloss, I guess; giving it a bit of an English sound and making it something more than had it just been finished off in Jamaica. And while some people may have preferred the Wailers' sound when they were with 'Scratch' Perry, Bob stuck with Chris Blackwell through many, many albums and Chris did him a lot of good. At least he reached more people thanks to that approach.

"There was this real honesty in all of the stuff that Bob Marley did. It was really what he was about. That was him. And while a lot of people may have made money off the back of it all, it was his music and his message that really came through. That's why he was such an energy and a force. You felt that when he was wandering around the studio. Even then, before he was massively famous, you felt that this guy had something." SOS

The Party Atmosphere

The idea that an album had to be recorded by a single engineer had yet to take hold at Island Records in the '70s. "Tony Platt and I both engineered the sessions at Basing Street — we were house engineers, and due to the way that studios worked in those days one of us would be on the session one week and the other would be on it the next. That meant we never necessarily followed projects through from start to finish, and while Tony and I were the two guys on the Bob Marley sessions at that point, if we weren't around there was always Phil Ault who also did some work with Bob. So it was very much a case of who was available for a particular session.

"During that era it was a bit like a long party at Island. We were all good mates, some of us lived together, and Tony was even the best man at my wedding. We were a very close bunch of guys, and so when I replaced Tony on a session he would fill me in by saying 'We've been doing keyboards this week with Rabbit [*John Bundrick*],' or 'Bob's coming in to do vocals.' We always knew what was going on."

IK Multimedia Miroslav Philharmonik

Virtual Orchestra For Mac & PC

Derek Johnson

Miroslav Vitous's orchestral library was the first of its kind, and more than a decade after its launch, has been reinvented as a self-contained virtual instrument. Is the original still the best?

Miroslav Vitous — composer, virtuoso bassist and founding member of Weather Report — can probably be credited with inventing the idea of a sample library that covers every instrument in the orchestra. Wanting just such a collection for his own use, he found nothing that would do the trick, and in 1993, started doing the job himself. State-of-the-art recording and digitising technology was used: initial sessions were recorded at 20-bit, with 24-bit technology used later in the process. The players of the Czech Philharmonic Orchestra — solo or ensemble, playing with various articulations and expressions — were recorded on stage in Prague's Dvorák Symphony Hall, sitting where they would normally appear in most orchestral situations. The resulting stereo samples required no further panning: instrumental groups added to a mix instantly fit, naturally filling up the stereo field, with the concert hall's ambience adding to the illusion.

Despite being huge and expensive, the collection became the mainstay of many media composers, and was eventually released for Roland and Emu samplers as well as the original Akai. Rather later, it was also converted to Giga format for Nemesys/Tascam's software sampler. Now, however, the trend is for sample libraries to be built into plug-ins, and so the next step on the road for the Miroslav Vitous library is *Miroslav Philharmonik*, driven by the same plug-in technology that also powers IK Multimedia's *Sampletank 2* and *Sonik Synth 2*.

Remake, Remodel

Sonik Synth 2 was created in partnership with Sonic Reality, and it's SR who have developed *Miroslav Philharmonik* from the original sessions, including material not previously released. The resulting plug-in crams 7GB of sample data, totalling more than 1300 instruments, onto two DVDs (the plug-in installer is on its own CD). And the best part? The plug-in costs a fraction of the price of the original library.

Strings, brass and woodwind instruments form the core of the collection, along with a full range of orchestral percussion (tuned and untuned), male and female choirs, pipe organ, Steinway piano, concert harp, classical guitar and harpsichord and even ambient noises — page rustling, musician chat and tuning up noodles. With *Philharmonik*, you can simulate the complete orchestral experience.

The *Philharmonik* box contains everything, including a full printed manual and a poster illustrating instrumental ranges and where on the stage of the Dvorák Hall the musicians were placed during recording. One document that isn't provided in print is the detailed sound manual. But this isn't surprising: comparing IKM's pre-release publicity (and even the user manual) with the final product seems to indicate that *Philharmonik* is equipped with a lot more instruments than were initially planned. The timely PDF on the installation CD reflects this, although it doesn't list every patch: you're provided with the data — stylistic abbreviations and Instrument types — to work out what a given patch will sound like.

The box's one surprise, a USB copy-protection key, is a first for IKM; their other software will follow in due course. Hardware dongles are a real bummer, but I appreciate software developers' need to combat piracy. At least the chosen Syncrosoft system is used by several other software houses, most notably Steinberg, and authorisations from various programs can be consolidated onto one key. Doing so is a bit of a faff but at least you only have to do it once, and at least it means you can install the software on more than one machine (though you can only work on one copy at a time!).

Installation and authorisation also seemed to be a bit convoluted, but again needs to be done once only. It wasn't entirely clear from the documentation what the user is meant to do with the sound library: sticking the sound DVDs in your drive reveals separate installers. There's no further help or info, such as whether this large library can be placed on a disk that's not your main drive. (It can.)

Being based on tried and tested *Sampletank* technology, it's no surprises that *Miroslav Philharmonik* is instantly compatible with any software that supports VST, RTAS, DXi and Audio Units under Mac OS X or Windows 2000/XP. Such comprehensiveness might seem obvious, but not every commercial plug-in developer has grasped the concept. However, there's currently no stand-alone mode. This option is almost essential for anyone wishing to use the plug-in as an audio proof source with a

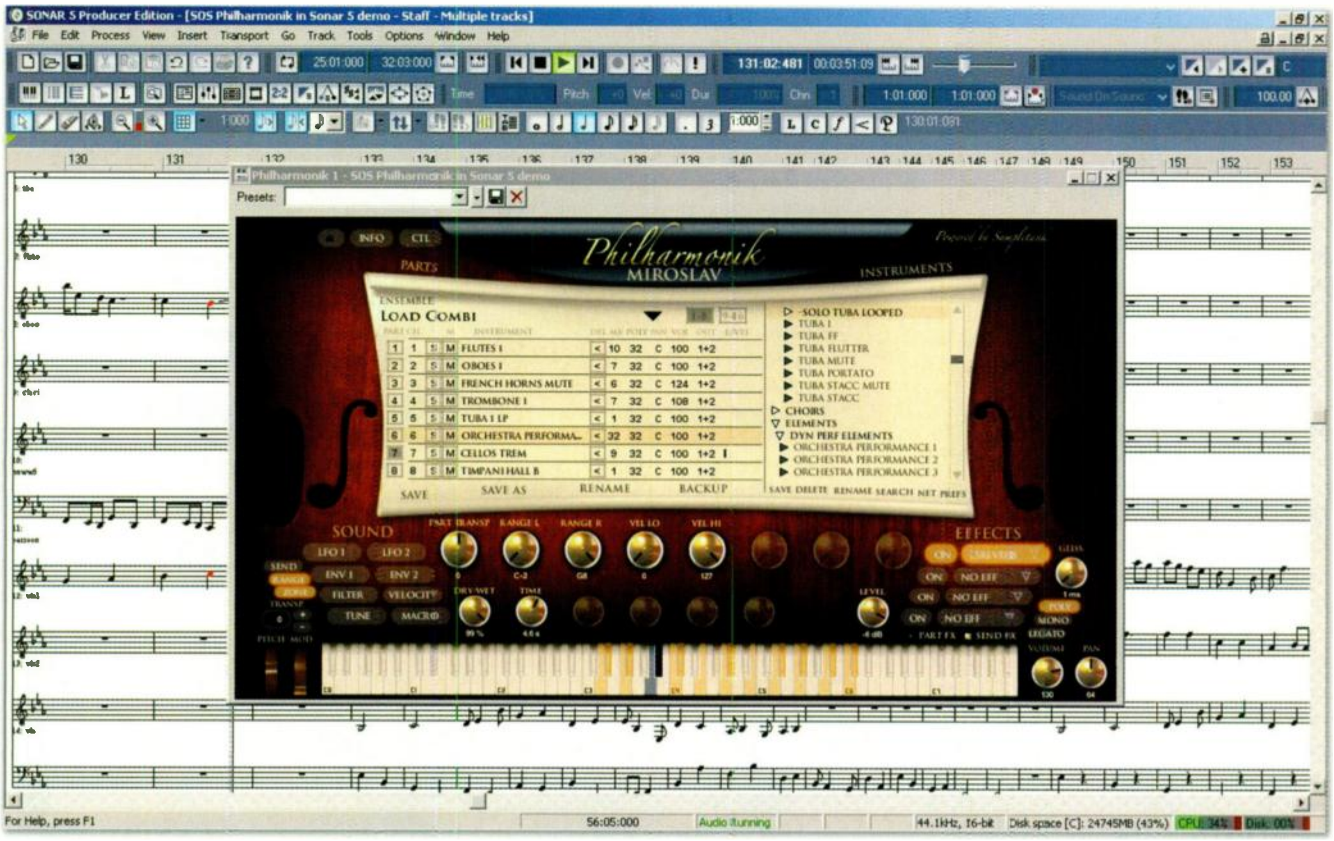
SOUND ON SOUND

IK Multimedia *Miroslav Philharmonik* **£399**

pros
- Great value all-inclusive library.
- Supports all major plug-in standards, Mac or PC.
- Sounds fab.

cons
- Demands a lot of its host computer.
- Keygrouping occasionally not detailed enough in exposed passages.

summary
Extraordinary value for a collection of some of the most playable, well-recorded samples around. I can't stop using it!



Miroslav Philharmonik running in Cakewalk *Sonar 5 Producer Edition*. Note the keyboard display showing off key split zones.

scoring package (few have the audio and plug-in hosting capabilities of full-blown sequencers), or for those of use who wish to offload plug-ins to a separate computer. IK say that stand-alone operation is coming to the *Sampletank 2* family soon.

Look Sharp

Both the look and user interface of *Philharmonik* feel like *Sampletank 2*, only this time with an attractive wood-effect hand-carved look, with scrolling and filligrees. Behind the wood display runs the *ST2* engine offering 16-part multitimbrality (visible in two banks of eight parts), 256-note polyphony, full MIDI control and IKM's latest pitch-handling and time-stretching algorithms.

The central display is also largely identical, bar colour scheme and a slight rearrangement of the file and search buttons. This ground has been covered in *SOS* before, but to briefly summarise, each of the 16 parts in *Philharmonik* is equipped with the same set of parameters. The 'Instrument' assigned to a part has its own slot, and your choice is made using the hierarchical instrument selector to the right. Any parts set to the same MIDI channel are layered; dig deeper and you'll find parameters that let you split or layer parts for true 'combination' creation. There are dozens of combis in the factory collection that put whole orchestras, sections and specialised soloists (complete with switchable performance effects) at your fingertips; sensible use of key ranges and velocity-switched layers makes these examples very playable and surprisingly authentic-sounding.

You're free to constrain the polyphony required by any part; the default is 32 voices, but managing polyphony (ie. keeping it as low as possible) can help the CPU load of the plug-in. One-voice polyphony doesn't always work well, though, even considering we're dealing with an orchestra-full of largely monophonic instruments. Taking advantage of natural decays, glissando effects and so on requires two or more voices to produce convincing results.

Solo, mute, level and pan controls make up the basic mixing parameter set, and audio can be routed to one of 16 stereo busses, providing your host software (or audio hardware) can handle this much audio streaming. There's also a little output-level meter, with further part feedback provided by part numbers flashing in response to incoming MIDI data and a column indicating the size (in MB) of the currently loaded instrument. If it says '0', don't panic: the assigned instrument uses less than 1MB of sample data.

Each part is equipped with four auxiliary sends, each switchable post/pre the level control, and a part can access up to four insert effects, as discussed in the 'Effects In *Philharmonik*' box. If you think 16 parts each running four effects, plus send effects on top, sounds like a recipe for CPU overload, it is. So use the insert effects sparingly if your computer seems to be struggling. And be sensible: reverb, for example, should suit nearly every situation if accessed as a send effect rather than inserted in each part individually.

Orchestral Maneouvres

As sample libraries grow, so they become harder to summarise in any sensible manner. I mean, this library has over 1300 basic Instruments (patches or presets, if you like). Add the factory Combinations and that's a lot of potential material to sift through. Luckily, IKM have provided the plug-in with sensible organisation. Open the plug-in, and you'll see the top level of *Miroslav Philharmonik*'s hierarchy in the browser window: there are folders labelled brass, strings, woodwinds, percussion, choirs, elements and orchestral elements. Some of those headings will be self-explanatory, others initially less so. For example, open 'other instruments' if you'd like to audition classical guitar, harp, cathedral organ, harpsichord and piano patches.

All major instruments, and some minor ones, of all orchestral instrument groups are represented (check out the '*Miroslav Philharmonik* Instruments' box for the full list). I was pleasantly surprised to find contrabassoon, bass clarinet, alto flute, bass flute, flugelhorn and an unexpectedly wide range of percussion in the library — not for Miroslav just bass, snare, timpani and crash cymbals. The full useable range of nearly every instrument is covered, so the lowest note the real instrument can produce is the lowest note you can use to trigger the instrument from a keyboard. Upper ranges are also capped to keep tyros within the boundaries of what is possible in the real world.

As you'll hear if you listen closely, a sample hasn't been created for every note in every multisample. But you'll have to listen closely: the initial samples were made sensitively and the subsequent keygrouping usually doesn't sabotage that work. And for Instrument types where one sample serving a large keygroup would become really obvious (tremolo strings, for example), more samples arranged in smaller keygroups make up the multisample.

The more obscure folders in the browser hide some rather interesting collections. First of all, the 'elements' folder provides patches that are aimed at getting the best out of performance-style combinations. For example, the dynamic performance elements are quite complex 'Instruments' that would normally be created in a Combi. Their role is to take up fewer parts and less polyphony in a combi, offering a faster way to a fuller sound.

At the other end of the spectrum are the 'mono elements'. Here, a 'best of' collection selected from the main library is provided in mono format, again mainly for use in Combis (they take up less RAM and polyphony and can be panned at will). As a bonus, they still sound great and can be used for normal multitimbral sequencing if desired. I find the mono elements Instruments to be particularly welcome when quickly scoring for small virtual chamber ensembles. Many solo instruments in this set are ideal for just this application.

There are several other families of patch that are primarily aimed at keeping the CPU load down during Combi creation, but which work fine in any situation. Special ranges elements offer low- and high-range maps of samples for plonking within specific Combi key ranges. The orchestral sections group, divided into string sections, mixed orchestra and brass/wind sections, provides rich full sound and sonic variety in individual patches.

Other elements collections include the orchestral player noises — coughs, page turns, talking and general hubbub. Last of all, the percussion elements provide individual percussion instruments mapped across the whole keyboard, allowing you to add a particular sound anywhere in the key range within a combi. More percussive variety comes in the form of percussion menu maps, in which different 'articulations' of a single drum are mapped across consecutive keys — different batter and attack types for the bass drum, for example.

The Real Thing

Within the main orchestral instrument sections, you'll find ensemble and solo folders (except for instruments, such as the tuba or bass clarinet, which don't tend to get used in ensembles), and as you move down the hierarchy, you'll find the performance variations — different articulations and dynamics and so on. Thus, string instruments have pizzicato, sul ponte, tremolo and détaché options, while brass and woodwinds have staccato, flutter tongueing and so on. Special effects such as string glissandi, timpani slides and so on are also part of the brew.

Some of the sonic variations — dark, bright, slow or fast attacks — use different sample sets, while others are created through filtering, EQ and enveloping, though the results still usually sound quite natural. Other variants are optimised for real-time control, programmed to work with breath-controller data, aftertouch or expression pedal, for example. A couple of options use the *Sampletank* 'Stretch' engine to tidy up the transitions between keygroups.

Philharmonik's patch organisation follows that of other *ST2* products. The user can't choose a multisample to work on from scratch, but an existing patch can be freely edited. It then becomes a 'child' preset of the original factory 'parent'. There are plenty of factory children, too, which can also be edited and saved.

One thing that becomes almost immediately apparent is that the main Instruments feature unlooped samples. This is obviously the case for staccato and pizzicato samples, but other performance styles feature sample lengths that are often defined by a lungful of air or a single bow stroke. For most applications, this will be long enough, and you have the advantage of hearing what a player can add as the note plays through. Subtle changes in timbre and vibrato and other examples of an individual's taste and style have been captured. Together with the faithfully reproduced attacks (if an instrument speaks slowly, so does its Miroslav equivalent), the result is often an uncanny representation of the instruments sampled.

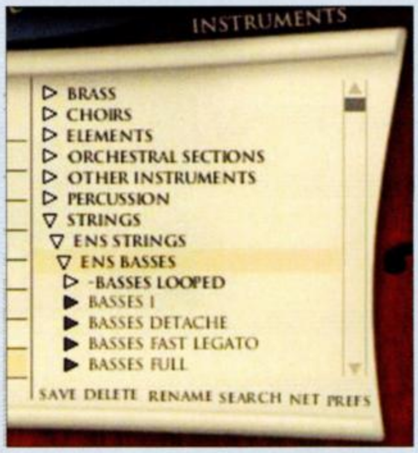
But if you find you need longer notes, there are looped versions of nearly every Instrument on board (percussion are some of the obvious exceptions). Not only do the looped Instruments use less RAM, but their sound is identical to the unlooped varieties — unless you're after the subtle, deeper 'feel' of the longer samples. All samples, looped and

Miroslav Philharmonik Instruments

You'll have a feel for what instruments are provided in *Philharmonik* by reading through the main review. But to make sure you're aware of the full range of instruments sampled, including some of the relative rarities, here's a list. There are actually more 'Instruments' (ie. *Philharmonik* patches) than listed here, because of the inclusion of variations such as solo, ensemble and different articulations.

- **Strings: violin, viola, cello, double bass.**
- **Woodwind: piccolo, flute, alto flute, bass flute, oboe, English horn, B-flat clarinet, bass clarinet, bassoon, contrabassoon.**
- **Brass: trumpet, flugelhorn, French horn, tenor trombone, bass trombone, tuba.**
- **Tuned percussion: glockenspiel, celeste, vibraphone, marimba, crotals, cowbell, tubular bells, plate bells, gong, timpani.**
- **Untuned percussion: agogos, bass drum, bell, bongos, castanets, chimes, cymbals, claps, metal plates, shakers, snare, tambourine, triangle, woodstocks.**
- **Choir: female, male, split, mixed.**
- **Others: harp, guitar, organ, piano, harpsichord, ambient noise.**

In addition, various 'articulations' are provided, though not all apply to all Instruments (there wouldn't be much point in flutter-tongueing a violin or trying to play a marimba pizzicato): crescendo, détaché, expressivo, flutter, glissando, legato, marcato, mute, pizzicato, portato, spiccato, staccato, sul ponte, tremolo and vibrato. Further variety comes with examples that are have faster or slower attacks and release times and are darker or brighter than the main Instruments. Mix and match until you get past 1300 patches!



A close-up of the Instrument browser, showing some main groups and various subfolders. The black arrow indicates a parent patch with one or more child patches derived from it.

Test Spec

- *Miroslav Philharmonik* v1.0
- PC with 3.06GHz Pentium 4 CPU and 1.25GB RAM, running Windows XP.

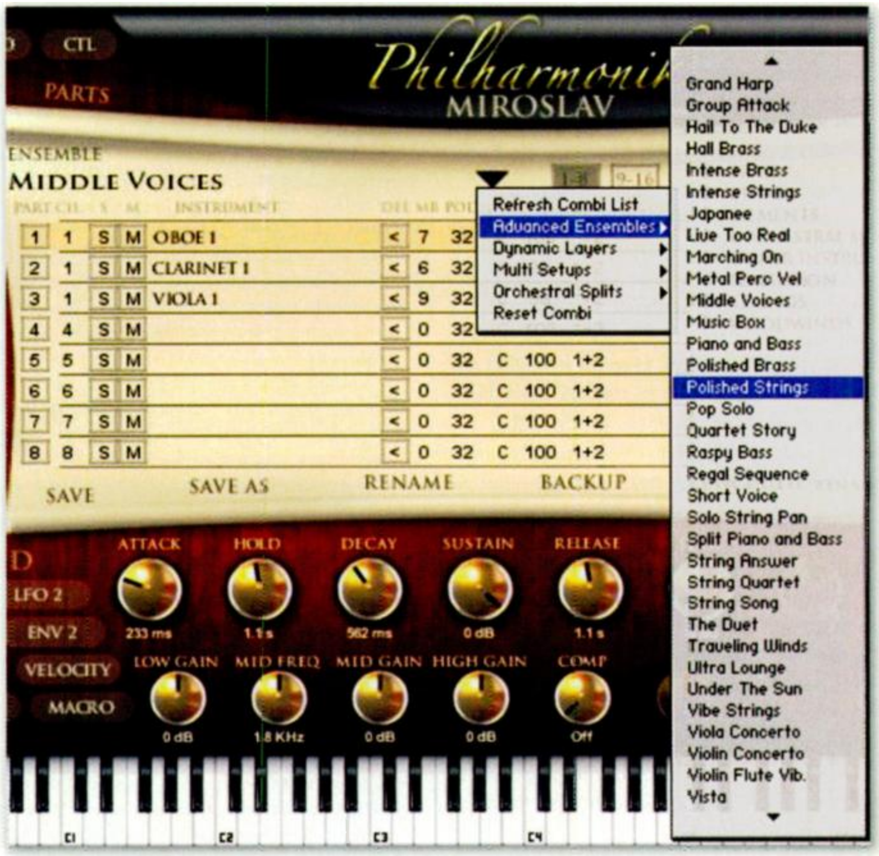
unlooped, benefit from the natural attacks, throatiness, breaths, bow noise and the like. I would tend to go unlooped for smaller ensembles, changing to looped if my RAM or CPU was beginning to complain.

Whichever version you use, constant auditioning is to be encouraged, both to help your familiarisation with the library and to help find the right Instrument for the current context. A neat side-effect of the detailed sampling, and authentic attacks, is that your playing will change to suit the sound being auditioned — it's part of the magic of Miroslav!

Sound Garden

The breadth of this set means highlights are hard to pick out. Even lowlights are relative: I found I didn't take to the choir collection as a group. I can hear that some great recording, editing and collation has gone on, and that attention has been paid to decent SATB vocal ranges, but the result is still a little too artificial, especially for fully exposed work.

The organ, on the other hand, is magnificent. I haven't that much use for one personally but it does sound and play fantastic. It can also add even more weight to an arrangement if the context can handle it. And there is delicacy, in the portative organ-like 'Cathedral Organ Flutes', and subtle depth in the fundamentally bass-heavy 'Bass Pedals'. Amongst the other unexpected instruments, I liked the piano and harpsichord, though the latter is a little too 'baroque' and big-sounding for most of my needs. The classical guitars are quite natural-sounding (some genuine finger noise helps), and the harp set includes a nice range of arpeggios and open tones plus straightforward notes.



A section of the enormous Combination list: many examples are virtual orchestras or whole sections playable from the keyboard.

Within the orchestra proper the main low points, for me, are amongst the solo strings. Violins seemed initially to be unconvincing, but one of IKM's audio demos inspired me to persevere and it's worth the effort. It's not perfect, but the vibrancy that's typical of the whole collection helps it win through in most situations, even fully exposed. You know it's not real, but it's not unpleasantly unreal as the average workstation synth violin might be. Violas have an exaggerated scratchiness that is often attractive but can make it difficult to create a balanced arrangement. I have no complaints at all about the cello collection —

Engine Room

The *Sampletank 2* engine that drives *Philharmonik* means that the new plug-in's 50+ parameter set is strongly biased towards subtractive analogue synthesis. These parameters let you stretch your plug-in a little further than you might generally want in an orchestral context, but you don't have to use them. However, applying these pure synthesis parameters to such a musically involved collection of samples does often result in synthetic sounds that have a similarly organic feel.

In common with other *ST2*-based software, the parameters are divided into several pages of up to eight parameters each. The upper row of knobs in the lower part of the window do the dirty work, with the pages selected by the 'Sound' menu buttons to the left. The menus are LFO 1 and 2, Envelope 1 and 2, Filter, Velocity, Tune and Perform, which is the *Sampletank* standby Macro. The latter are IKM-defined real-time sonic tweaking parameters; there are up to four per patch and they can be, like nearly every other parameter on board, assigned to external MIDI controllers.

The two LFOs are largely identical, with a good range of waveforms and a tempo-sync option; LFO 1 adds delay and free-running options and can be routed to level, pitch or the filter. LFO 2 has the same targets, plus pan. A 'hold' parameter is placed after the attack portion of the otherwise standard ADSR envelopes; Env 1 is routed to level, though, while Env can be routed to either or both filter and pitch. The filter is simple but effective: low-pass, high-pass or band-pass at 6, 12 or 24 dB/octave, with cutoff and resonance controls. Velocity response is well specified: a choice of input curves governs how your playing affects level, filter, pitch, resonance, LFO1 depth and/or envelope 2 sustain.

Like these subtractive synth elements, not all of the pitch-manipulation technology accessed in the Tune menu is relevant to an orchestral sound set. For basic coarse or fine tuning and setting pitch-bend wheel response, the Resampling mode, which just plays back samples as normal, is appropriate. It also offers, rather incongruously, a pan control, for the 'zone' option we'll encounter in a moment.

The Pitch-shift/Time-stretch option is best suited to looped rhythmic material, none of which is included here, but the Stretch resampling option is worth knowing about, as it allows the sample assigned to a keygroup to sound more natural when played above or below its central pitch, and when pitch-bent. Keygroup transitions can still be apparent, but the pitch-shifting artifacts you might normally hear are not so obvious.

There are a couple of other menu options, but these take us away from synthesis and towards the Combination. 'Range' assists in the creation of complex layers and splits, using patches assigned to up to 16 parts to create a monster Combination. A part transpose option lets you put the instruments you want in the key range you want and still have a musically useful pitch range. Whether the combi is multitimbral or a massive layer, you can save the final result.

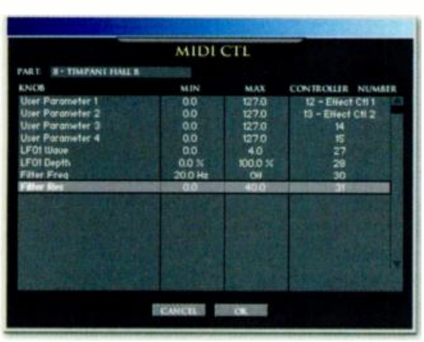
A little extra sonic flexibility is provided by the Zone menu. Firstly, selecting this menu offers graphic feedback (through *Philharmonik*'s mini keyboard) of how samples within an instrument's multisample set are assigned. But selecting a keygroup/zone when it's viewed in this manner allows you to modify synth parameters for just that zone. The original patches are unchanged, with any tweaks made here saved in a Combi.

true ear candy, with full body right down to the open C. Double basses have some of this weight and presence but work better in ensemble presets. In fact, the ensemble strings — for each section and *en masse* — are uniformly musical and rich, even the section violins. The selection of articulations — spiccato, pizzicato, sul ponte, tremolo and so on — is perhaps not as wide as in some more recent and expensive sets, but it will be more than enough to achieve what most of us are after.

Brass and woodwind keep the candy shop open: horns are a slight muddy low point, but there are solo and ensemble examples that blend well in many contexts. The tuba is a big surprise — one seldom hears such attention paid to this instrument in a sample-library context. There are perhaps not quite enough samples for the solo trumpets (and flugelhorn), but the results are still nicely brassy, with useful mellow and bright variations. There are some obvious jumps between keygroups in the lower register of trombone and bass trombone, but one can imagine different slide positions being the cause. Many other instruments have 'special performances', such as glissandi: why not the trombones?

Bassoons are fab — really rich and woody — and the contrabassoon is accurately captured (it doubles nicely with tuba or cello). Sampled flute can be disappointing in my experience, so it was a pleasant surprise to hear patches in *Miroslav Philharmonik* that actually sound like a flautist — and one who enjoys playing. I'll remark again on the presence of alto and bass flutes, and they've been treated just as well as the main C flute. Clarinets have nicely balanced lower, middle and upper ranges — a pleasure to play — and oboes are amongst the best I've heard. The bass clarinet is a little disappointing, but I'm still really glad to have access to it.

Percussion takes a little getting used to, but there are some weighty bass drums and snappy snares that work very well; all the little extras (chimes, castanets and so on) have real clarity and depth, and they play well, too. Although I'd have liked an E-flat clarinet and an orchestrally focused sax or two, I can't complain about the variety of this collection, especially for the money.



In the MIDI Control pop-up, any parameter can be assigned to an external MIDI controller.

Effects In *Philharmonik*

Effects processing is probably more obviously useful than subtractive synthesis components in an orchestral plug-in, but IKM's implementation can initially seem rather odd. There are 20 effects on board, and each part has access to up to four inserts, one of which is always EQ/compression; these insert effects are saved with a patch, but can also be modified as part of a Combi. Four further slots are available as auxiliary send/return effects, accessible by all parts. It's a shame that there are no 'insert' effects on the main stereo mix, though.

IKM try to tell us that the effects available are 'classic', implying, I think, that they suit the classical environment. Hmmm... Five delay and reverb/ambience choices are crowned by the CSReverb, a deceptively simple three-parameter effect that's derive from IKM's forthcoming studio reverb plug-in. On this evidence, that reverb is going to be worth a listen. While nearly all *Philharmonik Instruments* sound vibrant dry, mixes sparkle even more with a little of this reverb. IKM obviously think so, too, since it's provided as a default process whenever you start up the plug-in: you need to switch it off if you want to start properly from scratch.

Modulation effects — chorus, multi-chorus, phaser, flanger, auto-pan, tremolo — make up a big group and then come the creative effects. Lo-fi, distortion, and phonograph seem a little out of place, given that most users will have similar effects elsewhere if they need them. Welcome bread-and-butter processing comes in the shape of tone control, parametric EQ, channel strip, compressor and limiter.

Philharmonik's 20-strong effect list — check out lo-fi and phonograph if you want to add that 78rpm feel to your 21st Century productions!

Pedigree Sounds

Philharmonik has been constantly linked to my sequencing software since the start of this review — I like mixing up 'real' orchestral sounds with other bits I'm working on, and find it the perfect 'audio proof' when scoring or composing. Rather than the often stilted result one would hear with even a good General MIDI sound source or workstation synth, *MP* becomes a part of the process. Rather than inputting notes, you're actually playing the sounds.

The flip side of this situation is that existing MIDI data created with a different sound source in mind may not sound right, at least initially. You may have to spend a little time searching through *Philharmonik*'s presets to find voices that work, and then still be prepared to fine-tune the MIDI data or the preset to produce the desired performance.

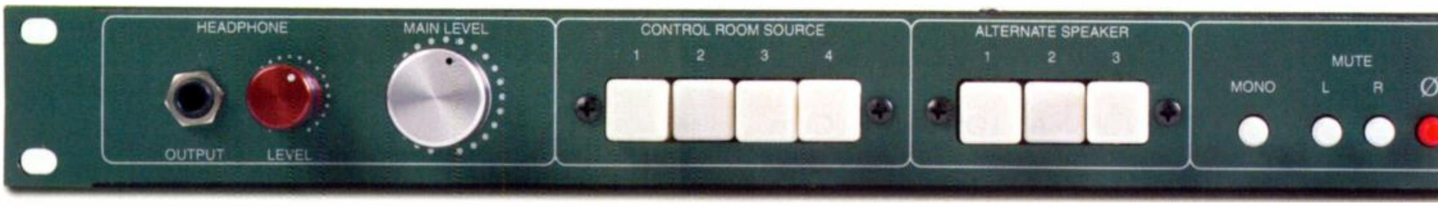
Gripes and problems are negligible. First of all, it stands to reason that the more RAM and CPU power you can throw at *MP*, the better. On a day-to-day level, I found that two Instruments would sometimes be required to perform one part, since no single Instrument had the right mix of playing techniques. *Sampletank* technology doesn't really take advantage of program changes or other easy ways to swap presets, but there are 'elements' patches that often have a little more flexibility. The other option is to use more than one part to create an effect.

Occasionally I felt there weren't quite enough samples to a multisample, and I found one odd instance of the pitch-bend control responding back to front, on just a few higher notes of a 'Stretched' cello preset. On the other hand, Stretch helps a number of patches make the most of the keygroups supplied.

Philharmonik isn't alone in the market — not even the sub-£500 market — but its pedigree gives it an edge, even though some raw material is over a decade old. The samples at the heart of *Miroslav Philharmonik* represented something of a high-tide mark in sample-library history. They broke new ground in terms of detail, recording quality and expressiveness. With the release of this plug-in, the set becomes the serious orchestral library for the rest of us. That we have it all, and more, in a ready-to-go plug in for under 400 quid, a fraction of the library's original cost, is amazing. SOS

information

- £ £399.
- T IK Multimedia UK +44 (0)1223 234414.
- F +44 (0)7005 96800.
- W www.ikmultimedia.com
- W www.philharmonik.com



Coleman Audio M3PH MkII Monitor Controller

Hugh Robjohns

This new unit from Coleman Audio makes a mark with its sonic integrity, despite its apparent lack of operational 'frills'.

Dedicated monitor controllers were rare back when everyone had a console of some sort in their studio. However, now that most of us work entirely with DAWs we have little need for a full console anymore. A few outboard preamps for tracking and a separate monitoring controller are more convenient, more cost effective, and take up a lot less space. As a result, the number of manufacturers offering monitor controllers has grown almost exponentially in recent years, with products spanning all budgets and requirements.

The subject of this review is a no-frills design from America, Coleman Audio's M3PH MkII. The emphasis is very much on quality and simplicity, so there is no provision for talkback or headphone cue mixes — just a simple four-source input selector, three-output speaker router, basic signal-manipulation functions, and an engineer's headphone output. The review model featured an entirely passive signal path, although an active and fully buffered model (the M3A) is also available. The term 'passive' is possibly confusing here — the M3PH does contain active electronics, but only for the headphone output and to derive a summed mono signal. All signal routing and conditioning is performed directly on the balanced signal using the front-panel switches, and the rotary volume control is a quad-ganged stepped attenuator with a claimed tracking accuracy of 0.05dB.

A Peek Inside

The unit occupies 1U of rack space, extending about 260mm behind the rack ears. Internal construction is to a high standard, with good-quality components throughout. There are five separate fibreglass circuit boards, starting with a small switched-mode power supply on the right-hand side of the chassis. A second board runs across the back to carry the XLR connectors — 14 of them (four input pairs, three output pairs) — and a third board occupies most of the central area to carry the selector switches and summing amplifier electronics. The remaining boards are for the headphone amplifier and a vertical card to connect the four sections of the switched volume control. Ribbon cables and header sockets are used to connect the boards.

A 'hidden' switch inside the casing on the main circuit board allows the third speaker output to be reassigned as a fixed line-level output (post the input selector), enabling its use as a feed to an external metering system or as a pass-through for the monitored signal. A switch on the power-supply board configures the unit for 230V or 115V mains supplies. There are no protective fuses in the mains or DC supplies, so you are reliant on the mains supply trip (or plug fuse in the case of UK models) should anything go wrong internally.

Since the XLR connectors occupy the entire back panel, there is no space for the mains cable connection. Instead, a captive mains cable exits the unit at the rear of the right-hand side panel. In most cases this unusual construction won't cause any problems, but in tight rack cases or frames this might make installing or removing the unit harder than usual.

In Use

The front panel is clear and intuitive, dominated by seven large white buttons:

SOUND ON SOUND

Coleman Audio M3PH MkII
£752

pros
- Passive signal path ensures sonic integrity.
- Very accurate switched-attenuator volume control.
- Speaker mutes allow mono on a single speaker or on both.
- Third speaker output switchable to become a fixed line-level feed.

cons
- No Dim facility.
- Phase reverse after mono switch limits usefulness.

summary
A passive, fully balanced monitor controller with a useful, if slim-line, range of facilities.

Photos: Mark Ewing

four interlocking switches to select the input source and three more interlocking switches to select the speaker output. The button caps can be removed and engraved if required, but there is also plenty of panel space for stick-on labels. To the left of the buttons is the rotary stepped attenuator, which has a light detented action, and to the left of that is a conventional volume knob for the adjacent headphone output socket. Over on the right-hand side are four smaller buttons which select the derived mono signal, mute the left or right speakers, and reverse the polarity of the right-hand output channel — this last button being coloured red to differentiate it. Finally, the mains power switch is at the extreme right-hand side.

As already mentioned, the signal path is entirely passive, and I was unable to hear any obvious degradation to the monitor signal when working with balanced sources and destinations. Operation is intuitive: select the desired input source and monitoring speakers using the buttons. The mono, phase-reverse, and mute switches allow quality and compatibility checks to be made as necessary. The two mute buttons conveniently allow mono to be auditioned from both speakers (as a phantom image) or just from one — an important function that few monitor controllers permit, strangely. Mono on both speakers allows the true phantom centre image definition and position to be checked, and mono on one speaker gives a better idea of the spectral balance of a derived mono signal than a phantom image.

The phase-reverse button is located after the mono selector in the signal path, which makes it impossible to hear the stereo difference (Sides) signal. This is a great shame, because being able to audition the difference signal is extremely useful, not least for the 'null' technique. (If you invert the polarity of one channel then sum to mono, equal-level signals cancel each other out, and this is a handy trick for matching channel gains or aligning stereo mic arrays.)

The only function missing completely is a Dim facility to allow the speakers to be turned down — perhaps to answer the phone — without upsetting the reference monitoring level. However, since the monitor volume is a switch, it inherently provides discrete level steps and is therefore fairly easy to reset accurately anyway.

This is a neat and well-built unit, with all the necessary features and facilities a typical home-studio/DAW installation would need. Its fully balanced and passive signal path ensures sonic integrity, and the ability to mute the two speakers separately is very handy for checking mono balances. Few home studios would probably run to three sets of monitors, so allowing the third output to be switched (internally) to provide a fixed output to feed meters or an external recorder is a welcome facility. SOS

information

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- T KMR Audio +44 (0)20 8445 2446.
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Image Line Sytrus

FM Soft Synth For Windows

The modulation matrix at the right of the *Sytrus* window is set and forget, while the modulation X-Y pad at the left is where the real-time action takes place. There is also a scribble scrip to put in information about your patch.

Combining frequency and ring modulation with analogue-style resonant filtering, Image Line's *Sytrus* takes FM synthesis in new directions.

Alan Tubbs

Yamaha's DX7 put to pasture the analogue synthesizer of the mid-'80s. With bang for buck in the voices department and an all-new 'digital' sound, frequency modulation synthesis dominated the field for the rest of that decade; and like many other '80s fads, FM has made a bit of a comeback lately. Native Instruments' *FM7* is a virtual update of the DX7, and there are numerous freeware and shareware FM soft synths out there. Bridging the gap between the expensive and the feature-limited is Image Line's *Sytrus*, a six-operator frequency and ring-modulation instrument in *FL Studio* native, VST and Direct X formats. Version 2 has just been released, adding new filters, arpeggiating envelopes and more presets to an already fulsome collection. *FL Studio* users will have to wait for *FL Studio 6* to get the native version, although they can, of course, use the VST and Direct X versions now.

Test Spec

- *Sytrus* v1.5 and v2.0.
- PC with 1.7GHz AMD Athlon CPU running Windows XP Pro SP1.
- Tested with Image Line *FL Studio 5*, Cakewalk *Project 5* v2.

Oscillating & Modulating

Image Line also make the popular *FL Studio* host application, formerly known as *Fruity Loops*, so they aren't some T-shirt operation when it comes to synth development. One of their synths for *FL Studio* was *DX10*, a simple FM synth, but *Sytrus* is in a different league altogether. Each voice not only enjoys six operators, but three filters, a chorus unit and three delay lines for effects, flexible multi-stage envelopes and extensive modulation capabilities, before one even gets to modulating any frequencies! There is a *très* cool assignable X-Y modulation controller, and an at-a-glance frequency/ring modulation matrix.

The modulation matrix is at the heart of *Sytrus*'s FM capabilities, and allows control of every operator's audio output and the amount of FM applied to each operator. If you've ever upped an LFO into the audio frequency range and then used it to modulate the frequency of another oscillator, the clangorous results are a form of basic FM synthesis: many patches on the venerable Minimoog used the third oscillator this way to thicken and distort the overall timbre. FM synthesis as implemented in the DX7 and in *Sytrus* is far more complex, as each of the six available oscillators ('operators' in FM-speak) can modulate its own frequency or that of any other. *Sytrus*'s modulation matrix is a neat arrangement for seeing at a glance the overall algorithm (as the feedback and output structure that links the operators is called in FM synthesis), as well as the relative amount of modulation. Each 'knob' can be muted by right-clicking, which makes tracking down a wayward operator's modulation effect instantaneous. However, I wish the matrix itself could be expanded while setting levels without expanding the whole synth panel.

The other big difference between digital FM synthesis and basic Minimoog-style analogue frequency modulation is the amount of control over the operator itself. In the DX7, each sine-wave oscillator has its own envelope generator. In *Sytrus*, every operator has its own envelope and LFO; in fact, every operator is its own single-voice synth in itself, with control over pan, volume, modulation input, pitch, phase and pluck damping. There is a tab for controlling the oscillator Shape, too, of which more later. All but the Osc Shape tab have their own envelope and LFO, as well as various other controller mappings. Each of these can affect the sound output, just as in any other soft synth. The power of FM synthesis, however, comes from modulating an audio operator (the carrier) with one or more other operators (called modulators). A single carrier modulating itself makes the output 'buzzier', and the fun starts when you alter the pitch ratio of the modulator: octave ratios produce harmonic changes to the timbre, while whole steps and cents produce inharmonic distortion that can get downright ugly. Judicious use of the amount of modulation and the modulator's envelope brings this distortion in and out. Sharp modulation with a quick decay is the basis for the infamous DX7 Rhodes piano sound heard on half the pop songs of the '80s, as well as the transient attacks of many other 'realistic' sounds. *Sytrus* also adds ring modulation to the fun, which generates new frequencies related to the sum and difference of the carrier and modulator frequencies, and produces a totally different tone from FM. While the implementation here does not sound as aggressive as many analogue ring modulators, it certainly provides another way to muck up your waveforms.

SOUND ON SOUND

Image Line *Sytrus* $179

pros

- FM synthesis with resonant filters.
- Assignable X-Y modulation pad.
- Makes FM synthesis about as easy as it can be without cheapening the feature set.

cons

- FM synthesis is one of the more complex methods of sound production and *Sytrus* only makes it easier, not easy.
- *Sytrus* presets from the *FL Studio* site requires conversion from the '.FST' format before they can be loaded in other VST hosts.
- The reference manual is a bit thin in some areas, especially for beginners.

summary

Sytrus is an impressive FM soft synth with a resonant filter twist.

Mention of waveforms brings us to the Oscillator Shape mentioned above. The DX7 dealt strictly with sine waves, while later Yamaha FM synths had the standard palate of wave shapes available. With *Sytrus*, you roll your own. Underneath the top row of tabs for choosing the Main and Operator pages are controllers for Operator Shape. Here you choose harmonics, then alter the wave shape via four faders named Shape, Tension, Skew and Sine Shaper/Pre-filter. While this is not as simple as picking from a given set of shapes, all the standard waveforms are there along with everything in between. All the changes show up in the oscilloscope display, while noise can be added via a fader before the output reaches the operator master controllers and the all-important Octave and Frequency tuning controls. Version 1.5 introduced a tab page for drawing in harmonics (along with a larger view of the waveform). *Sytrus 2* allows you to lock the harmonics once you have the shape pinned down, and then export the resulting waveforms as audio.

No Unnecessary Complication

As you start to count up the number of controllers for each *Sytrus* operator, you get six times eight. That is 48, multiplied by six operators. Add in a 'plucked' mode in the ▶

Oscillator page, which disables the FM but uses damping to simulate the roll-off of plucked strings, and the variable wave shapes, and the permutations might start to hurt your head. If FM synthesis seems like a tweaker's heaven and rather intimidating, it doesn't have to be. Just because all the functions are available doesn't mean you have to use all of them all of the time. Many interesting sounds — including many of the presets available for *Sytrus* 1.5 — use only one or two oscillators. Alternatively, you don't need to use the frequency-modulation capabilities at all: you can use *Sytrus* as a conventional six-oscillator subtractive synth. With unlimited numbers of breakpoints in the envelope generators for each operator and filter, long, evolving patches are no harder to design than with any other synth. Image Line have also added a new feature for envelopes that I haven't seen before: arpeggiation. Arpeggiation points and directions can be added within the envelope slope, just like breakpoints. When a chord is held down, all the notes sound, but the envelopes alternate, adding a sequenced feel to oscillators, filters or anything that can be controlled by an envelope.

Each of the *Sytrus* filters can be set to a gentle 12dB, a Moogish 24dB or an ultra-steep 36dB/octave mode, then passed on to the next filter. Version 2 adds another four filters to this list, in case the previous eight couldn't fill up your sonic plate. Low-pass, band-pass and high-pass filters are available all at once, as well as attendant envelopes, LFOs and other controllers for each type of filter. Another nice touch is control over the amount of filter resonance, not just cutoff. The sound of *Sytrus* leans toward the bright and digital, so it probably wouldn't be my first choice as a virtual analogue synth, although it can still do a respectable job. And with all the control available, it is more like a virtual modular rather than a hard-wired emulation.

With six oscillators per voice, *Sytrus* can also do simple additive synthesis, and many of the organ patches use this technique. The Hammond and pipe organ patches will please all but the purists, but my favourite organ patches (and one of the reasons I originally got *Sytrus*) are the Farfisa sounds. They cut cleanly through the thickest mix, give a guitarist keyboard envy and make everyone else in the band want to cover '96 Tears'. Other voicings that FM and *Sytrus* do particularly well are bells and other such inharmonic instruments. The strings are good alternative if you are looking for something other than realistic samples or thick analogue string machines. Finally, spiky Clavinets, bright basses and a host of other sounds that will take you back to the mid-'80s are meat and drink to *Sytrus*.

The settings for each of the six operators. The Osc tab allows you to draw in harmonics and provides a big oscilloscope view of the waveform. Both the envelope and LFO controller tabs also let you draw slopes and breakpoints, and the results can be saved.

Juicy Fruit

I encountered one or two problems with *Sytrus*, but nothing very serious. When I received the update to version 1.5, it defaulted to loading the DLL file into a different folder from the one where I had placed the original, causing a mixed-up image of the synth to be displayed. Deleting the old DLL and copying the new one solved the problem. The on-line manual is still rather brief; everything is there, but not in the most user-friendly form for an FM novice. The envelopes and LFOs are not calibrated by time or any other standard. In practice, this is not a bad thing, as it forces you to use your ears, not eyes, but can be disconcerting at first. Otherwise, *Sytrus* suffers only from the complexity of FM synthesis: unlike S&S or VA, changing a sound to make it more usable for the task at hand often involves more than just twisting a knob or two.

However, *Sytrus 2* ships with a couple of hundred preset voices, and is easy to use as a preset synth rather than a programmable one, if that is your bag, with the X-Y modulation controller offering plenty of scope for real-time manipulation. A quick Google search will get you the original DX7 patches and access to others, as well as more info on designing FM patches than most of us will ever use, while an on-line User Group at http://forum.e-officedirect.com/forum.exe?ForumName=FLStudio_SytrusPresets is the place to go to exchange or just download patches. Be aware, though, that most of these are saved in the *FL Studio* format, which won't load into other VST hosts; the only workaround is to download *FL Studio* (even the demo will work), then copy and paste the presets from an instance of *Sytrus* running in *FL Studio* to another running in your chosen host. *Sytrus 2* includes some other small housekeeping updates, including subdued, translucent pop-up options/presets replacing the mundane Windows look in earlier versions. It won't do much for your sound, but is nice for the price — the update is free.

FM synthesis will probably never again be as universal as it was back in the '80s — there are too many easier ways to get sounds now. However, *Sytrus* can add a whole other spectrum to your sound palate, even if you aren't a tweaker. There is an organic nature to the sound as the operators mutate one another in a surprising, and often unexpected manner. The addition of ring modulation and resonant filters add even more colour to the sound with little programming overhead. If you are not sure whether FM synthesis is right for you, you can download the demo version. It might not be worth the time it takes to learn FM programming, but it could be just the ticket for sonic explorations. SOS

information

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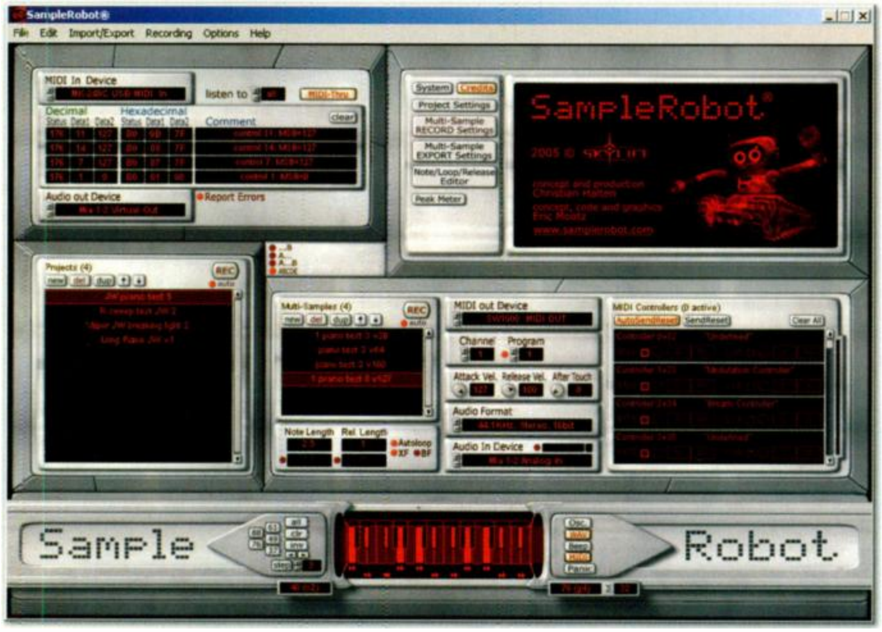
Skylife Sample Robot

Like the idea of having all your hardware synth sounds in a software sampler, but can't be bothered to do the necessary programming? Fear not, Skylife's *Sample Robot* will do it for you...

John Walden

With the advent of software samplers, more musicians have now got access to sophisticated sampling technology via their Mac or PC, and while many are happy to use commercial sample libraries, there are all sorts of reasons why creating a sample program from scratch might be the way to go. Creating your own, unique sampled sound can be satisfying and can give your music a truly original element; and it may be that no commercial library includes the sound you're after. In addition, many musicians moving to software-based setups still have patches on older hardware synths they'd like to take with them.

However, creating a convincing sample program from an instrument can require a lot of detailed work, and that's where applications like Skylife's *Sample Robot* come in. *Sample Robot* is designed to capture sounds automatically and convert them into multisampled instruments, including the creation of velocity layers — and if the source of the sound that you wish to capture is a MIDI synthesizer, it can carry out the whole process virtually without human intervention.



Auto-sampling Software For Windows

Robotic Basics

A *Sample Robot* Project can usually be thought of as a place to capture a single sound program from the target synth. Users can specify the range and spacing of MIDI notes to be sampled for each of as many sample layers as required. Within each layer, different settings for velocity and MIDI controllers can be specified, as can the note and release lengths. Settings can also be specified to automatically create loop points within the samples. With all the settings made, the recording process can be triggered and the entire process then happens without further user intervention.

For some, this will be more than enough to make *Sample Robot* interesting. However, it also has a couple of other tricks up its sleeve. First, while you forgo some of the automatic elements of the process, the *Sample Robot* interface provides a well organised environment for multisampling real instruments, which is great for those who really do like a genuine DIY approach to sampling. Second, the 'Import Single Sound Library' function streamlines the process of extracting samples from audio-format sample CDs. Where a single audio track contains multiple samples, *Sample Robot* will separate out each sample and map them to a range of MIDI keys.

Of course, all these captured samples are only of any use if they can then be exported in a format suitable for your favourite software sampler. In the current release version (v.1.32), sampling is restricted to a maximum of 16-bit resolution, and export in raw WAV, *Halion* and Soundfont 2 formats is provided. I did most of my testing with Steinberg's *Halion 3* and, on the whole, the process is very straightforward. Christian Halten, *Sample Robot*'s designer, informed me that other export options are in development, with *Reason NN-XT* support to follow shortly and discussions ongoing with a number of the other major software sampler manufacturers. That said, many people will probably be happy with the benefits *Sample Robot* brings in capturing the samples, even if they then have to use

SOUND ON SOUND

Skylife *Sample Robot* £169

pros
- Generally very easy to use.
- Can save a huge amount of time when sampling from scratch.
- Capable of excellent results.

cons
- At present, a limited range of export options.
- Probably a little expensive to appeal to the hobbyist.

summary
If you are a dedicated DIY sample creator, then *Sample Robot* might just change your life — it's a real 'Swiss Army knife' for creating sampled versions of hardware synths.

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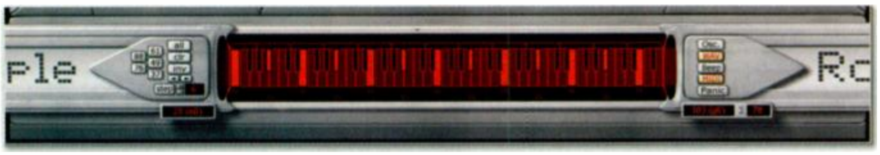
The Production Line

Installation of *Sample Robot* proved simple enough. The software uses a challenge-and-response form of copy protection which worked without a hitch on my test system. It operates as a stand-alone application and, as shown in the various screenshots, the user interface is split into five main areas. Top left is the MIDI monitor window. Here the user can specify the MIDI In and Audio Out devices to use, while the bulk of the display provides a means of checking MIDI activity. Beneath this, the currently open Projects are listed in the Project window, where they can be created, duplicated or deleted. Once a Project has been configured, the Rec button at the top right of this window will trigger the automatic recording process.

Projects can consist of either a single sample layer or multiple layers. The layers themselves are created and configured using the Multisample window in conjunction with the MIDI and Audio Settings window and the MIDI Controller window, which are located in a strip to the right of the Project window. The Multisample window lists the layers that exist within the currently selected Project, and allows you to create, edit and delete them. The rest of the settings within this section of the interface all specify the properties for the selected layer. These include Note Length, Release Length and the Attack, Release and Aftertouch values. Usefully, the MIDI Out device and channel and program number can also be specified for each layer, allowing you to (for instance) create a velocity crossfade between different synth sounds by specifying different program numbers for particular velocity layers.

The MIDI Controller window allows settings for any available MIDI controller to be set for a particular layer, which might be useful if the character of the sound you are trying to capture depends heavily upon particular properties of the original synth. Once captured, *Sample Robot* layers could then be used to crossfade between different controller settings within your software sampler. The bottom of the display contains the virtual keyboard window, which can be used to specify the range and pitch of notes to be sampled for each layer. Again this offers considerable flexibility, allowing the sampling to be as sparse or detailed as required.



***Sample Robot*'s Keyboard window is used to specify the keyboard range and note intervals to be sampled.**

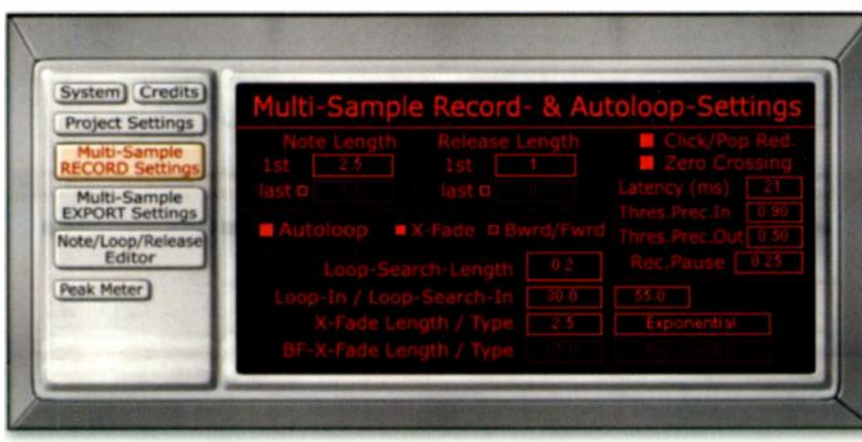
Modern disk-streaming technology sometimes means that long samples can be used to avoid the need for creating loop points. If loops are required, though, the editing involved can be very time-consuming. Thankfully, *Sample Robot* includes an Autoloop function which attempts to automatically create loop regions for each sample within a program. Loops can be set to either crossfade (the default setting) or to loop backwards and forwards. Detailed control of the loop-creation process is one of the functions provided within the Multi-purpose window which fills the top right of the *Sample Robot* display. As shown in the screenshot, the Multisample Record Settings page provides a range of options to change the way loops are created.

The Multi-purpose window also provides access to the screens for the Note/Loop/Release Editor, Multisample Export Settings, Project Settings and a Peak Meter. The Note/Loop/Release Editor allows the user to fine-tune the settings created by the Autoloop function for an individual sample. While the display is quite small, it is possible to zoom in on the region of interest, and all the parameters associated with the attack, loop and release sections of the sample can be adjusted. Usefully, users can reposition the in/out markers for the Autoloop search function and then re-run the search to find the best loops points.

The Multisample Export Settings provides a way of specifying how the sample layers of a project will respond once exported. The most obvious task here is to define the range of velocities over which a particular velocity layer will respond and, again, *Sample Robot* can make these settings automatically for you, including specifying an overlap between different velocity layers if required.

Robot Chores

Having browsed through the PDF manual (which is well worth doing), I tested *Sample Robot* with a number of different tasks, starting with the most obvious: sampling a sound program from a hardware MIDI synth. Creating a new Project and a series of multisample layers within *Sample Robot* is really a breeze, with the only real decisions to make being the number of velocity layers, the key range/interval to sample and specifying appropriate Note Length and Release Length settings. Having made the appropriate MIDI and audio connections, clicking the Project window Rec button brings up a small 'recording' dialogue, and the sampling starts once the Start Recording button is pressed. It is then simply a question of sitting back and letting *Sample Robot* do its thing for a couple of minutes.



The Autoloop settings take some getting used to but the results can certainly be worth the effort.

In capturing a variety of different sounds in this fashion, including piano, pad, bass and lead sounds, aside from an very occasional individual sample that didn't seem to get recorded (but which could easily be captured by repeating the recording process), my only initial difficulty was getting smooth results with the Autoloop settings. These require some experimentation in order to find the most suitable settings for a particular type of sound, and while the manual describes the

Test Spec

- *Sample Robot* v1.32.
- 3.2GHz Pentium 4 PC with 2GB RAM, Echo Mia 24, Egosys Wami Rack 24 and Yamaha SW1000XG soundcards, running Windows XP Pro SP2.
- Tested with Steinberg *Cubase SX* 3.1.0 and *Halion* 3.0.1.

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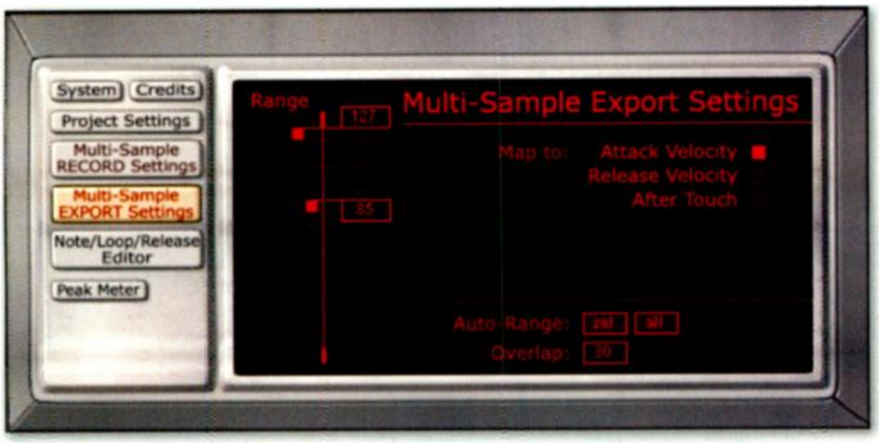
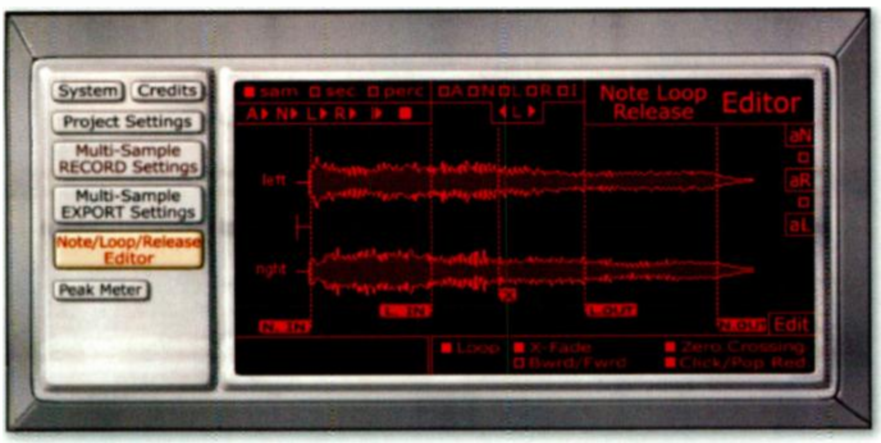


If the project includes velocity layers, their response can be configured via the Multisample Export settings.



If you need to get down and dirty with your loop points, *Sample Robot* provides a small but perfectly functional editor for the task.

controls in some detail, I felt that a few tutorial-type examples might get new users off to a quicker start. However, a little perseverance brought considerable improvements, and as the whole automated sampling process is so efficient, it is no great loss if a little trial and error is required in order to get a smooth loop for a particular sound — it certainly beats creating loop points manually!

Exporting Projects to *Halion* also proved very straightforward. Selecting 'Export Selected Project To Halion' from the Import/Export menu renders the samples as WAV files to disk and places the mapping information within the Windows clipboard. It is then just a question of pasting the mapping information into a suitable empty program slot in *Halion*. For *Sample Robot* Projects based upon multisamples and velocity layers, this worked very well. Things are slightly less straightforward for programs that contain, for example, a mod wheel 'layer' from the source instrument. These layers require a little work within *Halion* in order to set up a mod wheel-controlled crossfade, but this procedure is clearly explained in the *Halion 3* manual.

With an appropriate MIDI loopback connector and audio routing, I was also able to use *Sample Robot* to capture sounds from various software synths. While it might seem pointless to sample a sound that already exists in a software form on the host PC, this could be useful if you wished to bring a selection of sounds into a single sampler front end, particularly with a laptop setup that might be used in a live context. Anyway, it works well enough for those who might need it.

Sampling an acoustic instrument is, of course, a somewhat more labour-intensive task, but *Sample Robot* does make this task a whole lot easier than it would be using a conventional audio recording and editing application. Finally, I was also impressed by the 'Import Single Sound Library' function. Though it's a fairly simple offering, simply breaking a single audio file containing several samples into individual samples each mapped to a specific key, even this helps to reduce the 'chore' of dealing with audio-format sample CDs. This said, it might be nice if this feature could be expanded in scope in future releases. For example, the ability to split a series of drum samples from an audio file and stack them on a single key with velocity response would be useful. In addition, the ability to automatically trim drum loops from an audio sample collection to exact bar lengths would be a big time-saver.

In using the current release version of *Sample Robot* (v.1.32) for the review, there were a couple of other minor niggles. First, the screen layout is quite busy and I was occasionally having to squint to check what I was doing. This is particularly true within the Note/Loop/Release Editor window, even with the ability to zoom in on the waveform display. Second, there was no support for ASIO drivers, although in most cases this would not be of any significance as real-time response is not required when automatically recording samples from a MIDI module. However, just as I was finishing the review, Christian Halten provided me with a beta of version 1.5. Amongst a number of tweaks and additions, this includes options for resizing the display and the beginnings of support for ASIO and 24-bit audio, and although both of these are not yet fully implemented, development is clearly moving on at some pace.

Conclusions

Sample Robot is a very clever application and, while it is probably too expensive and too specialised to appeal to the merely curious, it could save a huge amount of time and effort for dedicated sample-heads. My only reservation would be the somewhat limited range of Export options, but even if you just used it as a means of capturing your multisamples in an organised fashion before building your sample programs from the raw WAV files, *Sample Robot* would be a big time-saver.

Sample Robot is clearly useful for those wishing to move some classic hardware sounds into the software realm, whether for studio use or to streamline a live setup. If Skylife are successful in their plans to add export support for more of the major software sampler formats, then *Sample Robot* ought to build a dedicated following. At the time of writing, *Sample Robot* is only available on-line, but a boxed version is in the pipeline, and will soon be available worldwide via ESI Pro's network of distributors. Anyone interested should download the demo version from the web site without delay! SOS

System Requirements

- **Pentium III 600MHz or faster, 256MB RAM, 30MB free hard drive space, Windows XP, MME-compliant soundcard, MIDI interface.**

information

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Line 6 Echo Park and Liqua-Flange

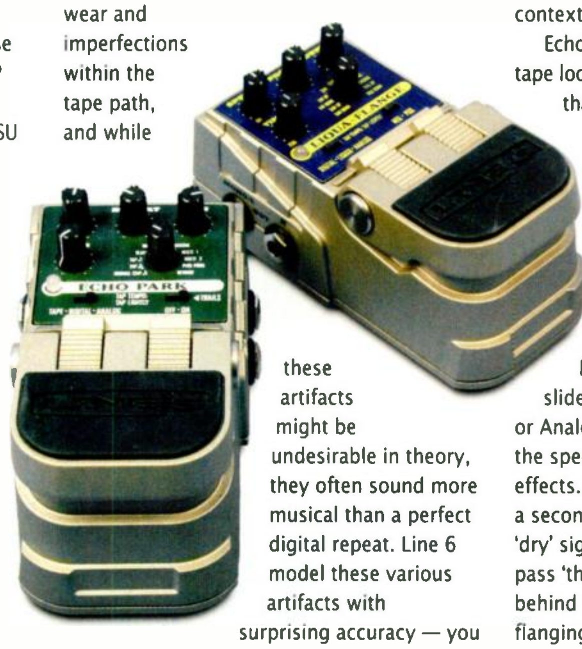
Stomp box effects

SUMMARY: Although these are primarily guitar pedals their stereo I/O capability also makes them applicable to studio use. They represent the best 'vintage' emulations I've heard to date and their build quality is exceptional, but you will need to buy a PSU as the battery drain is pretty high.
www.line6.com

The Line 6 range of Tone Core pedals are based on digital modelling, powered by identical DSP engines within the base of the pedal, with a modular, removable control/algorithm section. The two units each offer authentic recreations of vintage effects as well as cleaner-sounding digital versions. The pedals themselves are made of heavy cast metal, and their batteries can be changed without the use of tools by squeezing release buttons on the side of the pedal hinges. DSP engines use a lot of power, so it is more practical to run the units from an external PSU (9V 70mA): Line 6 offer an optional PSU for this purpose.

Echo Park offers a maximum delay time of 2.5 seconds and has a tap-tempo function activated by pressing the pedal gently; there's also a manual delay time control, as well as knobs for mix, feedback and modulation depth. By including modulation, you can get some nice chorus-like echoes out of this unit. All the Tone Core pedals have dual input and output jacks to cater for the processing of stereo signals so they are perfectly usable in the studio, even though the effect itself is generated from a mono sum of the two inputs. An 11-way rotary control selects from a number of slapback, single-tap and multi-tap echo treatments as well ping-pong delay and the famous Line 6 reverse delay. There's also a ducking delay option, which suppresses the delay level when an input is present. Sweep adds a swept filter effect to the delays, and the Trails switch lets the user decide whether bypass kills the echoes or allows then to fade away naturally. Overall character can be set to Tape, Digital or Analog.

A tape-loop delay adds more distortion and noise every time a sound is fed round to create multiple repeats, and the signal loses some high-end. There's also some pitch modulation due to tape wear and imperfections within the tape path, and while these artifacts might be undesirable in theory, they often sound more musical than a perfect digital repeat. Line 6 model these various artifacts with surprising accuracy — you get everything except the tape breaking in the middle of your guitar solo!



In the Analog setting, emulating the solid-state, charge-coupled analogue delay pedals that became popular during the 1970s, the sound again becomes 'blurry' but in a quite different way to tape-loop delay, and there are subtle clock noise modulations that have also been emulated here. Although the analogue delay has a rather dull character, it can work very well in a musical context.

Echo Park comes so close to capturing the tape loop and analogue delay echo sounds that I think few people would be able to tell the difference, especially in the context of a mix.

Liqua-Flange also offers a choice between modern-sounding digital flanging and emulations of classic analogue flangers, the definitive model probably being the original Electro Harmonix Electric Mistress. A three position slider sets the character to Digital, Liquid or Analogue, with rotary controls adjusting the speed, feedback, time and depth of the effects. The Liquid switch position brings in a second delay line which slightly delays the 'dry' signal, enabling the flange sweep to pass 'through zero' (in other words, from behind to in front), just as it did in tape flanging when one recorder overtook the other. There's also a smoothed-random step modulation option to avoid the audible cycle of an LFO. Liqua-Flange is the nearest thing I've heard to the Electric Mistress, and though it still doesn't capture the elusive magic of tape flanging, it gets a lot closer than any other pedal I've tried. *Paul White*

COOL STUFF

The **Digitech EX7 Expression Factory** is a new pedal offering a range of wah-wah, whammy and other popular effects. The EX7 is identical in dimensions and control layout to Digitech's Artist Series Jimi Hendrix Experience pedal, reviewed in *SOS* December 2005 (www.soundonsound.com/sos/dec05/articles/digitechartist.htm) — one knob selects the type of effect and a further three dual-concentric pots adjust effect parameters, and the pedal has separate amp and mixer outputs with on-board cabinet modelling. There are seven different effects, or 'models', to choose from. The first two are wah-wah effects, one based on the Dunlop Crybaby and the other on Vox's Clyde McCoy Wah. Models three and four are based on two of Digitech's own effects, namely the XP300 Space Station and the original Whammy pedal. Model five is based on the Unicord Univibe vibrato and chorus effect, model six recreates the sound of a Leslie rotary speaker and, finally, model seven is based on the A/DA Flanger effect. Seven distortion effects culled from Digitech's DF7 accompany the models. The EX7 is out now and costs £229.
Sound Technology +44 (0)1462 480000.
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Danish manufacturers **Carl Martin** have brought out three new affordable effects pedals based on technology found in some of their existing high-quality effects. The Vintage Series pedals certainly look suitably retro, with their colourful die-cast metal casings and cream chicken-head knobs. **Crush Zone** (£49.99) is a high-gain distortion pedal with Level, Tone and Distortion controls. **Surf Trem** is a simple '50s-style tremolo effect with Depth and Speed controls, employing a circuit taken from Carl Martin's analogue Tremovibe tremolo/vibrato. **Red Repeat**, which is based on the Delayla pedal, provides up to 600ms of delay and an echo circuit capable of analogue-style self-oscillation. All three are available now.
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TECHNIQUE Re-recording with Aura and Mama Bear

Processors such as Fishman's Aura (www.fishmanaura.com) and D-Tar's Mama Bear (www.d-tar.com) can work wonders with brittle-sounding, DI'd acoustic guitar pickup signals in recording situations where you can't use a mic. However, it can be difficult to commit to a particular balance between the direct and processed signals at the time of recording, especially if you are playing at the same time as other musicians, as you may often need the more immediate and focused sound of the pickup to dominate over the more diffuse processed sound.

However, provided that you also record an unprocessed DI'd signal alongside your processed signal, you can always route it back through the processor and re-record the results, with the luxury of then being able to set the pickup/image balance in the context of the whole mix. Both the Aura and the Mama Bear are optimised for a pre-amped instrument-level input, so there's no problem sending them a line-level signal and recording the processed output back to another track.

There are two strategies available here: one is to monitor the processor's output in the context of the mix in order to set the balance between dry and processed audio. The other is to record a fully processed track with no dry component and maintain the flexibility to tweak the balance right up until the final mix. The latter would seem to have obvious advantages, but there is one more factor you need to bear in mind, namely, the fact that any kind of digital processing incurs a delay.

The Aura delays its dry signal to keep it time-aligned with the processed signal, so even if your re-recorded track is a blend of the two, there will be no unwanted cancellations or comb-filtering effects. If you want to keep any of your original dry track in there to maintain your mixing options right to the end, you will have to establish the size of the delay and pull the processed signal back in time with it (the Aura alone adds about 3ms, and the converters in your audio interface will also contribute). The Mama Bear is different in that it doesn't delay the dry signal, but its processed signal is quite different too, and whilst there is a distinct level drop when dry and processed signals are combined equally, there is no really nasty comb filtering.

With the Aura, I am often happy to use a 100 percent Aura mix in a recording, whilst, to my ears, Mama Bear always requires a majority of original pickup, with the processed signal just providing a softening 'halo' around it. The Mama Bear signal at 100 percent is way too diffuse and woolly to be useful, in my opinion, but that may be partly what helps it work as part of a non-time-aligned mix.

There's one more trick you can use, but only with the Aura, because it can work with a 100 percent wet mix, and that is to re-record the same part more than once, with different sound images each time. You don't get double tracking, as that relies on subtle timing and pitch differences, but you do get a musically useful result. You can sit them in the same place in the mix, for a thickening effect, or pan them apart for a wider quasi-stereo without any unwanted frequency cancellation due to the lack of correlation between sound images. *Dave Lockwood*

Terratec Producer Hi Jack

Guitar preamp

Hi Jack, from the 'Producer' division of Terratec, is a very simple preamp and impedance-matching box for high-impedance electric instruments. It is ideal for situations where you want to get a clean, unaltered guitar or bass signal into your recording system either for later re-amping or use with amp-modelling software.

Electric guitars work best with an input impedance of about 1MΩ (unsurprisingly, the approximate input impedance of most classic valve amp designs), as this ensures that their high-impedance pickups are not excessively 'loaded' by the input stage and that the higher frequencies are transferred intact. Active DI boxes usually offer an impedance of around 1MΩ, and these are fine in this sort of application so long as they are connected to a high-quality, low-noise mic amp. The Hi Jack, however, has up to 40dB of on-board gain available from its discrete FET circuitry, variable via its single control knob, and can therefore plug straight into a line input. The output is a low-impedance, balanced quarter-inch jack with a signal level anywhere between instrument and line level. The all-metal box is powered by an external 12V AC adaptor.

Hi Jack's input impedance is actually 2MΩ, although this is not a simple 'higher is better' situation. In fact, there is a complex relationship between the source impedance (the pickups and the volume pot), the input stage and the capacitance of the guitar cable, and if you stray too far from the 'normal' values, things often don't sound quite right. In practice, the Hi Jack's 2MΩ is fine unless you've already got a 'compensating network' on your guitar's volume control to alleviate top loss when you turn it down; then things can start to sound a bit brittle. A switch to drop the input impedance down to 1MΩ would have made a nice addition, but when the Hi Jack is so affordable it hardly seems fair to complain.

The Hi Jack is impressively quiet and seems to have ample headroom to cope with any passive instrument output. It imposes no character of its own and the sound doesn't alter with gain level. The only down side is that it weighs nothing at all and if you connect a guitar cable of any substance to it, you are liable to pull it straight off the work surface! The Hi Jack costs £49. *Dave Lockwood*

SUMMARY: 'Does exactly what it says on the tin': it provides clean, quiet, variable gain optimised for passive instrument-level input. Perfect for getting a guitar signal into a line-level input of an audio interface.
www.scvlondononline.co.uk
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Miami is now a hip-hop centre to rival New York and LA, and Cool & Dre are two of its most active beatmakers, songwriters and producers.

Cool & Dre

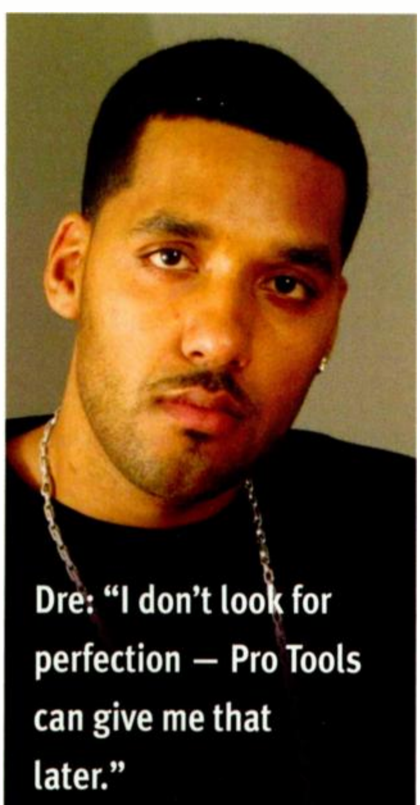
Producing Hip-hop

Dan Daley

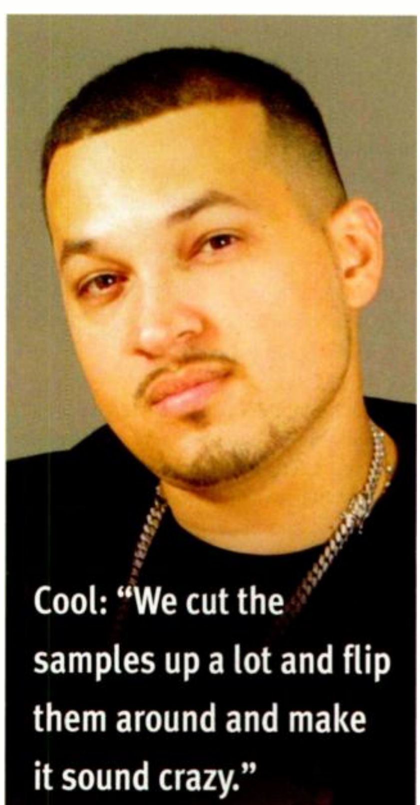
High school has been the starting point for countless bands over the last 50 years, as it was for Marcello 'Cool' Valenzano and Andre 'Dre' Lyon, who met when they sang together in the honours chorus at North Miami Senior High a decade or so ago. "You know when you meet somebody you just know, y'know?" says Cool, in a verbal shorthand that explains the chemistry between he and Dre. This is a chemistry that has resulted in huge hits for Fat Joe and Ja Rule, and beats, songs and tracks for artists including Diddy, Trick Daddy, Trina, Buju Banton, Juvenile and Killer Mike. They have also had soundtrack cuts in *Chasing Papi*, *All About The Benjamins* and *Bad Boys II*, and their own custom label deal with Jive Records, dubbed Epidemic Records, which is launching Miami rappers Dirtbag and Fat Joe Terror Squad alumnus Tony Sunshine.

Miami is rap's Third Coast, where the explicit rawness of Trick & Trina records mingles with Latino silkiness. In the legendary wars between New York's and LA's Urban gangstas, Miami came on quietly and then exploded. Rappers and producers including Timbaland have moved there *en masse* in the last few years and Cool & Dre, natives to Miami, have watched the migration with satisfaction. "Miami is the place you go to kick back while you work — go in the ocean, go to the clubs, go to the studio — all the same vibe," says Dre, a New Yorker of Jamaican descent who moved to Miami when he was six.

"We work a lot outside of Miami — New York, LA and Atlanta — but we do most of it right here," says Cool, originally from Venezuela. "It's the vibe, it's the weather. Even with the damn hurricanes, it's paradise. People feel creative. We go to Circle House [*Studios*] and you got a pool and a cabana. Hit Factory/Criteria is more corporate but incredibly professional. You want the A&R guys there. Shit gets done there. I don't think there really is a 'Miami way' of producing records, but it brings out creativity in different people in different ways. It's a place to hang out and bang out records."



Dre: "I don't look for perfection — Pro Tools can give me that later."



Cool: "We cut the samples up a lot and flip them around and make it sound crazy."

Getting Out There

"Bang out" is right. Like the other production teams in hip-hop, including the Neptunes, with whom C&D have often been compared, they are prolific, as the number of artists they've worked with attests to. However, tracks have a funny way of getting around, especially the one that gave the duo their first big hit. Ja Rule's 'New York' typifies C&D's sound, says Cool. "That beat was our sound, but what really made it different was that no one had put together those synths into a hip-hop track before," he says. The synths in question included Korg Trinity and Yamaha Motif ES7 keyboards and Novation Supernova and Roland JV880 and XV5080 sound modules, controlled from a Kurzweil PC88. "The sound worked in the street and in the clubs. It's a big sound, not just a beat.

"We were working with Jadakiss and Fat Joe was in the studios and heard it and said 'I got to have that beat.' So we said Jada, we love you, but Joe needs this record. But Joe takes forever to record. Finally we get into the album and Joe's like 'I'm not feeling this any more,' and we're like 'Damn, you made us take that from Jada.' So then Ja [*Rule*] hollers at us and we've got this beat still and Ja hears it and loves it. But it's still a beat; we've got the hook but not the song yet. Irv Gotti hears it and says the same thing. And I start singing '100 guns, 100 clips...' and Irv and Ja go crazy for it. The funny thing is, when we made the record, it ended up having Fat Joe and Jadakiss singing on it as guests. So everyone was all over that beat."

The hook in 'New York' hook is a faceful of synth pads, which overturned rap orthodoxy like electric guitars did in the late 1990s: an instrument at the time considered so off-genre that it was poised to make an impact the moment someone added them in a certain way. "The record came on people like 'What the hell was that?'" says Cool. "But that combination of hook and beat is the way we produce — no set formula, experiment with

A writing area set up in Studio E at the Hit Factory for Cool & Dre, with turntables, MPC2000 and their favourite synths.

sounds and beats, and see what works with the song."

Keeping Notes

Both producers have home studios, with the one in the guest room at Dre's house being the more sophisticated of the two. It's equipped with a Pro Tools HD system, though Cool says they also like Propellerhead's *Reason* and Apple's *Logic 7*, even *Garageband*. And as much as they enjoy messing with synth sounds, they prefer the virtual instrument route to collecting vintage keys. "I prefer that because then it's all there in front of you," says Cool, though he has a soft spot for their introduction to synthesis. "The Ensoniq ASR19 was our first synth. We were down at Ace Music and the guy pressed one button and the whole song played and that really set us off. We knew we had to learn to produce records."

Early production attempts taught them about the components of sounds. "One day we went to a studio to mix some stuff and the engineer said 'Those drums sound real weak. You can't just take the sounds as they are — you have to create them.'" They've been ardent samplers ever since, building their library into a 20GB mass using mostly the classic Akai MPC2000. "We cut the samples up a lot and flip them around and make it sound crazy," says Cool, who has been known to sample directly from his iPod. "It's noisy, but sometimes noise is a big part of a sound, like on a kick drum. We'll stack several kick drums together to create one big kick, but still keep it distinct from other elements in the low end. We put every sound in there for a reason." Sometimes they'll stack entire tracks: the groove on The Game and 50 Cent's 'Hate It Or Love It' used five separate samples from the Fabulous Tramps' 'Rubberband Man' that they had chopped and rearranged. (And properly cleared; hip-hop records now have more paperwork than the military.)

That huge sample library is kept on their *Pro Tools* drive, readily accessible for songwriting. "We used to use an Iomega Zip drive to store stuff, but now we'll dump beats right to *Pro Tools* and get rid of it on the drum machine. The ideas are all there at our fingertips." Cool & Dre are meticulous note-takers, naming each beat and logging it by date, then backing up to CD-R and cataloguing them by month. "We do a lot of original music, but our big hits have a lot of samples so it makes it seem as if we rely on samples," says Dre.

One thing that sets Cool & Dre apart from other hip-hop producers is an affinity for live musicians. "That's a big part of our sound," says Cool. "We have a group of musicians in Miami who are like family and we use them all the time in the studio. We work with a guitar player and do 30 takes and we'll keep all of them and then turn them into samples." However, live drums tend to be overdubs, adjunct to sequenced samples. "We'll have the drummer come in and do a fill over the tracks, or just play an open hi-hat, which is the hardest drum sound to get good as a sample. We'll have them play the whole track through and take the best four bars and loop them. Live players give a track that good dirt."

Robert Brisbane, a friend of the producers for the last eight years and their main recording engineer for the last five, says he counts on players like guitarist Dave Cabrera to walk into the studio with much of their sound ready-made with stomp boxes, and also likes to tap a DI through an Avalon mic pre and into the console. But he'll also experiment with microphones on amps. "One of the ways I get good results is to put a Shure SM7 close in on the speaker to capture the attack, and then put a U87 somewhere in the back of an open cabinet to get the amp's own ambience."

Mic Picks & Vocals

Just as C&D tend to stick with technical platforms that they are familiar with, like drum machines, their microphones tend to be drawn from a short list of favourites. "We stick to the standards — the [*Neumann*] U87, the [*AKG*] C12, and the Sony C800," says Cool. "The 800 has a nice high end to it, so it's good for female vocals and on a lot of R&B singers, like Remy Martin and Christina Milian, and helps the vocal ride above the track. The U87 has a lot of thickness — we used that on Trina's vocals and it sounds great." ▶

Brisbane says he likes to use a Neve 1073 or an Avalon VT737 mic pre, and gently touch up the vocal chain into a Urei LA2A, Neve or UA 1176 compressor. "If those aren't available, I'll go through an SSL channel and use those dynamics and EQ," he says. "But I'd prefer to start out with what gives me a great sound and not have to add EQ or other processing until later, if at all. The more you add, the more noise you add." Noise also comes in the form of vocals recorded by the artists themselves and emailed on MP3 files. "Then you have to use EQ to undo what they do to the tracks, which is annoying," he adds. Brisbane will often combine outboard effects with plug-ins, such as using an 1176 and a Bomb Factory *BF2A* compressor. "I love the convenience of pulling up a plug-in on a screen," he says, "but analogue compression gives a track some life when you're recording digitally." And like C&D, Brisbane tends to keep a limited selection of processors set to go in preconfigured ways when recording, ready to capture what's happening, and adds any processing later on. "Hip-hop sessions are impulsive," he says. "Not like a rock record where you spend a lot of time on the sounds before you hit Record."

Vocal sessions are straightforward affairs, and one of the aspects of record-making that Cool & Dre are very hands-on about (mixes tend to go to some of the heavyweights, including Serban Ghenea and Manny Marroquin). "We usually know which mic is going to work best, but sometimes we'll set up a few, and decide as the artists does a few passes to warm up," says Cool. But while they do like to have a complete vocal performance on a track, they're not at all averse to flying in the best bits of the repeating lines such as choruses. "This is the fly-in capital of the world," he laughs. "It's rare when you can find someone who can nail all six or seven harmony parts. Even when we were working on a DA88 in the beginning, after a certain point we just sampled it and flew the part in and then stacked them up."

Talking Perfection

Dre tends to be psychologist half of C&D; it's he who sets the mood for vocal sessions. "I'm an artist, too, and I know how it feels to be trapped in a vocal booth for two hours," he says. "Pro Tools has made the process of vocals very clinical, and vocals are the last really human part of the production process, especially in hip-hop, which is so machine-driven. I don't look for perfection — Pro Tools can give me that later. I try to get the feel of the song from the vocalist. I try to make the singer forget he or she is in the studio, because it's too easy to forget we're making music, not just a product. One of the things I do is tell the singer to talk some of the lines at certain points. When Alicia Keys sings 'You Don't Know My Name' there's a line in there that she talks — 'Let me call that boy' — and it's at that moment that you hear Alicia the person, not Alicia the singer. Michael Jackson used to talk a lot of his lines. It really gets the feel across on a song."

The essence of a producer isn't so much a matter of technical chops as it is an ability to relate to an artist. Cool & Dre have a touch when it comes to that. Tony Sunshine, one of Epidemic's first signings, says in a published interview of his production team: "Cool & Dre are like my brothers... I was signed to Loud Records, and when they folded, it was hard for me to get signed to another label because of politics and being Hispanic. Cool & Dre seen my struggle, and they've always been there for me. They had their own situation going on at Jive, and they felt that they had to come back and get the kid. They helped put me on the right track. When they got their little thing going, they... came back for me when no one else believed that I could do it."


Cool & Dre with Christina Milian.

Cool is appreciative of the sentiment, but says that it's what a producer should do. "A life story is important and we take that into consideration. You do a lot of records and often you don't get a chance to learn about people. But you figure out pretty quick when the chemistry is there. We did a couple of tracks with Sunshine and that turned into the whole album."

If the technical end of hip-hop sometimes seems less than complex, the politics can make up for it. They are generous in crediting influences including Sleepy Brown of Outkast and Goodie Mob producers Organized Noize, whose falsetto Dre can emulate precisely, and Cool cites Pharrell Williams of the Neptunes as another vocalist/producer who can cross hip-hop and R&B well. But the critics' comparisons of the two producer teams might have raised some hackles. "We dealt with that one night at a club where we ran into the Neptunes," Cool recalls. "We told them that that wasn't us saying that we were the 'new Neptunes', that we were the new duo in town. It was the magazines. We told them we were big fans of theirs. We said 'We respect y'all.' Without that, there would have been some tension. The Neptunes, they had a huge run. Most producers never get more than two, three hit records if they're lucky. But that's why we admire the Neptunes: they stay up there. That's what we plan to do." SOS

Keeping Track Of The Tracks

As well as looking after their beats and archives, Cool & Dre are scrupulous about keeping track of where their tracks are in the studio. The 'leaked' mix is becoming either a problem or a guerilla marketing tool; C&D prefer that it remain the latter, particularly since they are now label moguls themselves. "The whole download game has to do with control of the material, and to control it you have to keep track of it," says Cool. "Usually, you don't want leaks on a high-profile artist, though sometimes it can also be a good thing, if you're in control of it. A download about a week before the album release date can be a good time to do it on purpose. 'New York' got leaked and it was at a thousand spins before the album came out. But when we're in the studio, we try not to give copies of anything to anyone, including the artist. Sometimes the clients are the ones who leak it the worst, leaving it in a car or someone's house or through a 'friend'. At the end of each session, we back up to DVD and clean off the hard drive before we leave. You gotta control the track."

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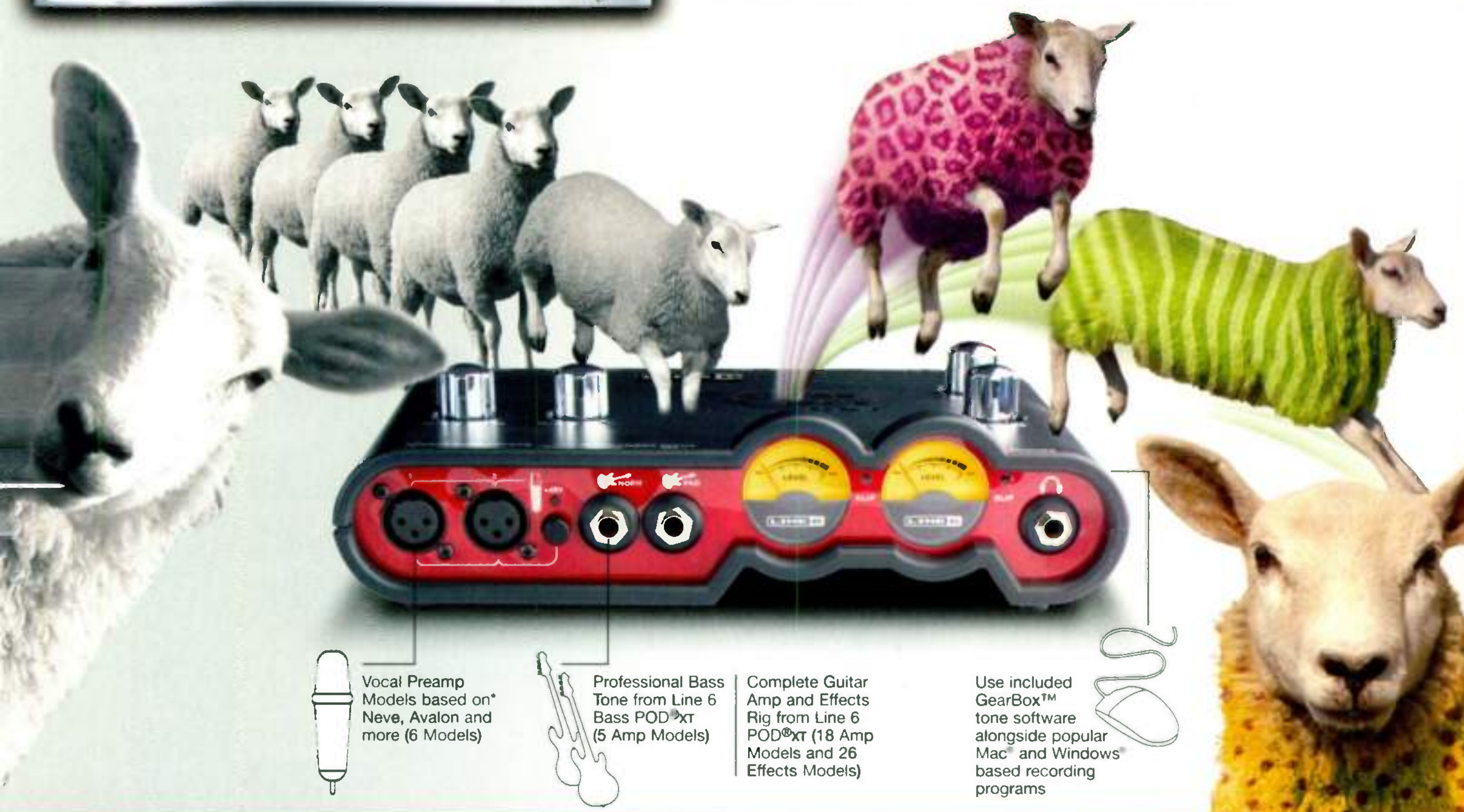

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In the blue corner is Intel's Pentium D range, offering good performance, PCI Express and good compatibility with existing PCI cards...

Twice As Nice?

... and in the red corner is AMD's Athlon X2, with a few compatibility problems, but easier to cool and offering better performance.

Making The Change To A Dual-Core PC

If you've been waiting and wondering whether to 'go dual-core' in your next PC upgrade, which processor to choose is only one of the factors to consider. We discuss the options and implications.

Martin Walker

I've always stuck by my recommendation to upgrade your PC's processor only when you can get at least a 50 percent increase in CPU performance. Any less than this and all the expense isn't really rewarded. However, for the many PC musicians, like me, with a Pentium 4 3GHz processor or similar, the required 4.5GHz model never appeared. Instead, Intel abandoned plans to 'break the 4GHz barrier', and both Intel and AMD switched to multiple-core technology, which places two or more processor cores on a single piece of silicon. Each of the cores would be under-clocked, compared with the single-processor equivalent, and would run at a lower voltage, so while a multi-core CPU would be faster overall, it shouldn't run much hotter or require an exotic (and possibly noisy) cooling regime.

Dual-core processors duly appeared from both AMD and Intel during 2005, but at first no-one knew what practical improvements could be expected with audio applications. However, *SOS* has now reviewed models from both camps, so I can safely say that for anyone who has been waiting for a good reason to update or replace their current computer, upgrading to a dual-core system is worth doing. In my case, I estimate that it could provide an increase in processing performance of 100 percent, or even more.

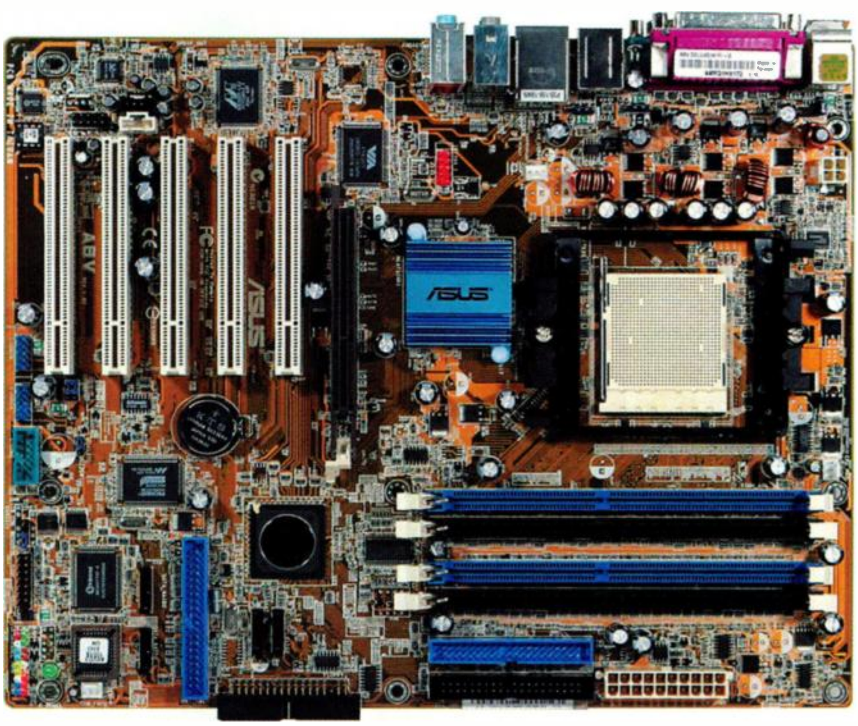
The Fly In The Ointment

Despite the undoubted advantages, many musicians are hanging back because of reports of audio conflicts, incompatibilities and other problems associated with the PCI Express chip sets used on many dual-core motherboards. I first discussed these problems in PC Notes September 2005, but apart from this exposure they are not widely known outside the *SOS* PC Music forum and a few other audio-specific forums. Some musicians have thus bought new PCs only to find themselves with a variety of symptoms, including unexpectedly high CPU readings and incurable audio stuttering. Some have found themselves having to run their audio interfaces with extremely high buffer sizes, resulting in latencies approaching 50ms.

If you already have a pre-PCI Express Socket 939-equipped AMD Athlon motherboard, there's one easy way to avoid such problems: see if a BIOS update is available from the motherboard manufacturer to support the newer dual-core Athlon X2 processor range. If this proves to be the case, all you have to do is update the BIOS of your current motherboard, remove its single-core processor, pop in a new dual-core model and carry on with greatly improved performance.

PCI Express

For those faced with the prospect of buying a new motherboard that supports either AMD or Intel dual-core processors, PCI Express



The easiest and cheapest way to achieve dual-core performance is to ignore PCI Express and use an older motherboard, like the Asus A8V shown here, coupled with an AMD Athlon X2 processor.

becomes a fundamental consideration. Should you buy a motherboard/chip set that supports it or not? Although I first discussed this new technology way back in PC Notes of November 2002, no hardware actually appeared on the market until mid-2004, and even now the practical benefits of PCI Express to the musician are shrouded in mystery. So what exactly is it all about, and just how relevant is it to the musician?

The classic PCI buss is 32 bits wide and runs at 33MHz, providing a peak bandwidth of 132MB per second. This bandwidth must be shared between all the devices attached to the buss, but nevertheless it's possible to run up to 32 input and 32 output channels of 24-bit/192kHz audio, and the huge number of very capable music PCs out there prove that PCI is quite sufficient for many of us.

PCI Express is a new, higher-speed buss that provides each device with direct access, rather than making it fight for its share of PCI bandwidth with other devices, and it also supports hot-plugging. Since a single switch-chip is in control of resource-sharing decisions it can also prioritise data packets, so that (for instance) real-time streaming of audio and video data can take priority over other less time-critical data.

Each connection between a PCI Express device and the PCI Express switch controlling I/O traffic is termed a 'link', and is composed of one or more 'lanes' that can send and receive one byte in each direction simultaneously. The simplest PCI Express expansion slot supports a 'x1' link that has one dedicated lane capable of transmitting 2.5 Gigabits per second simultaneously in each direction. With error correction it takes 10 bits to transmit an 8-bit byte, translating to 250 megabytes per second. PCI Express Motherboards generally offer a 'x16' graphics slot, running 16 lanes in parallel, to boost its bandwidth to 4GB per second, but it can support any combination of link widths from the range x1, x2, x4, x8, x12, x16 and x32.

AMD & Intel Costing Compared

If you want an Intel-based system, you only currently get three choices of processor in the Pentium D range, while the AMD Athlon 64 X2 range currently comprises five models. I compare the performance of the two ranges in the main text, but there are a couple of further points to bear in mind.

First, the AMD range has two cache sizes, and the jury is out on just how worthwhile the larger size is for audio applications. If you run a lot of reverb algorithms, it's possible that having a larger cache will benefit overall performance, but many experts don't think it's worth the extra money. Consequently, lots of musicians choose the 4200+ model over the 4400+ with identical clock speed, and the 4600+ over the 4800+. Anyone considering the 3800+ model should seriously consider spending a further £50 to gain the additional 10 percent of clock speed of the 4200+ model. The latter is the current favourite.

The second thing I want to point out is that while one of the older nForce3 or K8T800 Pro motherboards to partner an Athlon X2 can be bought for between £60 and £80, PCI Express motherboards to partner an Intel Pentium D processor may cost you £130. New technology is always more expensive, and the extra £50-70 may let you buy the next fastest AMD CPU.

CPU	Clock Speed	L2 cache	Approx Price
Pentium D 820	2.8GHz	1024KB	£170
Pentium D 830	3.0GHz	1024KB	£220
Pentium D 840	3.2GHz	1024KB	£360
Athlon 64 X2 3800+	2.0GHz	512KB	£230
Athlon 64 X2 4200+	2.2GHz	512KB	£290
Athlon 64 X2 4400+	2.2GHz	1024KB	£360
Athlon 64 X2 4600+	2.4GHz	512KB	£450
Athlon 64 X2 4800+	2.4GHz	1024KB	£570

Here are the eight dual-core processors most popular with musicians, along with guide prices, as of January 2006.

A PCI-to-PCI Express 'bridge' is needed to provide support for older PCI devices, and such bridging can either be on the motherboard (so that it can offer PCI slots as well as the newer PCI Express ones) or even on a PCI Express card (some early PCI Express cards may fit in the shorter PCI Express expansion slots, but they are actually PCI designs with an on-board bridge). Although many PCI devices seem to work fine in PCI Express motherboards with legacy PCI slots, it seems to be the bridging implementation on some motherboards that's causing problems when musicians plug PCI audio interfaces and DSP cards into them. Because the USB and Firewire ports all hang off the PCI buss, devices plugged into these can be affected as well. In the worst cases, even hard drive performance can suffer.

PCI Express Peripherals

So, after that little burst of theory, what does having PCI Express slots actually mean in practice to the average punter? Well, their main *raison d'être* at present is faster and more sophisticated graphics, courtesy of the single x16 PCI Express slot. However, for most musicians a performance boost over budget AGP graphics card is largely unwanted, because they don't need the extra graphic power to run 3D modelling applications and games and will usually want to avoid the cooling fans found on most faster PCI Express graphic cards. Furthermore, with some PCI Express chip sets, many of the current PCI soundcard click and pop problems can be reduced or eliminated by replacing esoteric x16 graphics cards with much more modest alternatives.

So if the x16 PCI Express slot doesn't end up all that exciting to most musicians, what can we do with the remaining x1 PCI Express slots? At this stage we have to take a leap of faith, since few PCI Express peripherals are yet available. I have spotted some TV tuner cards and a few PCI Express to Firewire adaptors, but for musicians I suspect PCI Express motherboards will only have something really ▶

significant to offer once soundcards and other audio-related products become available. One or two may be available by the time you read this, such as Digidesign's PCI Express Core and Accel cards, and maybe even new products from some of the audio interface manufacturers I talked to in *SOS* December 2005, who were cagey about revealing details of forthcoming products too soon. It's too early to say whether these will all simply be replacements for existing PCI models, or whether they will offer new 'must have' features, such as more simultaneous I/O channels than can currently be accommodated by PCI bandwidth. PCI Express also offers a feature called QoS (Quality Of Service), which some believe could offer guaranteed 'no-glitch' audio delivery, given suitable drivers.

Overall, PCI Express offers tremendous potential, PCI devices will eventually die out, and the current teething troubles will no doubt be eradicated as new motherboards are released. However, just at the moment, it offers few practical advantages to the PC Musician.

If you want a motherboard equipped with PCI Express slots, the current safest approach is Intel-based, and one of the most popular models is this Asus P5WD2.

Dual-core Chip Sets

The two most popular dual-core processor ranges are Intel's Pentium D and AMD's Athlon 64 X2 , so I'll be concentrating on them.

Those who want to build or buy an Intel-based dual-core PC must embrace PCI Express, since it's an integral part of the associated chip set. Intel were the first off the starting blocks with PCI Express-equipped PCs in mid 2004, courtesy of their 915, 925X, and 925XE chip sets for the single-core Pentium 4 processor range. However, audio problems mean that these cannot be recommended to musicians with PCI soundcards. (If you've already bought a PC with one of these chip sets you can greatly reduce audio problems by abandoning high-powered PCI Express graphics cards in favour of budget models.)

Intel managed to put audio problems behind them with the more recent 945/955X series chip sets for their Pentium D dual-core range. Systems using the 955X chip set in particular seem to be popular among specialist music retailers, with the Asus P5WD2 being the most popular motherboard. It offers three PCI slots, one x1 PCI Express slot, one x16 PCI Express slot, and one 'universal' x16 format slot that can run in x2 or x4 modes.

AMD enthusiasts face a rather more confusing set of choices. The easiest option (and currently one of the most popular) is to totally ignore PCI Express, as mentioned earlier, and simply opt for an older motherboard model that now supports the newer dual-core processors via a BIOS update. (Do be aware that you'll probably need a single-core processor in place first to update the BIOS).

The most popular motherboard in this category seems to be the Asus A8V Deluxe, which features the K8T800 Pro chip set, along with five PCI slots and one AGP slot, and comes highly recommended by many DAW builders, especially as it's one of the few X2 boards that's proved compatible with Universal Audio's UAD1 card (see box below). Another popular model with identical chip set and slots is Abit's AV8 (as used in the Scan PC system I reviewed in *SOS* January 2006). These both work well with most soundcards, including those from Edirol, Emu and M Audio, although Echo's five-year old Mia won't work with them.

Audio DSP Cards

Buying a dual-core system can be problematic enough in itself, but if you've already got PCI hardware that you want to carry on using with it, you have to be even more careful. Most PCI audio interfaces should prove compatible with the hardware I recommend in the main text, but a wrong choice could result in you having to increase buffer settings considerably, so I would recommend that anyone currently running an audio interface with an I/O count of more than 16-in/16-out contacts the local distributor of their interface to check on their model's compatibility with the chip set/motherboard they intend to buy. Better safe than sorry!

The hardest job falls to those with audio DSP accelerator cards, which can put a tremendous strain on any PCI-based system. Future PCI Express versions of the Powercore and UAD1 are likely to prove very popular for this reason, but if you've already invested in the current PCI versions, the PCI implementation in a dual-core system is particularly important.

The best place to check for up-to-date information is the DSP-card manufacturer's web site, but I've had reports from various musicians that the Asus A8V Deluxe is quite happy running UAD1 DSP cards. Making sure you're running the latest version 4 drivers helps. Universal Audio themselves say that their version 4 has an "optimised Turbo DMA engine" that "provides full support for the latest multi-processor and multi-core systems". In the (hopefully) unlikely event that Turbo DMA mode causes any glitches or drop-outs in your system that weren't there before, you can disable it with a special Registry key. Universal Audio specifically warn against using the Nforce 4 chip set, because it causes choppy audio and CPU spikes. They also provide warnings about other motherboards, which you can read at www.uaudio.com/support/software/UAD-1/motherboards.html.

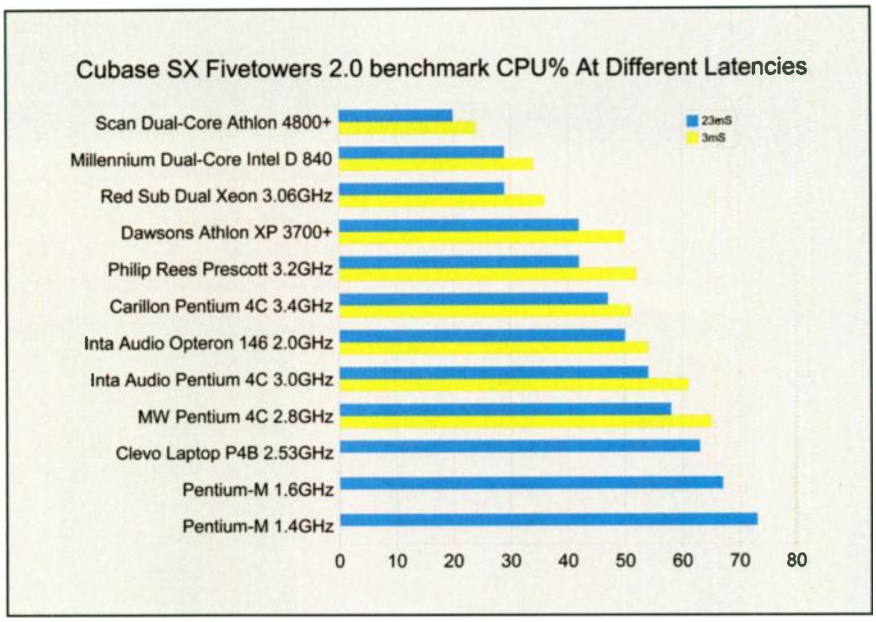
Owners of newer Powercore cards generally seem to run into fewer conflicts, and TC Electronic themselves say that no issues have been reported regarding Powercore Compact and Firewire units running with Athlon X2 systems. However, those with older Powercore versions may apparently run into problems getting them recognised by the BIOS on some of the latest computer systems. If you come across this problem, a workaround is to press the system restart button before Windows starts loading, to give the system a 'second chance' to detect it. TC are currently working on a driver update that they hope will resolve a few such issues with Powercore Element, PCI and PCI Mark II units.

CHANGING TO A DUAL-CORE PC

A third and possibly more compatible alternative (which Echo have confirmed works with their Mia) is MSI's K8N Neo 2 Platinum, which features Nvidia's Nforce 3 chip set and a similar complement of slots, but unfortunately also has a chip-set cooling fan, unlike the other two.

PCI Express arrived on the AMD scene in late 2004 with Via's K8T890, ATI's Radeon Xpress (the company's first foray into the AMD chip set market), and Nvidia's Nforce 4 chip sets, available in three 'flavours': Nforce 4, Nforce 4 Ultra and Nforce 4 SLI. Unfortunately, Via's K8T890 chip set proved to be incompatible with AMD Athlon X2 processors when they were subsequently released, and although a newer version of the K8T890 was released later, this chip set hasn't proved particularly popular with motherboard mainfacturers. A similar fate befell ATI's Radeon XPress.

This leaves the Nforce 4 chip set, which has proved very popular with gamers but extremely troublesome for musicians installing single-core Athlon 64 processors. However, as I write this in early 2006, the latest news is that if you instead install a dual-core Athlon X2 model the problems are significantly reduced — so much so that some DAW builders are reportedly starting to use Nforce 4 Ultra motherboards such as MSI's K8N Neo 4 Platinum model (with four PCI slots, two x1 PCI Express slots and one x16 PCI Express slot). Even Digidesign (normally extremely cautious to recommend the latest PC technology) are recommending the Asus A8NE motherboard, again with Nforce 4 Ultra, for use with their MBox and *Pro Tools LE* 7.0 for Windows XP.



Dual-processor results take the top three positions in my *Cubase SX* performance table, but the AMD Athlon X2 range is significantly ahead overall.

Audio Performance: Intel Versus AMD

Leaving PCI Express completely out of the equation for a moment, there are two further things to consider: acoustic noise and AMD/Intel relative performance. The first is easier to quantify: the Athlon X2 series dissipates about two thirds of the power of the Intel series, and it has proven easier to keep an AMD system cool with slower running fans (and therefore lower noise) than an Intel Pentium D system of roughly equivalent performance.

However, actually establishing systems of roughly equivalent performance is a trickier matter. Everyone is agreed that dual-core processors provide far better performance, when running most software applications, than their single-core predecessors running at the same core speed, but it's becoming very difficult to give hard and fast performance comparisons of today's computers, as relative test results seem to vary so much depending on the chosen application and its settings. The results are further confused now that the internal architectures of AMD and Intel processors are so different that you can now longer rely either on clock speed or model number to indicate relative performance.

I've seen test results I trust that suggest an AMD X2 3800+ system is roughly equivalent to an Intel D 820 system in performance on large *Nuendo* projects, although in other audio tests with Cakewalk's *Sonar* it takes a significantly faster AMD Athlon XP X2 4400+ model to equal the D 820. However, this balance shifts with latency: on this point the AMD 4400+ often moves significantly into the lead, with buffer sizes of 12ms and lower (after all, the more desirable values for 'real time' performance and playback).

My own results with *Cubase SX* suggest that in some tests AMD's slowest X2 3800+ model can hold its own against a rather more expensive Pentium D 840, so I conclude that, as well as being easier to keep cool, AMD's X2 series must be declared overall winners in the performance stakes, particularly at lower latencies. However, in its favour the Pentium D-series does offer a considerable performance boost over single-core P4s, and similar performance to some Dual Xeon systems at significantly lower cost.

Application Benefits

Since dual-core processors offer two CPUs running on one chip, you'll only achieve improved performance if your software can take advantage of both of them. As with a true multi-processor PC with physically separate processors, and Intel's Hyperthreading processors, this requires the software to support multiple software threads running simultaneously. Fortunately, most audio software shows significant performance improvements with a dual-core processor.

Applications confirmed to benefit from dual-core processors include Steinberg's *Cubase SX* and *Nuendo*, Cakewalk's *Sonar* and Digidesign's *Pro Tools* for Windows XP. Some other software may be written to run as a single thread, and will only therefore utilise one of the two cores on a dual-core processor. Since each of these cores runs at a slower speed than a comparably priced single-core CPU, this may actually result in worse performance.

One exception is Tascam's *Gigastudio*, whose sample-streaming engine is tightly written for a single-core processor. Polyphony shouldn't suffer, as this is more dependent on disk I/O than CPU performance, and *Gigastudio* plug-in performance can apparently take advantage of the other core.

Final Thoughts

So, after all these caveats and conundrums, should you take the plunge and upgrade to a dual-core system? I think the answer is yes, because of the potential improvement in performance. For instance, if you currently have a reliable but aging 3GHz Pentium 4 Northwood PC, or thereabouts, you will get about double the performance by buying an Intel Pentium D 840 PCI Express or an AMD Athlon X2 3800+ PCI system with Nforce 3 or K8T800 Pro chip sets. Those who can afford a faster X2 processor can achieve even more.

The big question is whether or not to adopt PCI Express. There are those that will ridicule buying 'old technology' like an Asus A8V motherboard with Nforce 3 chip set to partner an X2 dual-core processor, but the fact remains that if you've already got PCI cards you want to carry on using, and

Cakewalk's *Sonar* is one of the audio applications confirmed to benefit from dual-core processors.

particularly if you use Powercore or UAD1 cards, such a system is currently likely to provide the best performance.

Having some PCI Express slots will provide you with some future-proofing and the option of higher-powered graphics, and for greatest compatibility you should currently buy an Intel Pentium D system, at the expense of slightly reduced performance. Buying an Athlon X2/Nforce 4 system will provide better performance than this, but the UAD1 won't work with it at the moment, and you may still not get quite the same overall performance as a cheaper Athlon X2/Nforce 3 system.

Of course, there's another approach, and that's to wait a little longer. There are plenty of interesting releases on the horizon, including AMD's 940-pin Socket AM2 format, expected by mid 2006, along with a new range of compatible X2 processors. By that time the current AMD/PCI Express audio issues may have been resolved. Intel may have released their Yonah CPU range, a dual-core version of the extremely popular Dothan models used in Centrino laptops, by the time you read this, and if it proves to be as powerful and easy to cool as some are hoping, and finds its way into desktop systems, musicians may be queueing up to buy it instead of a Pentium D system. However, there's always something better around the corner, and many of us have been waiting long enough! SOS

A software editor is available allowing you to edit and archive Voice Pro patches. Here you can see the stand-alone version, but there is also a version which can operate as a VST plug-in within your sequencer, enabling patches to be stored within each of your projects.

sent between the Voice Pro and a MIDI sequencer/keyboard. A standard BNC word-clock input is also present, and digital I/O is provided via AES-EBU on a 25-pin D-Sub connector. When using the digital I/O, the outputs can be configured to split the dry lead, Virtualead, and four harmony voices to individual outputs for external mixing.

Slightly more odd inclusions on the rear panel are a pair of Ethernet ports and an RS232 port. The manual suggests that the Ethernet ports are to be used for implementing software upgrades and that the RS232 port is 'currently not supported'. I can only assume there is some technical reason for the use of the Ethernet ports rather than either USB or Firewire connectivity.

For those taking advantage of MIDI to connect the Voice Pro to a sequencer, one additional element not included in the box is very much worth acquiring. A free software editor (produced by Psicraft) is available for as a 13MB download from the TC-Helicon web site. The editor can run as a stand-alone application or as an MFX or VST plug-in.

Basic Operation

For the purposes of testing, I hooked up the Voice Pro via its analogue I/O to my mixer and fed the unit with some vocal parts via *Cubase SX*. The manual makes some comments about the latency induced by the processing of the Voice Pro, and a Utility page allows this latency to be adjusted between 'minimum' and 'high' settings, with various stops between. This setting only affects the latency of the Virtualead output (between 15ms and 34ms), whereas the latency of the dry lead and harmony voices stays at 2ms and 34ms respectively regardless. While the manual does not make it explicit, I assume the lower-latency settings are achieved by some compromise on the processing quality. This seemed to be confirmed in auditioning the Virtualead vocal in isolation, although the differences were only really noticeable on the very lowest-latency setting and on heavily processed sounds. In a studio context, the latency can, of course, be compensated for by sliding track timings within your sequencer.

Although most operations can be carried out using just the Navigate cursor keys and main Select/Scroll wheel, things are made considerably easier by the four Edit knobs, the functions of which are defined by labels on the currently active screen. Given the multitude of processing elements, editing is a little daunting at first. However, even with only a little experimentation the process soon becomes familiar, and the unit is by no means difficult to use. Processing blocks are accessed via the appropriate button (Harmony, Multi-FX, Pitch, or Character, for example) and pressing the OK button will toggle a block on and off. Editing the parameters for each block simply requires a combination of the cursor keys and the Soft Knobs and, given the excellent display, it's actually quite speedy.

Sound Effects

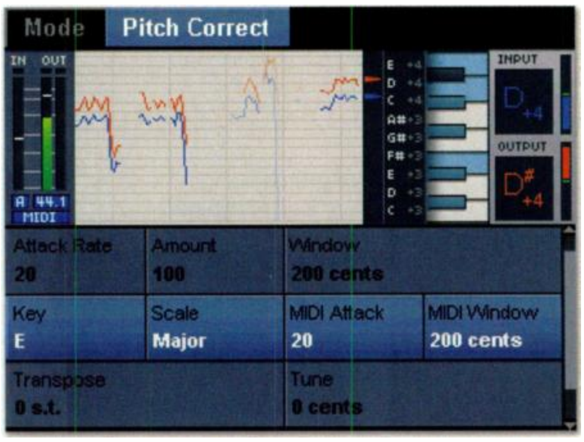
While the Voice Pro's generic effects are not the key selling point of the unit, they are worthy of some comment. Under the Multi-FX group are the µMod, Delay, and Reverb effects, and these operate as send-return loops within the Voice Pro — a separate send control is available for the dry lead, Virtualead, and harmony parts to this effects chain. The µMod effect provides a variety of chorus, flange, panning, and other detuning-style effects, with some weird and wonderful special effects thrown in for good measure. There is plenty of user control on the effects, with over a dozen editable parameters for those people that like to tweak. Each of the other generic effects has a similar degree of editing, so you can see that this is a serious multi-effects processor.

The Delay and Reverb blocks both offer flexibility and, to my ears at least, very good audio quality. For example, the reverbs — which I suspect borrow heavily from parent company TC Electronic's other processors — are very nice indeed. The Dynamics block, which provides compression, limiting, and de-essing, and the EQ block are equally well specified. For example, the compressor features threshold, ratio, knee, release, and make-up gain controls, and can be positioned before or after the EQ. The display also includes a graphic showing the amount of gain reduction. The EQ block has low and high shelving filters with variable frequency, and two fully parametric mid-bands. A further low-cut filter is provided with presets to roll off below 60, 80, or 120Hz.

The Transducer section provides a range of more specialised effects. Here, you can simulate the sound of the voice being passed through a radio, a megaphone, a telephone, and various amplifiers, and there are some additional sound-mangling options, including a nice tube emulation that is capable of adding just a little warmth.

Pitch-correction & Harmony Generation

Three modes of pitch-correction are provided by the Voice Pro. The scale-based automatic mode is, in essence, similar to the automatic mode of *Auto-Tune*. The user specifies the key, scale, attack rate, and amount of pitch-correction to be applied, plus a 'window' around each scale note where correction will occur. As with *Auto-Tune*, custom scales can be defined if required. Provided that you do not expect it to rescue a very poor singer, the results are transparent and comparable to those obtained with *Auto-Tune*. That said, if you do want to abuse the pitch-correction for creative effect, then there's plenty of scope to do that.



This screen shows the settings for the automatic pitch-correction algorithm.

The second mode of pitch-correction also uses scale-based automatic adjustment, but allows the user to override this on a temporary basis via MIDI. Any MIDI note played will become the target note until released. This can be used to fix the occasional note that lies outside the specified scale. The final pitch-correction mode simply takes this MIDI control further for full 'manual' playing of the pitch, although this requires some MIDI editing within your sequencer to create a natural result.

The largest number of the Voice Pro's 250 factory presets focus on automatic double-tracking and harmony generation. For double-tracking, under the Harmony block's Humanise page, the user can control the degree of pitch and timing tightness to the original dry vocal. In addition, the style and degree of both vibrato and scoop (the 'scoop' that most singers apply at the starts of notes as they slide into the required pitch) to be added to the double-tracked vocal can be adjusted. As suggested by a range of the preset names (for example, Subtle Double, Thicken, Loose Double, Octave Double), almost any combination of super-tight through to downright sloppy double-tracking can be created.

Further variety can be added by including elements from the Character block to give the double-tracked voice a different character to the dry lead vocal — more on this in a moment. While some of the Character-based presets do require careful use, the double-tracking can be very convincing even when heard in isolation. Within a full mix, even some of the more extreme presets are totally acceptable. Given the degree of control available, the ability to ▶

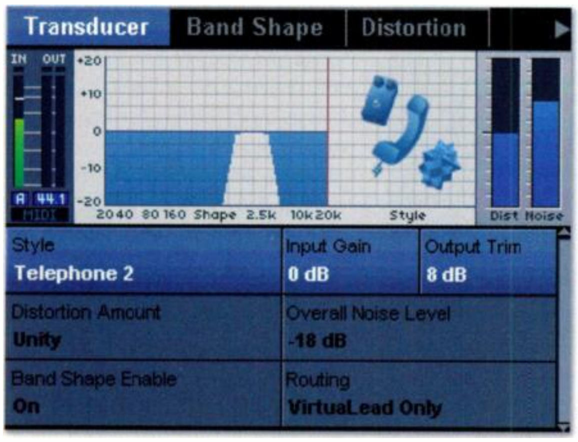
TC-HELICON VOICE PRO

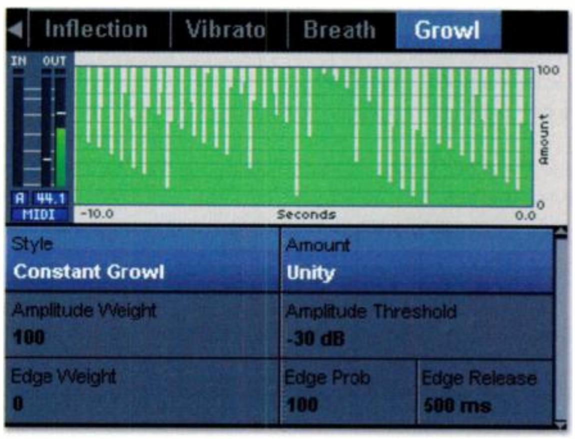
simply create the 'ideal' double-tracked feel from your original vocal line is hugely impressive.

The various harmony presets allow up to four-part harmonies to be generated in exactly the same fashion as with the double-tracking — and with the same degree of control over the relative timing and pitch tightness for each of the four voices. The Voice Pro can construct the harmony parts in a number of different ways, but the three most useful modes are Scale, Chord, and Note, each of which will produce a different style of harmony. In Scale mode, the key and scale need to be set (custom scales can be created if required) and the pitch of each harmony part is set relative to the pitch of the dry lead vocal. For example, harmonies might be set at +3rd or -6th relative to the lead voice, and a range of ±2 octaves is available. This mode tends to produce fairly dynamic harmony parts, as every change in pitch in the lead vocal also produces a change in pitch in each harmony part.

In Chord mode, the harmonies are created intelligently from chords received via the MIDI In (most easily done via a sequencer, although you could play them in live). This tends to produce slightly less pitch variation within the harmony voices, and would certainly suit some kinds of material better than Scale mode. For total control over the harmony construction, Notes mode uses the pitch of MIDI notes to define the harmonies. Harmonies are only generated when notes are being played, and this can work very well if you only require certain words within a vocal line to be harmonised.

There are all sorts of additional settings within the Harmony block that offer further variety and help create a natural end result — volume, pan, and gender can also be adjusted. Many of the presets also add in some further elements from the Character block to help differentiate the various harmony voices even further. As with the double-tracking, the end results of the automated harmony parts can be very, very good indeed. The more subtle ones work well enough *a cappella*, but within a mix even some of the more adventurous presets can work well enough to be convincing. What's more, the Voice Pro provides a means of experimenting with harmony parts that simply wouldn't be possible with a live singing group — unless they happened to be extremely talented, patient, and have the stamina to perform for hours on end!

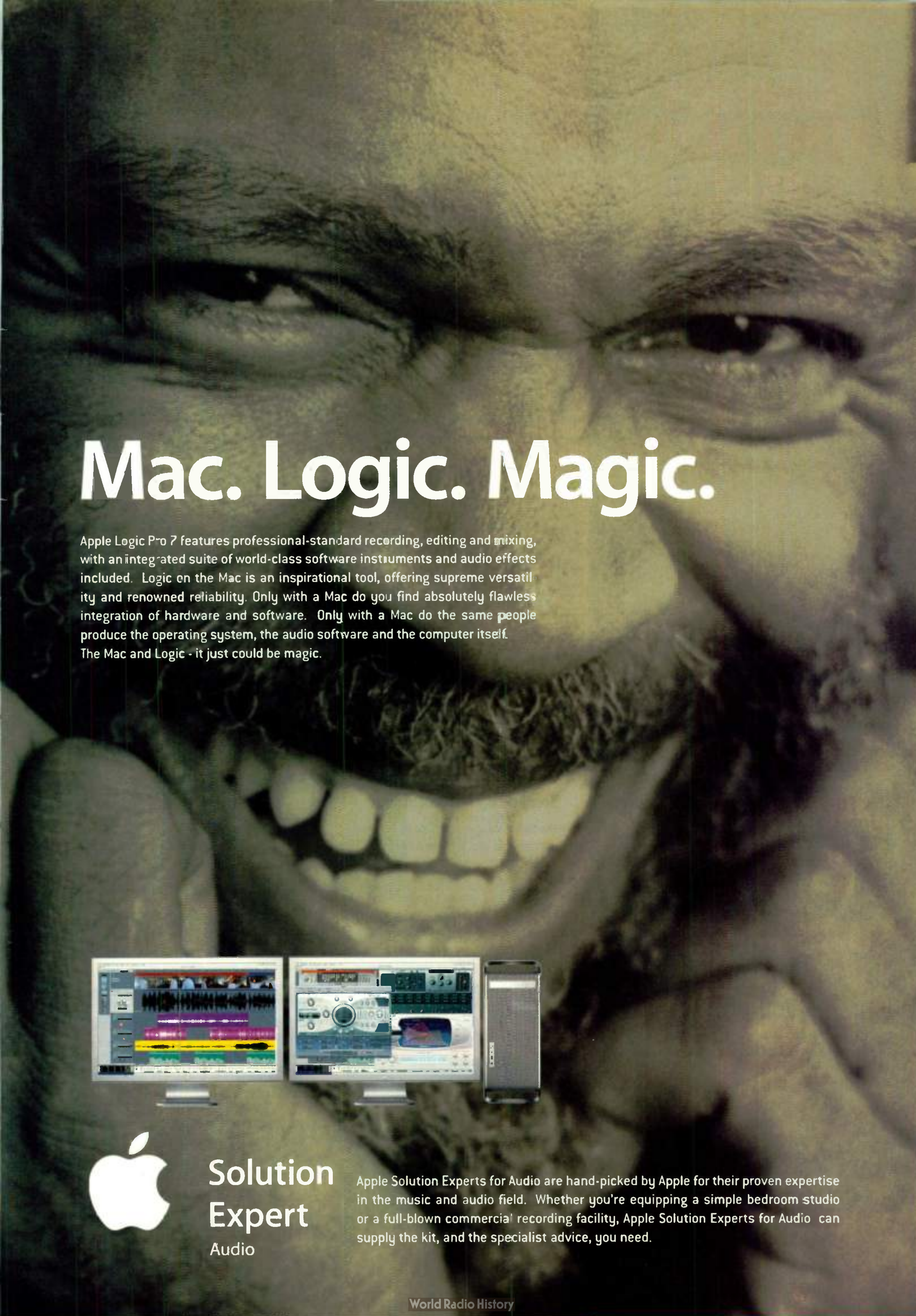




Alongside common dynamic and EQ processes available within the Voice Pro are more unusual and creative effects. The Transducer options (top) allow you to put the vocal through simulated electronic devices such as telephones or valve amps, while the Growl page (bottom) can make the voice rougher, grainier, and more 'rock & roll'.

Character Assassanation

We last looked at TC-Helicon's voice-modelling technology in the review of the TC-Helicon *Voice Modeller* plug-in back in *SOS* February 2004. The Character processing block contains six sub-sections: Resonance, Spectral, Inflection, Vibrato, Breath, and Growl. The Resonance settings produce changes in the position of the harmonic content of the voice. For example, the Length parameter simulates a change in the length of the vocal tract


Mac. Logic. Magic.
Apple Logic Pro 7 features professional-standard recording, editing and mixing, with an integrated suite of world-class software instruments and audio effects included. Logic on the Mac is an inspirational tool, offering supreme versatility and renowned reliability. Only with a Mac do you find absolutely flawless integration of hardware and software. Only with a Mac do the same people produce the operating system, the audio software and the computer itself.
The Mac and Logic - it just could be magic.
Solution
Expert
Audio
Apple Solution Experts for Audio are hand-picked by Apple for their proven expertise in the music and audio field. Whether you're equipping a simple bedroom studio or a full-blown commercial recording facility, Apple Solution Experts for Audio can supply the kit, and the specialist advice, you need.

Beyerdynamic MCE72

Stereo Condenser Microphone

Paul White

If you're looking to record high-quality audio to your portable Minidisc or MP3 recorder, then this small battery-powered stereo mic could be just what you're looking for.

The Beyerdynamic MCE72 must be the answer to many a location recordist's prayers: it is a stereo back-electret capacitor microphone that offers a near-professional technical spec, yet operates from a single 1.5V battery. This means that if you have a portable MP3 or Minidisc recorder, or a video camera with a low-quality built-in mic, you can now team it up with a serious microphone without having to worry about where to find a source of phantom power. Indeed, the Beyerdynamic MCE72 isn't designed to run off phantom power at all — it's strictly batteries only.

In appearance, the MCE72 looks much like any professional stick mic, except that a fine line around halfway up the body indicates the part where the outer shell unscrews to reveal the battery compartment. A single alkaline AA battery is all that's needed to drive the mic, and the operation time is in the order of 75 hours, depending on the brand of battery. A slide switch on the side of the body switches off the microphone and conserves battery power, and there is also an interim position that illuminates an LED to show that the battery is OK. The overall size of the mic is 25 x 196mm, and the correct orientation of the mic head is shown by an on-body schematic — the end of the mic should be pointing towards the action. A cable is included, where the other end of the five-pin XLR lead terminates in a stereo mini-jack, which is probably the most common connector type for consumer-style portable recorders. Adaptors to convert to other formats, such as quarter-inch jacks or RCA phonos are readily available.

The capsules themselves are cardioid (pressure-gradient) types and are angled at 120 degrees (plus and minus 60 degrees either side of the frontal axis) to give adequate stereo coverage under most conditions. A frequency range of 60Hz-20kHz is quoted, though without those all-important 'how many decibels down' limits, and the sensitivity is a realistic 12mV/Pa. A small frequency-response plot in the back of the manual shows the curve to be nominally flat, with a very gentle low-end roll-off starting at just above 200Hz, and the high-end starting to roll off at 16kHz or so. From this I'd estimate that the 60Hz-20kHz response refers to the -6dB points.

As you might expect from a microphone that's powered from a low-voltage source such as a battery, the headroom is somewhat limited compared with what's achievable with 48V phantom power, and in this case, the maximum SPL is only 123dB, but again this is fine as long as the mic isn't used close to something exceptionally loud — such as when recording in the same time zone as a Prodigy concert, for example! The 'A'-weighted equivalent input noise is 26dB, which is a bit on the high side if you take a typical studio

SOUND ON SOUND

Beyerdynamic MCE72 £188

pros

- Operates from batteries, so no phantom power is needed.
- Generally honest, unhyped sound.
- The UK price is sensible, as most stereo mics cost rather more.

cons

- Higher noise figure than for typical studio capacitor mics.
- Headroom limited because of battery powering.

summary

The MCE72 delivers near-studio-quality performance in a portable battery-powered format. In this respect it provides a practical means to upgrade recording systems based around consumer Minidisc recorders or video cameras.

mic as a guide, but again it shouldn't be a problem if you're recording medium-to-loud sources close to the microphone. If you're looking to capture the mating calls of crickets at 500 metres, expect a little hiss!

A standmount clip comes with the mic, as well as a cable and a zip-up vinyl pouch, but the optional shockmount would be a good idea when using the mic in environments prone to vibration or on top of a video camera.

Testing, Testing...

My test with voice and a variety of sound sources showed this mic to be generally flat and honest-sounding, which means it will capture most sounds without undue coloration or flattery. To check its sensitivity I recorded an acoustic guitar from around 12-14 inches away, and found the signal-to-noise ratio quite acceptable, though it was possible to detect a low level of background hiss when nothing was playing if monitoring on headphones. For that reason, I wouldn't recommend using this mic for anything much quieter or much more distant. For jobs such as nearby dialogue, choir, and acoustic-ensemble recording, it should do the job perfectly well. It could also be useful for collecting sound effects on location, provided that they aren't too quiet.

Although the MCE72 falls a little short of what you'd expect from a good studio mic in terms of noise and headroom, it still provides much better audio quality than you'd normally expect from a battery-powered back-electret mic, and should significantly improve the results when recording on domestic MD recorders and suchlike. Its stereo capability is fine for most tasks, though not being able to adjust the angle between the mic capsules means that in some situations the stereo image may be slightly too wide or too narrow. As a general-purpose stereo mic for use in situations where there is no phantom power, the MCE72 gets the general thumbs up, provided that it's used within its limitations, and as a means of recording live gigs onto consumer equipment it's almost ideal. In this respect it's a bit like a good-quality 'point and shoot' camera — you can get good results provided that the subject is not too dimly lit, not too brightly lit, and not too far away! SOS

Information

£ MCE72 with cable and MKV8 stand clip, £187.53; MCE72 with cable and EA86 shockmount, £217.14. Prices include VAT.
T Beyerdynamic +44 (0)1444 258258.
F +44 (0)1444 258444.
E sales@beyerdynamic.co.uk
W www.beyerdynamic.co.uk
W www.beyerdynamic.de

On-line Mastering Services On Test

Web Masters

A new business model based around the Internet promises to make professional mastering more convenient — and much more affordable. But is on-line mastering a real alternative or a false economy? We put the UK's leading services to the test.

Sam Inglis

The received wisdom about mastering goes something like this: mastering is a crucial stage in the production of any single or album, and one that shouldn't be done by the engineer who mixed the music. Instead, someone with specialist expertise and a fresh pair of ears should balance the levels of the different tracks, EQ them for a consistent sound, compress and limit them for maximum punchiness and handle any fades and segues.

That's all very well if you've spent three months in Abbey Road recording your masterpiece, and the money you're spending belongs to a large record label. Sadly, most of us aren't in that position, and for people recording at home or in low-rent studios, the cost of professional mastering has always been a stumbling block. A typical album session, with the client and the mix engineer in attendance, would take at least a day of the mastering engineer's time, and could set you back at least £1500, often substantially more. Add to that the cost and inconvenience of getting to the studio, and it's easy to see why a lot of people either try to do their own mastering or simply don't bother.

Now that broadband Internet access is widespread, however, there's another option: some mastering houses have begun to offer Internet-based services. The source files are submitted electronically via an FTP or Web site, the engineer works on them at his or her convenience, and the results are either made available for download or returned via the post if a physical master disc is required. These services are far more affordable than an attended mastering session. Having a 10-track album mastered on-line could cost as little as a third of what you would pay for a conventional session with the same mastering engineer! What's more, since on-line services charge by the track rather than by the hour, they make it much easier to calculate mastering costs within a fixed budget.

I wanted to find out whether on-line mastering is as good a deal as it sounds. Does the Web really provide a good enough

Three mastering houses have teamed up under the Mastering World banner: Donal Whelan's Hafod Mastering (top), John Dent (below left), Jason Mitchell and John Wilkins of Loud Mastering, and Simon Heyworth's Super Audio Mastering.

interface between client and mastering engineer? Will someone who's charging as little as £35 per track really do a professional job on your music? And most importantly, how great are the differences between the various on-line services on offer, and which one should you choose? To find out, I approached three of the UK's leading on-line mastering specialists, with the aim of comparing their services and the results they could achieve. We agreed that I would submit the same three tracks to each, with the same instructions. I would then make the resulting files available to the artists and engineers responsible for those tracks in a blind test.

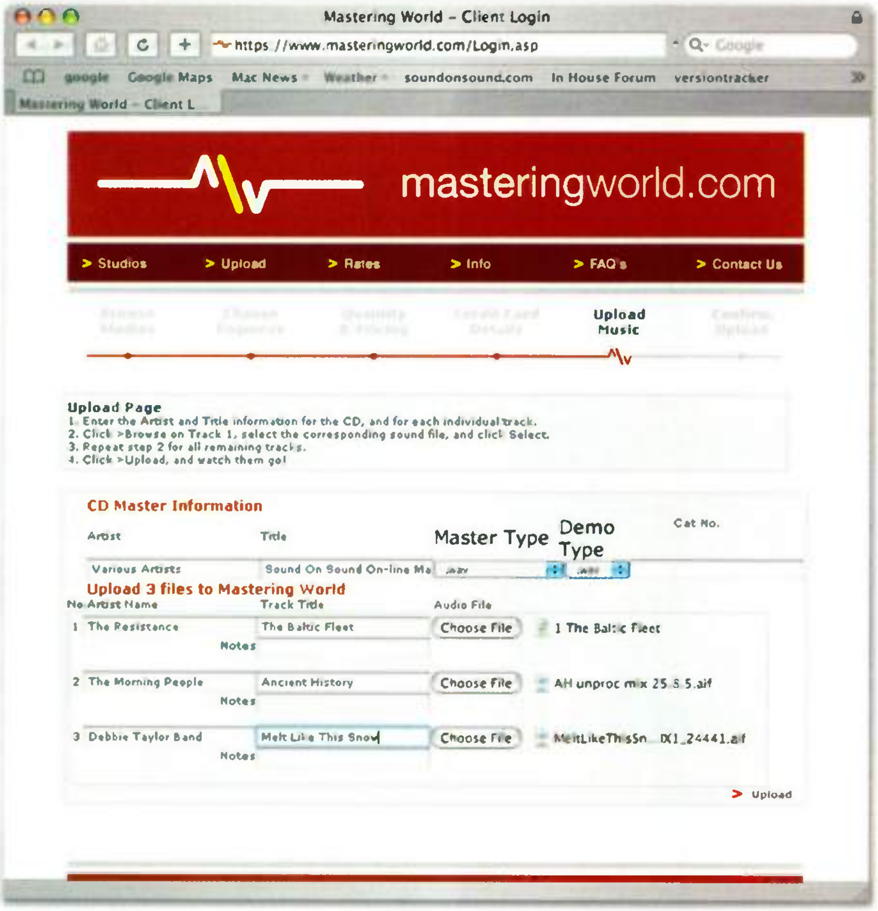
Mastering World

The first service I went to was Mastering World, which has been set up in collaboration by three established British studios: Loud Mastering, Super Audio Mastering and Hafod Mastering. All the engineers involved have long track records in the business and lengthy credit lists; you'll see all of their names on plenty of recent major-label releases.

Initially, I'd only intended to use one of Mastering World's studios, but it turned out that all three were keen to have a go at our test tracks, which was great. Although they all use the same on-line front end, Mastering World's partners include both the most affordable (Hafod), and the most expensive (Super Audio Mastering) mastering services in the whole test, so the inclusion of all three should give us a clearer picture of what you get for your money.

Mastering World's web site is admirably easy to use. Before you get started, you can view profiles of the engineers and studios whose services are on offer, and once you've selected an engineer, you can then click Upload. At this point you'll be asked how many tracks you wish to upload, and a quote will be generated based on a pricing structure that depends on which engineer you choose. The most affordable is Donal Whelan of Hafod, who charges £50 for a single track, £40 apiece for two to five, and £35 per track for larger jobs; at the other end of the scale, Simon Heyworth and John Dent — the big names on Mastering World's books, with credits to match — will cost you between £100 and £150 per track, depending on what special offers are running when you book. Non VAT-registered clients in the UK and Europe will also need to pay VAT on top of the total cost.

New clients then need to register, which involves filling in a simple ▶



Preparing to upload tracks to the Mastering World server.

▶ Web-based form. Mastering World uses the popular Worldpay engine for credit card and debit card payments, which accepts all card types in common use and seems to work very smoothly. I received email confirmation of the transaction about five minutes after making it.

Once your payment has been accepted, you can then proceed to upload your music. This, again, is done using a Web-based form on the Mastering World site, rather than an FTP client. This is fine, except that it won't let you select a whole bunch of tracks and apply the same artist information to all of them in one go, which could be a pain if you're mastering a lengthy album. For each track that you upload you can enter comments to be passed on to the mastering engineer. In this case I included sample-rate and bit-depth info, and a note to point out that the three tracks were not intended as part of the same album, so there was no need to attempt level-matching and so on. Once you then select a file and click Upload, a window appears charting the progress of the transfer. From our fast Internet connection, an 86MB AIFF took less than six minutes to upload to Mastering World's server. The only possible fly in the ointment I could spot was that there seemed to be no way to upload tracks consisting of more than one file, such as split stereo files, but this shouldn't be a problem for most people.

Once your tracks have been uploaded, you can enter a title and artist name for the album, and select the type of master you want from WAV, AIFF, *Jam*, *Nero* or DDP. For some reason, the summary field showed incorrect lengths for the 24-bit masters I uploaded, and pressing the Continue button didn't do anything. However, Mastering World's Donal Whelan assured me that these were minor bugs in the system which will be cleared up soon.

One of our test tracks, 'The Baltic Fleet' was mixed almost in mono, and I was impressed that Mastering World's engineers spotted this and emailed me to find out if it was intentional before going ahead. It's easy to accidentally convert a stereo mix to mono without noticing, so full marks to them on their attention to detail.

When they have done their stuff, your project will be available from the Mastering World site as a single, Zipped 'audition' file. The idea is that you download this, unZip it and check the songs to see that everything is OK. If it is, you can then tick the 'Approved' box on the site, and your DDP file will be made available.

Metropolis iMastering

The Metropolis Group, based in West London, offer a whole range of creative services from tracking and mixing to CD packaging and design. Their mastering division is one of the most successful in the world, and employs no fewer than 10 mastering engineers.

Metropolis's iMastering service doesn't give you a choice of engineer, simply promising that mastering work will be undertaken by an 'unspecified Metropolis Mastering Engineer'. Unlike Mastering World, they offer mastering for vinyl in addition to

The Music

As you'd expect, there's a lot of music going on in and around the *SOS* office, so I took my sample tracks from three bands featuring *SOS* staff members — this was the easiest solution from a practical point of view, and I think we were all selfishly curious to see what would happen to our music! I also felt that it was important that the results could be auditioned by the people who'd originally recorded the tracks and knew them best. I wanted the range of music on offer to be as diverse as possible, and *SOS* staffers' musical interests run the gamut from delicate folk to full-on electronica:

• The Debbie Taylor Band: 'Melt Like This Snow'

Taylor is the stage name of *SOS*'s own Debbie Poyser, who sings and plays rhythm guitar on this acoustic ballad. The rest of the instrumentation features live drums and upright bass, and lead acoustic parts played by *SOS* Publisher Dave Lockwood. Dave also tracked the song, which was mostly recorded live, and mixed it in *Logic* using TC Powercore effects. He supplied a 24-bit, 44.1kHz file for mastering.

• The Morning People: 'Ancient History'

This slightly frenzied country-rock song was recorded by my band, the Morning People, which also features News Editor David Greeves on electric guitar. I sing and play acoustic guitar, there's live fretless bass (DI'd) and drums, and a MIDI keyboard part which was bounced to disk using the *Galaxy Steinway* VST piano instrument. The band was tracked live to *Pro Tools* via a Mackie desk and M-Audio Firewire 1814 interface. I later replaced the lead vocals and acoustic guitar and added backing vocals. Again, the master file was a 24-bit, 44.1kHz AIFF.

• The Resistance: 'The Baltic Fleet'

Assistant Editor David Glasper's instrumental band features him on a Mac laptop, running Ableton *Live* and NI *Reaktor*, and two guitarists, both DI'ing their instruments through lengthy effects chains. 'The Baltic Fleet' combines glitchy electronic drums with atmospheric tones and psychedelic waves of noise. David gave me a 16-bit, 44.1kHz AIFF file for mastering.

One of the mastering studios at Metropolis, in West London, as used by their iMastering service.

CD: the former costs £100 per side and the latter £75 per track, not including VAT. You can choose to have your masters delivered as downloads, production master and reference CDs, dub plates and/or lacquers. Even the downloadable CD master includes a DDPI file and *Nero* or *Toast* files as well as the WAV audio, which is a nice touch. Your files can be 24- or 16-bit and any sample rate up to and including 96kHz, but higher sample rates are not supported. Once again, split stereo files are not accepted.

Here too you have to create an account, and this time you need to wait for a password to be emailed to you in order to proceed. However, this only took a couple of minutes in my case, and you can then enter your payment details. Like Mastering World, iMastering uses the Worldpay system.

The upload system adopted by iMastering uses quite a complex Java applet, which will only work with some recent Web browsers; in particular, Mac users will need to be running *Safari* or *Omniweb 5*, and not *IE* or *Firefox*. This applet displays the file structure of your local machine in conventional 'tree' style on the left, and the iMastering secure server on the right. I found navigating through my files slightly harder using this applet than the system adopted by Mastering World, but it really wasn't at all difficult, and because you can select batches of files, it would be more convenient if you were handling a large number of tracks. Once again, a status bar pops up when you transfer each track, and the transfer rates were about the same as for Mastering World.

'Engineer's Notes' are filled in in one go, in a separate form on the iMastering site. Required information includes filename, track title, artist and approximate duration. You can also add ISRC codes and any notes you might want to communicate to the engineer. Once again, I used this feature to point out that these three tracks were not supposed to form part of a consistent whole.

When all is complete, you click a button that says 'I've finished, please master my tracks!' You'll receive a confirmation of your order via email and Metropolis's finest will get to work.

Like Mastering World, iMasters also came back to me with an emailed query about the Resistance's track. Their concern was that the mix had been heavily compressed using multi-band compression before submission, and they asked if we would like to submit an unprocessed mix instead. This surprised David, because he hadn't in fact used any compression on the mix of 'The Baltic Fleet', but I was pleased that they contacted us to check this out, rather than simply going ahead.

When the masters are done, you'll receive another email to tell you so, and you can then download them from your account with the same Java applet used to upload them. You need to do this within two weeks.

Emasters

Emasters, the on-line arm of The Sound Masters, differs from the other two services ▶


Emasters is the on-line division of The Sound Masters, and uses their main mastering studio as well as a separate edit suite.

on test here in that you don't have to pay for your mastering up-front. Instead, you fill in a simple registration form on the site to create an account. Once you have an account, you'll find 1GB of free library space on their server available to you, and you can upload files to this at will without committing yourself to anything. If you do go ahead, you'll be charged £50 per track for mastering, with additional charges for vinyl lacquers and dub plates; if you want a physical CD-R, that'll be an extra £2. The upload procedure is similar to that adopted by Mastering World: you click on a Browse button, then use the Mac's Finder or the Windows Explorer to locate the required file on your hard drive. Again, this is simple to use, but could be tedious where a lot of files are involved. It did, however, display the correct sample rate and bit rate for all the files I submitted.

I had a few problems with the Emasters web site. At first I couldn't create an account; then I had trouble logging in to it; and then I couldn't make the checkout section of the site work. Meanwhile, although the uploader worked fine, it gave no indication of progress, and appeared to lock up my Web browser. I'm not sure what the cause of these difficulties was, but I suspect it might have something to do with the *SOS* firewall, which has been known to kick up a fuss with some sites. Either way, Emasters are easily reached by phone if you do encounter problems.

Once you've uploaded some files, you can then select any or all of them by ticking the adjacent box, before choosing a range of actions from a pop-up menu. The most important of these, obviously, is 'Get eMastering'. Clicking this will bring up an on-line form where you can enter mastering notes for each track. I used this in the same way as before. A further pop-up menu also allows you to order additional materials such as dub plates and vinyl or CD production masters. Emasters, like Mastering World, spotted that 'The Baltic Fleet' was almost in mono and checked whether that was deliberate.

The Tests

As all three studios operating under the Mastering World umbrella offered to take part, I ended up with a total of five professionally mastered files for each song. Since I hadn't originally intended to have 'Ancient History' properly mastered, I had already bodged together a 'home brew' master, and Dave Lockwood and David Glasper were able to give me home-mastered versions of the other tracks too. Our listening tests thus featured a total of six versions for each song, plus the original unmastered mixes for reference. All of our 'home brew' masters were generated before we had heard any of the test tracks, and I thought it would be particularly interesting to compare our own efforts with the professional results.

However, it's important to point out that there are some crucial aspects of mastering that can't be evaluated in a test like this one. The mastering engineer's job is not just about making each track sound as good as it can: it's about making a collection of tracks work together as a coherent whole. Decisions about dynamics, equalisation and so on will normally be taken in the context of creating a consistent sound across an album, where a track that doesn't sound bass-light in isolation, for example, might still need some low-end boost to fit in with the rest. It would have been a bit of a cheek to ask all the

Halfway (Mastering) Houses

Mastering World, Metropolis iMastering and Emasters, the companies that took part in our tests, are as far as I'm aware the only three UK-based mastering studios to offer a specialist on-line business model. With each of them, you can create an account, upload your files, pay for the mastering and have the results delivered without leaving the comfort of your studio. Because they charge on a per-track basis, you also have the reassurance of knowing that you can set a budget for mastering and stick to it.

However, it should be pointed out that the rest of the mastering world is hardly stuck in the Dark Ages, and that most other mastering studios are now on-line at least in the sense of being able to receive files via FTP. This makes it possible to arrange a conventional unattended mastering session, again without leaving your home. If you opt for this approach, the main differences are that you'll be charged on a per-hour basis rather than a per-track one, you won't be able to pay on-line, and you'll probably have to use generic FTP software to send and receive files, rather than the specially developed applications used by the three companies in this test.

participants to master an entire album just for the sake of this article, so we weren't able to test their skill in this regard, but it's something that should be kept in mind.

I decided that the fairest way to evaluate the differences would be in a proper double-blind test, so I created a folder on one of the *SOS* servers and placed each version of each song in it. All were converted to WAV format and given the same date information. I then renamed each file in an arbitrary order, using only letters of the alphabet to distinguish them, and asked one of my colleagues to further rename those files in a different order. That way, none of us would know which version was which until we compared notes as to what naming convention we'd used. This folder is reproduced on the *SOS* DVD (www.soundonsound.com/dvd) exactly as it was in our office, with six versions of each song named 'Ancient History 1', 'Ancient History 2' and so on. The original unmastered files are included in a separate folder: I didn't bother to disguise them as they would have been easy to spot in any case, being much quieter than the mastered versions.

As soon as we started listening to the files, we all found that the variation in apparent loudness completely overshadowed every other difference between them. This was particularly noticeable in 'Melt Like This Snow', where there was almost 5dB difference in average RMS power between the loudest and quietest version. Most of those who took part in the listening tests therefore agreed that it was only possible to compare other issues such as spectral balance, stereo width, distortion and dynamic variation if the versions were first level-matched in some way. This would also allow us to compare all the mastered versions directly with the original unprocessed mixes.

Unfortunately, setting up a fair level match isn't quite as straightforward as it sounds. For example, if your track has a quiet verse leading into a loud chorus, and you have two masters, one more compressed than the other, levels that match in the verse won't do so in the chorus and *vice versa*. As a compromise, some of us took the approach of matching the level during the loudest part of each song, while I decided to calculate the average level of the entire track using *Cubase*'s Statistics function and adjust mixer levels to suit. The files as presented on the DVD are not level-matched, so you can choose your own approach to doing this — but it's definitely worth doing.

In Brief

Mastering World
- **Engineers: Donal Whelan (Hafod Mastering), John Dent & Jason Mitchell (Loud Mastering), Simon Heyworth (Super Audio Mastering)**
- **Prices: Hafod £35-£50 per track; Loud Mastering £65-£150; Super Audio Mastering £100-£130.**

W www.masteringworld.com

Metropolis iMastering
- **Engineers: there are 10, but on-line clients don't get a choice.**
- **Price: £75 per track.**

W www.metropolis-group.co.uk

Emasters
- **Engineers: Kevin Metcalfe, Streaky (The Sound Masters)**
- **Price: £50 per track.**

W www.emasters.co.uk

Nobody Said This Would Be Easy!

In order to give as many *SOS* readers as possible the chance to do the listening tests under the same double-blind conditions, I'm going to wait until next month to reveal which mastering engineers were responsible for which of the test tracks. If you can find the time to do the listening tests, we'd be very interested to hear your views: in the new Mastering section of the *SOS* forum at www.soundonsound.com/forum/postlist.php?&Cat=&Board=Mastering, you'll find a thread where you can post your preferences and your comments. In the meantime, the tests provoked some fairly heated discussion in the *SOS* office, which I've boiled down as follows.

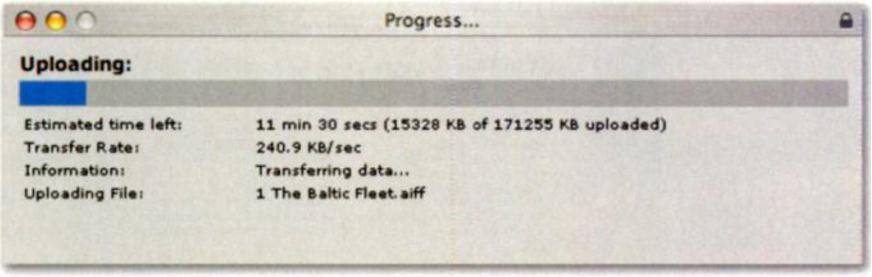
The big surprise, once the variations in loudness had been eliminated, was how small the other differences were in every case — so small that most of us found our preferences weren't even consistent at different monitoring levels. In fact, on first listen, several of the versions for each track were almost indistinguishable; we all found that we needed to audition small sections of the tracks repeatedly to pick up the differences, and I was surprised at how difficult it was to pick out my own 'home brew' master of 'Ancient History' from the professionals' efforts. It would be nice to think that this was because the original mix was so perfect that it couldn't be improved, but somehow I doubt that this is true! In fact, we were all aware of flaws in our mixes — there were some boomy bass notes in 'Melt Like This Snow', while several of us felt the vocal was mixed too low in 'Ancient History' — and we were surprised to find that none of the engineers in the test had tried to do anything about them, or asked us for a 'vocal up' mix of the latter. Nor could we detect much in the way of stereo width enhancement, or any more than the most gentle equalisation.

'Melt Like This Snow'

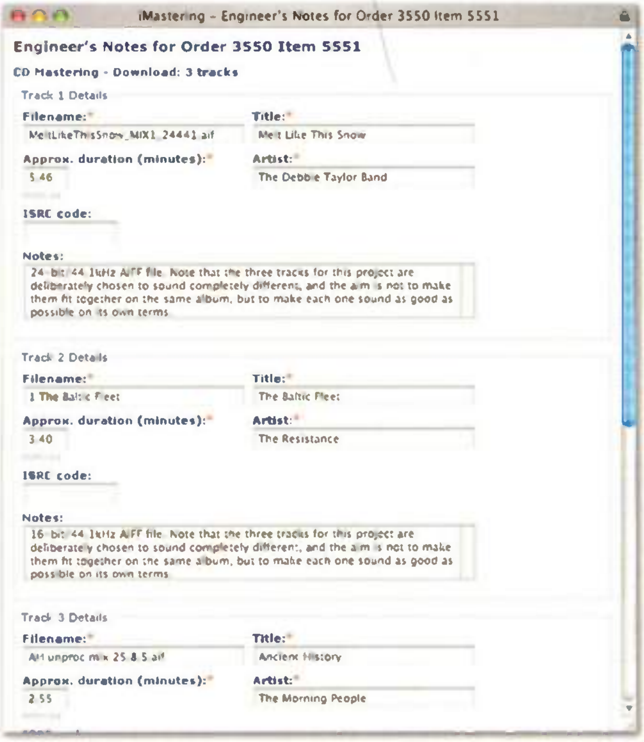
If we were in any doubt that mastering is above all a matter of taste, the first track would have confirmed it. Even the two members of the Debbie Taylor Band taking part had wildly different preferences, and there was precious little agreement among the rest of us. In fact, although everyone commented that three or four of the versions were almost indistinguishable, no-one could agree which three or four versions they were! What was noticeable in all the mastered versions was that dynamics processing had brought up the level of the reverb in the mix. As you'd expect, it had also ironed out some of the dynamic variation, most notably in version 3. Some of the engineers had applied gentle EQ boost in the high-frequency area of the spectrum, with the aim of making it more sparkly or airy — again, this is perhaps most obvious in version 3 — while the bottom end also varied slightly between versions.

Debbie herself, along with Mike Senior and Hugh Robjohns, disliked version 3 vehemently. They felt that both the EQ and the dynamics processing had been overdone, with the resulting track sounding harsh and over-compressed. Dave Lockwood, by contrast, almost picked it as his favourite, for the same combination of loudness enhancement and high-end boost. Eventually he, along with Paul White, settled on the similar but slightly more subtle qualities of version 5, and this scored solidly from everyone. Debbie preferred version 2, which she felt brought the vocals forward slightly; others also liked this, but found the bottom end variously 'woolly' and 'a bit lightweight'.

Mike and I both liked version 6, which seemed to have a slightly richer and deeper



The Mastering World site gives you clear feedback about the progress of your file uploads.



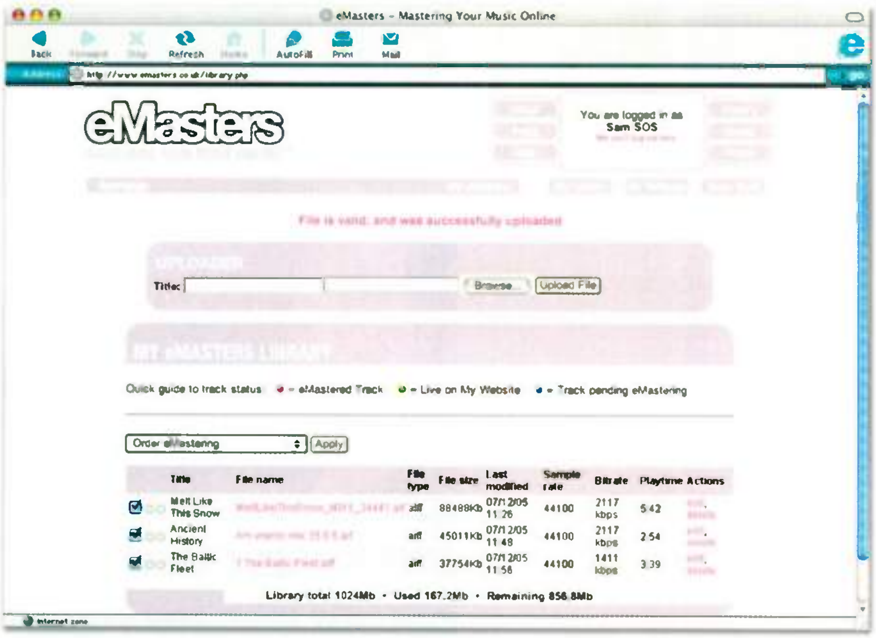
On the Metropolis iMastering site, you fill in a single Engineer's Notes page that covers all your tracks.

bottom end than the others, whilst keeping plenty of detail and sparkle in the high frequencies. Hugh picked version 4, which was also popular with Mike, but which was disliked by both Debbie and Dave. In fact, feeling that some of the high-frequency detail had been lost as a side-effect of limiting, Dave said he would have sent this version back to be redone. The remaining master, version 1, was not hugely different from the original mix: it was the least compressed and thus the quietest of the masters, with a very similar balance of frequencies, and was no-one's favourite or least favourite.

'Ancient History'

There was a difference of almost 3dB in average RMS level between the loudest (number 2) and quietest (number 5) masters of 'Ancient History', and as you'd expect, the louder versions tended to be more noticeably compressed, with the dynamic contrasts in the song somewhat ironed out. (These are perhaps most noticeable at the end of the guitar solo,

The iMastering site uses a Java applet to display the contents of your hard drive and their server in a familiar tree structure.



Like Mastering World, the Emasters site is easy to use, but neither of them makes batch file transfers as convenient as is the case with iMastering.

and in the sections where the guitars come crashing in on the word 'history'.) Dynamics processing reduced the impact of the drums in all the mastered versions, especially during fills like the one at around 1'43", but interestingly, the worst affected was number 4, which was by no means the loudest master.

On some of the versions, the engineers had tried to warm up the bottom end of the track, with the aim of bringing the bass guitar out a little and making the electric guitar sound warmer. Others, meanwhile, seemed to concentrate on adding sparkle in the high-mids instead. I felt the bass lift had gone a bit wrong in version 1, making the track a little dull-sounding, with a slightly thick and woolly low end, though this version was popular with some. Version 2 had a full bass and a nice warm guitar tone, but the mastering had also boosted a subsonic resonance in the kick drum, which Dave felt should have been eliminated. This version also seemed to have some top boost, which helped the energy of the track to survive the heavy limiting, but the combination of EQ and limiting had significantly altered the drum sound, making it splashy and unfocused.

In version 3, the subsonic resonance was gone, but unfortunately, the fundamental of the low bass notes had gone with it. It also seemed to most of us that this version was a little brighter and thinner-sounding than the original, which wasn't an improvement, and this one got mid-table rankings from everyone. Version 4 struck all of us as being tonally neutral, with perhaps just a touch of bass lift, and was the favourite of both Mike and Paul. Despite the weedy drum fills, I agreed that this one did the best job in terms of 'transparent' loudness maximising. In a strange sort of way, though, I was also drawn to version 2, for exactly the opposite reason: while the over-the-top limiting made it most people's least favourite, it was also the most different from the original mix. Having mixed the original track, I think I wanted the mastering process to make obvious changes, and I felt a bit cheated by the versions that didn't seem to do anything!

Versions 5 and 6 divided opinion. Number 5 was Dave's favourite, but although we agreed that the engineer had done a good job of rounding out the bass, several of us felt that the track as a whole sounded a little flat and lacking in excitement, and it was my least favourite along with version 1. Version 6, on the other hand, went in the other direction. With a brash and sparkly quality that retained the energy of the track, it scored highly from Mike and Debbie, but was felt by others to be overly harsh. On inspection, this version turned out to be quite heavily clipped.

'The Baltic Fleet'

Although the mix hadn't been compressed, this track didn't leave an awful lot of dynamic room for manouvre at the mastering stage, and the six masters were much closer together in terms of perceived loudness than was the case with the other two tracks. Nevertheless, it managed to split the listeners at the *SOS* office even more successfully than either of the other two!

Again, the iMastering site offers a helpful progress display while uploading is taking place.

The mastering engineers had clearly taken different attitudes towards what Hugh described as a 'tendency to a wild and uncontrolled bottom end', which is most apparent in the breakdown at around 1'52" onwards: some had tried to control it by applying a high-pass filter to remove subsonic frequencies, whilst others had made it more of a feature. Further up the frequency spectrum, some of the engineers had concentrated on adding weight to the middly bass sound that plays through the rest of the song, while others had boosted the treble to emphasise the glitchy percussion. Some of the engineers had also attempted to obscure the distortion on the first couple of notes, while others had accepted it as part of the track.

Version 2 was the one that completely polarised opinion in the *SOS* office, to the extent that it came either first or last in everyone's list! Most of us felt that it was the loudest, but some thought the treatment had made it over-compressed and lifeless, whilst others found it the most punchy of all. Hugh, Paul and Dave all singled it out as their least favourite, while Mike, David, Debbie and I all liked it the best; in Mike's description, it was simply 'the most engaging to listen to, in some indefinable way'. Those who liked it felt that it did a good job of making the bass weighty and interesting without clogging up the low-mids like some of the others.

There wasn't much more agreement on the other versions, with the exception of number 6, which everyone found indistinguishable from the original. Version 4 showed the most obvious signs of an attempt to control the low bass, and also seemed to have had the most treble boost applied. This divided opinions in the office. Mike and I both felt that it was a shame to lose the subsonic detail, which was interesting and gave the track depth, and found the treble boost unnecessary. However, it was Hugh's favourite, scored highly from Debbie, and was one of only two versions David Glasper picked out as improving on his original mix. Version 3 got good marks from Hugh, but was described by Dave as sounding 'congested' and by David as 'dull and muddy... earning a special distinction as the least appealing'. David disliked versions 1 and 5 on similar grounds, yet these were Paul's favourites, and 5 also scored highly with Mike and Debbie.

When this feature was first suggested, I don't think any of us expected the

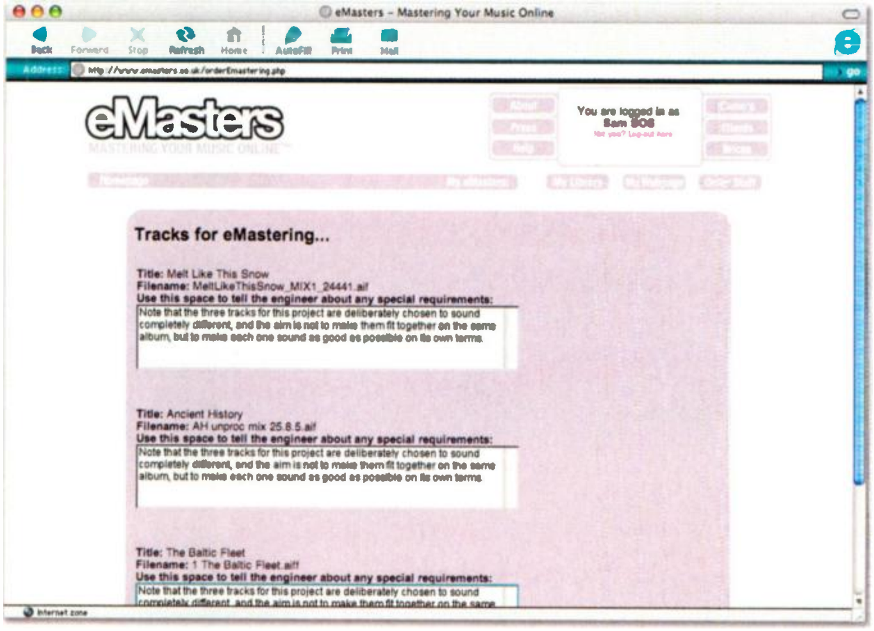
differences between the tracks to be so small, or that there would be such widespread disagreement about which were better or worse. I'll leave it until next month to reveal which masters were created by which engineers, but it probably isn't giving too much away to say that, with opinions differing so wildly among our listeners, there is unlikely to be a single winner that's head and shoulders above the rest.

The On-line Alternative

When I started writing this feature, the question I wanted to answer was 'Is on-line mastering a good alternative to the traditional approach?' Having finished it, the short answer is yes: if I was going to have an album mastered, I would be happy to have it done on-line, and on-line mastering will save you money compared to an attended session. However, there are a couple of fairly big 'buts', the first of which runs something like this. Communication with the mastering engineer, both before and during the session, is absolutely vital, and the best way to achieve that communication is to be in the studio with them. On-line mastering will always be a compromise in this respect, and of the three on-line services I tried, the ones I liked the best were the ones that did most to make this communication possible.

For that reason, my least favourite was Metropolis's iMastering, where you don't even get to know the name of the engineer who masters your tracks. I've no doubt that all of their engineers are highly competent and experienced, and Metropolis has an enviable reputation as a mastering house, but when you're organising something as personal as mastering, I think it's very important to deal with an individual rather than a brand.

Both of the other two services allow you to choose who will master your tracks, and make it possible to deal with them directly, which seems to me the key. Of the two, I slightly favoured Mastering World, partly because I found their web site easier to use, but mainly because they seemed genuinely keen to encourage as much communication as possible between engineer and client. There are times when email just isn't adequate to explain something, and Mastering World's engineers all seemed happy to pick up the phone, even for an on-line mastering job.



Once again, notes for the engineer are entered in one go on the Emasters site.

What's Mastering For?

The second 'but' is that writing this feature has changed my expectations about what mastering can and can't do. Perhaps naively, I expected that professional mastering engineers would be able to sprinkle some sort of sonic stardust onto our tracks that would be impossible to achieve with the project-studio gear we all used to record and mix them, but this simply didn't happen. With the exception of loudness changes, the differences between all the mastered versions were minimal, and with at least one of the three tracks, we found it impossible to pick out the 'home brew' master with any confidence. In the cases where mastering improved the tracks, the improvement was slight, and in some cases we felt that the process had made the tracks worse. And where we felt there were faults in the mixes, the engineers had been either unable or unwilling to deal with them.

What this brings home is that mastering is not a magic bullet. If you hire a mastering engineer because you think they can make your music sound radically better, or cover up flaws in your original mix, you're going to be disappointed. Mastering engineers can make a disparate collection of tracks into a coherent album, find an acceptable balance between loudness and sound quality, cut an acetate where the needle won't jump out of the groove, or create an error-free digital master to send to a pressing plant. They can provide a fresh perspective on your music, and even point out areas where you might have gone wrong: but ultimately, it's the artist, the producer and the recording engineer who have to take responsibility for the way their recording sounds.

However, that's not to say that the sonic differences between our test tracks are unimportant. When choosing an engineer, you want someone who is at least sympathetic to your tastes, even if no-one else agrees! Next month, therefore, we'll be revealing which engineers were responsible for which versions. Were the studios with the best Web sites also the ones that did the best masters? Did individual listeners tend to prefer the same engineers on all the tracks, or did they shift allegiances? How well did our 'home-brew' masters score? All will be revealed in our April issue. SOS

Is Louder Better?

The more I listened to the different versions with their levels matched, the more it reinforced the idea that dynamics processing at the mastering stage is a matter of compromise between loudness and sound quality. Level-matched, for example, I found I liked the unmastered mix of 'Ancient History' better than any of the mastered versions; to my ears, any mastering improvements such as a warmer and clearer bass were always outweighed by the loss of dynamics and the lack of impact in the drums. In an ideal world, we'd be able to release tracks in this state and rely on listeners to turn their hi-fi up!

In the real world, however, no-one level-matches tracks on their hi-fi or radio, and relative loudness is something that can't be ignored. What surprised me was how much of a difference loudness made, and how consistently our ears were drawn to the loudest masters when we listened without level-matching. Sure, the attraction of the louder masters is superficial, and other versions would certainly be preferable for long-term listening: but it's a competitive world out there, and that extra half a dB really does make a difference to the casual listener. Having been through these listening tests, I find myself more willing to compromise sound quality for loudness than I used to be.

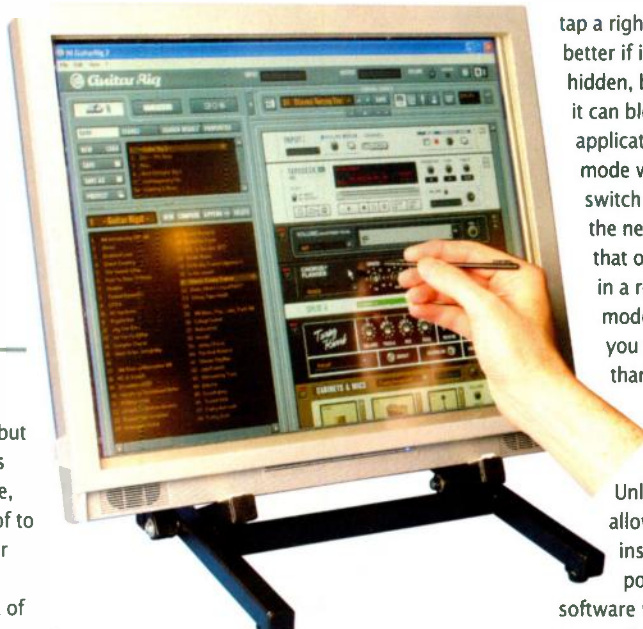

MultiMix16 FireWire
ALESIS
16 channel
12 channel
8 channel

Intatouch Touchscreen

Touchscreen Add-on For Mac & PC

Hate mixing by mouse? Intatouch's neat monitor add-on allows you to draw control data directly onto the screen.

Sam Inglis



Everyone loves to gripe about mice, but although touchscreens and graphics tablets have been around for a while, Intatouch are the first company I know of to market one specially for musicians. Their design is a simple pressure-sensitive, transparent panel that fits over the front of your flat-screen monitor and connects to your Mac or PC via USB. A small stylus allows you to position and click the on-screen pointer, meaning that — in theory — you can do everything that's normally done with the mouse by 'drawing' on the screen. It starts at just £159 for a 15-inch version, and most people will be able to use it with their existing flat-screen monitor.

Invisible Touch

You can attach the touchscreen to any securely mounted flat-screen monitor — it just hangs over the top — but positioning it so it's comfortable to use is more of a problem. Obviously, it needs to be within easy reach, and in many ways, the ideal plane would be closer to the horizontal than to the vertical. Also, you really need to be looking straight at the screen: because there's a gap between the touch screen and the LCD beneath, the two appear slightly out of alignment if viewed at an angle, making it hard to work out exactly where to draw. Intatouch supplied the review unit with a guitar amp stand, as seen above, which is a cheap and reasonably effective answer to the problem.

You can use the touchscreen with a single display, but dedicating a second LCD to it would allow you to have your primary screen mounted in a conventional position, with the touch screen horizontal for easier drawing. A second screen could be run at a relatively low resolution, so you wouldn't have to draw so precisely; this would also compensate for the loss of visual sharpness caused by the extra layer of 'glass'.

Once you've got the monitor working, a simple calibration utility asks you to touch the four corners of the screen in turn, and you're ready to go. A blue and white box floats above your applications at the bottom right of the screen. By default, tapping the stylus on the screen generates a left-click, but touching this box changes things around, making every tap a right-click. This is fine, but it would be better if it was more flexible. The box can be hidden, but not moved around the screen, so it can blot out bits of your music applications. Also, I would have liked a mode whereby touching the box would switch over to right-click behaviour only for the next click, not permanently — it's not that often I want to do lots of right-clicks in a row. There's an alternative operating mode where a click is registered when you take the stylus off the screen, rather than when you hit the screen, but I didn't find this awfully useful.

Quick On The Draw

Unlike a mouse, the touchscreen allows you to move the pointer instantaneously between any two points on the screen. It's fantastic for software where you need to 'draw' things like automation curves, EQ responses, waveforms, envelopes, patch cords and the like, and it is great for manipulating faders and drawbars. On the other hand, I found it even more awkward than the mouse with rotary knobs, while dragging and dropping can be frustrating — you have to maintain a continuous pressure on the screen in order to avoid accidentally dropping things. The mouse will still work in conjunction with the touch screen, so you could always go back to it for these tasks.

Unlike Jazz Mutant's Lemur, this touchscreen is 'monophonic' — you can't use multiple styluses to manipulate several controls at once. However, it's also unlike the Lemur in not costing £1500-plus, and not needing complex *Max/MSP* or *Reaktor* programming to be useful. In general, this touchscreen is a very attractive alternative to the mouse, especially if you use graphics software too, and an affordable one. It also has obvious benefits for those suffering RSI or similar. The only real difficulty I can see is finding the best way to mount it, and that's far from an insoluble problem. SOS

SOUND ON SOUND

Intatouch Touchscreen
£159-£199

pros
- Affordable.
- Better than the mouse for many functions.
- Should work with your existing monitor.

cons
- Might be hard to find the best position for the monitor in a studio environment.
- Dragging and dropping is fiddly.
- Floating on-screen window can be a nuisance.
- Some loss of image sharpness.

summary
Intatouch's touchscreen is a neat and inexpensive alternative to the mouse.

information

£ 15-inch version £159; 17-inch version £179; 19-inch version £199. Prices include VAT.
W www.intatouch.com
E sales@intatouch.com

Apogee Rosetta 200

A-D/D-A Converter

Apogee's new converter builds on the reputation of the original PSX100 and Rosetta units, upgrading the sonics and adding a pile of new facilities, including sophisticated jitter-reduction processing and sample-rate conversion.

Hugh Robjohns

The Rosetta 200 is the newest two-channel converter in Apogee's impressive range, and it replaces both the highly regarded PSX100 and its simpler sibling, the original Rosetta converter, both of which I reviewed back in *SOS* November 1999. The converter and clock technology employed in the Rosetta 200 is essentially the same as that in the Rosetta 800 (reviewed in *SOS* March 2004), with full support for sampling rates up to 192kHz (including SMux facilities for the ADAT port) and comprehensive I/O connectivity. There is also a slot for an optional interface module, with cards available to couple directly with Pro Tools (either new HD or older Mix systems) and for a Firewire interface for native DAWs.

The PSX100 was a superb unit, and the Rosetta 200 is designed to build on that, offering excellent signal routing and format-conversion functions, but perhaps the real icing on the cake is the new Coda 'Audio Finishing Module'. This provides (for the first time in an Apogee product) high-quality sample-rate conversion, along with an automatic level-setting facility for the A-D converter called the Aptomizer. The familiar UV22HR dithering algorithm is also included, as is Apogee's Soft Limiter — an fast-response analogue input limiter.

Connectivity & Routing

The rear panel of the Rosetta 200 carries balanced line-level analogue inputs and outputs on XLRs, adjustable for peak levels anywhere between +2dBu and +26dBu. AES3 inputs and outputs are also on XLRs, dual sockets being provided to accommodate double-wire interfaces for high sample rates.

Phono (RCA) sockets are used for coaxial S/PDIF in and out, along with a pair of Toslink optical sockets (configurable for ADAT or S/PDIF formats). Word-clock input and output are on BNCs (with selectable 75Ω input termination), and there is also a pair of MIDI sockets (in and out), a removable plate over the interface card slot, and an IEC mains inlet (accepting anything from 90-250V AC). The MIDI sockets are provided for software updates, but if the Firewire card is installed then MIDI data can also be passed via these sockets to the host computer.

The Rosetta is configured almost entirely through various front-panel button combinations, and this makes it very easy to change settings, as well as allowing a complete reset to default factory conditions if required. The only configuration option set 'mechanically', by an internal jumper link, is the power-on mode, with a choice between permanent powering, or power switchable from the front-panel button).

Apogee have always been praised for their stable, low-jitter clocks, and the latest Intelliclock design employs a dual-stage jitter-reduction technique that uses a FIFO buffer to isolate external, unstable clocks

SOUND ON SOUND

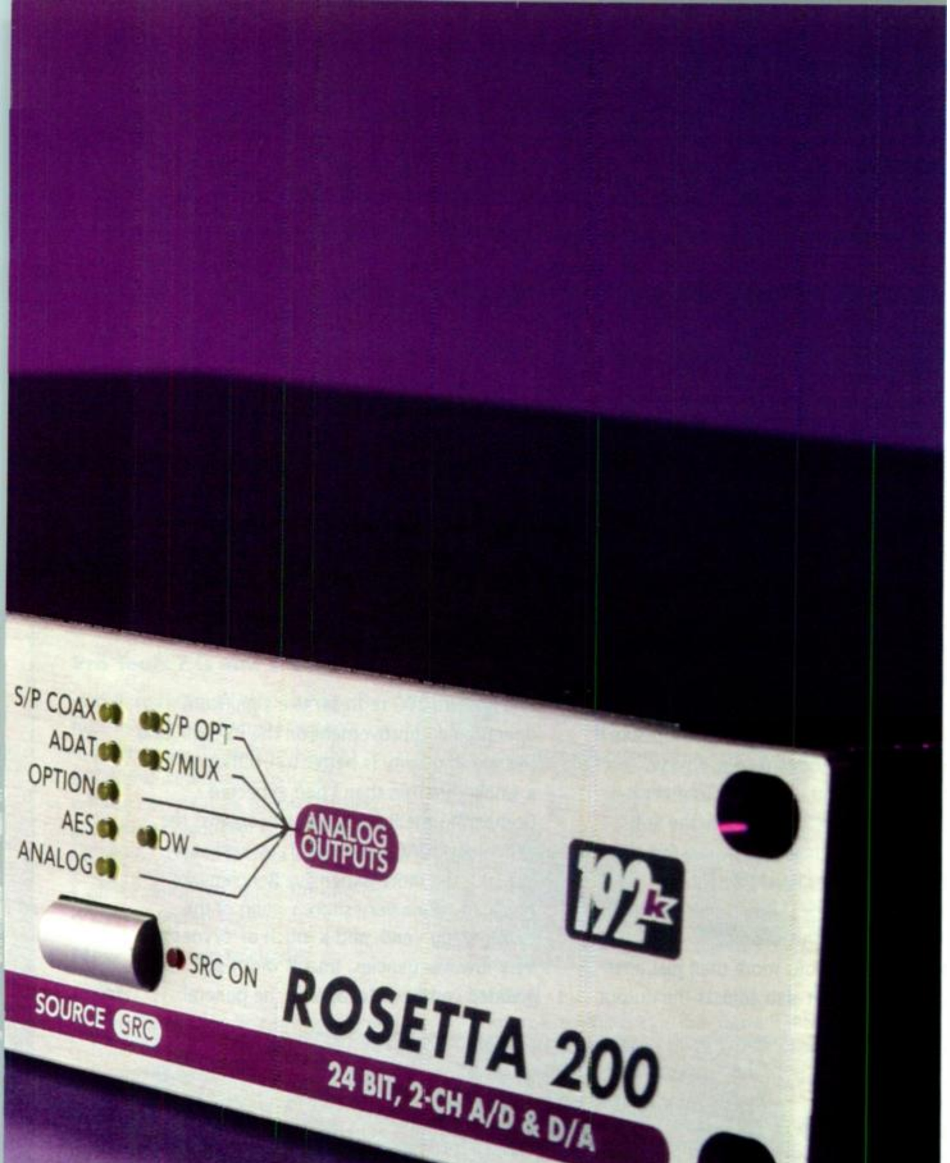
Apogee Rosetta 200 £1639

pros
- Very good converter quality.
- Superb I/O flexibility and routing.
- Sample-rate conversion facilities.
- UV22HR dithering.
- Aptomizer automatic level-setting function.
- Front-panel configuration.
- Full output metering.

cons
- Aptomizer leaves negligible headroom.
- Not possible to restrict Aptomizer to A-D only.
- Automatic SRC function not yet available.

summary

A superbly flexible stereo converter, combining excellent A-D and D-A performance with comprehensive I/O routing, format conversion, sample-rate conversion, and 16-bit UV22HR dithering options.



Photos: Mike Cameron

from the internal reference clock — a FIFO buffer (first-in, first-out) is a form of short-term memory. Incoming audio samples are loaded into the FIFO buffer using a 'write' clock derived from the incoming digital signal, and a high level of jitter and instability can be tolerated here, since stable timing isn't required. The audio data is then retrieved from the FIFO buffer using a very precise 'read' clock, which permits jitter-free D-A conversion. Of course, the FIFO buffer introduces a short storage delay, but this only amounts to a few samples, and is irrelevant compared to the normal conversion-filter delays. A similar approach is used to ensure that the A-D conversion is jitter free, even when using an external reference clock.

All the input sources are effectively connected using a buss arrangement, feeding two independent source selectors controlled by front-panel buttons. One selector feeds all of the digital outputs together, while the other feeds the D-A converter, and hence the analogue outputs. Two front-panel buttons determine the clocking. There are six standard sample rates derived from the internal crystal between 44.1kHz and 192kHz, plus an external input. If the external-input mode is selected, then the next button selects which source to use: S/PDIF, ADAT (or SMux), AES, word clock, or the option-card input. A pair of LEDs arranged to resemble an exclamation mark indicate the lock status: both lights mean a stable tight lock has been achieved, whereas just the upper LED means that the reference is unstable.

Dual-function Controls

Most of the front-panel buttons have a second function, activated by holding the button down. For example, the sample-rate selection button also determines the AES format (single or dual wire) when a dual- or quad-speed sample rate is selected, and also the optical output format (S/PDIF or ADAT). The external-clock selector button also sets the word-clock I/O ratio, which allows the unit to operate at a different sample rate to the external reference, or to generate a different clock output rate from the internal rate. The digital output source button also engages a sample-rate converter, allowing variable or asynchronous sources to be connected.

The sample-rate converter is a very ▶

Choosing The Right Reverb

Recording engineers have been adding artificial reverberation to recordings for many years, and have developed a variety of different ways of doing this. Home recordists now have access to modern versions of all these tools, so let's look at how they compare to each other, and how best to use each.

Paul White

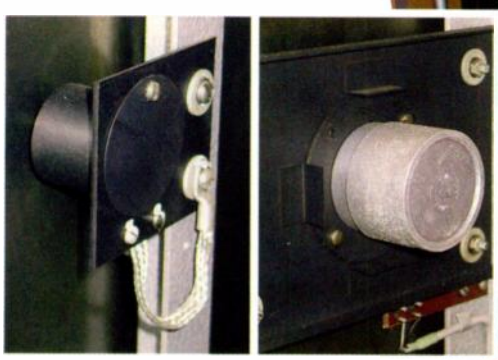
Artificial reverb is an integral part of music production, as it puts back the sense of space and place that's removed by close-miking voices and instruments in an acoustically dead studio. In the real world, reverb is created by sounds reflecting and re-reflecting from surfaces in an enclosed or partially enclosed space, and the resulting pattern of sound is infinitely complex. The geometry of the space and the materials from which it is made affect both the pattern and intensity of the reflections, and the rate at which different frequencies decay. Our brains derive information from these audio characteristics, enabling us to learn something about the nature of the space without necessarily seeing it. In music production, this means that the reverb type and its settings need to be chosen carefully if the human hearing system is to accept it as natural — or at least believable.

In the early days of recording, there was no artificial reverb, so the effect was created by placing microphones and loudspeakers in a reflective room such as a tiled basement. Rockfield Studio in Wales even had a room made with suspended glass plates to create a variable reverberant environment. The different rooms built by different studios were often instrumental in how popular the studio turned out to be, but most of these rooms became obsolete when artificial reverb was invented. Probably the first successful artificial reverb device was the spring, though all kinds of weird and wonderful devices were created, including coiled pipes with a speaker at one end and a mic at the other.

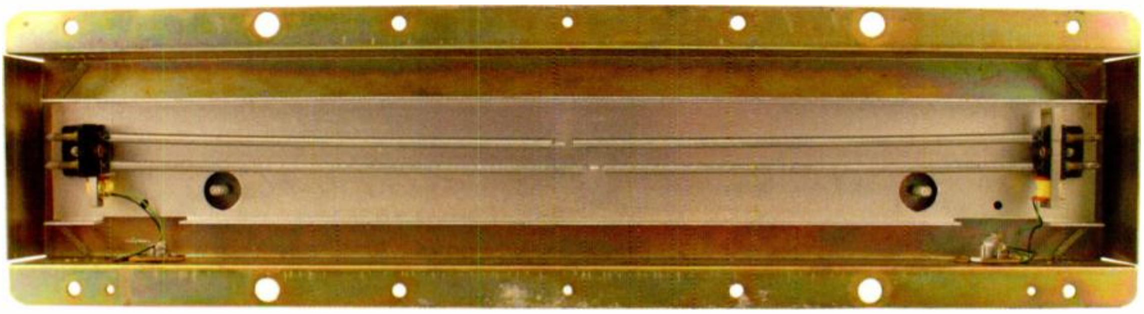
Analogue Reverb: Springs, Plates & Delays

Apparently the first spring reverb was developed by accident after a couple of sound engineers experimented by dangling a coiled spring from a gramophone pickup cartridge, shaking it to create thunder effects for radio. By fitting a transducer to the other end of the spring, they found that sounds passed along the spring were picked up with a kind of reverb added, and so the spring reverb was born. Springs are still used in guitar amplifiers, and they have such a distinctive sound that modern digital reverb units often include spring emulations.

While the spring worked well enough on vocals, guitars, and electronic organs, it wasn't so good on percussive sounds, as it had a tendency to 'twang' — the sound fluttered and pulsed as the reflections bounced back and forth along the length of the spring. You can get an idea of what spring reverb sounded like on vocals by



The reverb plate was the first really serious artificial reverberation technology, and was adopted by numerous studios before the advent of affordable digital reverb. The main functional component is a large metal plate within a soundproof box (above right), which is vibrated by an output driver much like that in a loudspeaker (above centre). The vibrations are then picked up by a separate transducer (above left) for each output channel. The sound of this type of reverb became so popular that it is still emulated by digital reverb units and plug-ins such as Universal Audio's *Plate 140* (right).



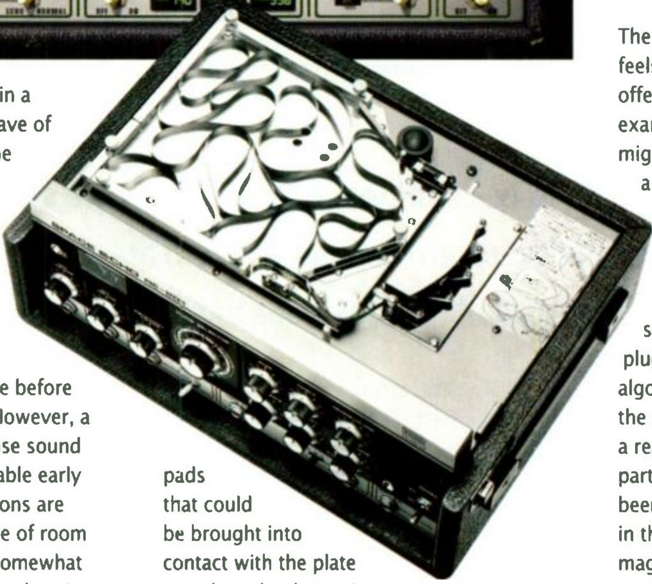
Before the reverb plate came on the scene, the most common artificial reverb option was based around steel springs, like these ones from Accutronics which can be found within some Fender guitar amps.

trying one of those toy microphones with a spring and a diaphragm inside. (In fact some people have used these for recording in various ways!) All kinds of tricks were tried to improve the quality of spring reverb — immersing the spring in oil, using multiple springs with different characteristics, using heavy limiting on the input signal and extensive EQ — but it wasn't until the reverb plate was developed that artificial reverb really got serious.

As its name suggests, the plate reverb utilised a thin metal plate, which could be as large as one metre square or more, suspended in a soundproof box and driven into vibration by means of a transducer similar to the voice-coil assembly of a loudspeaker. This arrangement was fed from a mixer aux send, via a suitable amplifier to drive the plate. Pickups mounted on the surface of the plate picked up the vibrations and fed them to a preamplifier, and the signal was then fed back to the mixer's aux returns. Normally two pickups, mounted at different positions on the plate's surface, were used to give a pseudo-stereo output from a mono input.

"Because we tend to be conditioned by the music we grew up with, synthetic digital reverb and plate reverb are still the first choices for most pop work, but fashions are always changing."

The way the plate works is that sound energy from the input transducer moves very quickly from the transducer near the centre of the plate to the edges of the plate, where it bounces back. These reflections in turn encounter other plate edges and continue to re-reflect, losing energy but gaining complexity. Because the plate is rectangular ▶



Before digital delays became widespread, a common way of implementing delay was using tape loops. One of the most popular tape delays was the Roland RE201 Space Echo, which held a long tape loop in a bin just beneath its lid, as shown in the black and white picture below. The unit also included spring-based reverb which could be combined with the outputs of the machine's various tape heads. Tape delay remains popular, but is most likely to be encountered in digitally modelled form in plug-ins such as Universal Audio's new RE201 recreation (left).

and the sound waves propagate in a circular manner, even the first wave of reflections is very complex, as the expanding wavefront will encounter different parts of the edge of the plate at different times.

The early part of real-life reverberation in larger spaces comprises a number of discrete reflections that are clearly audible before the dense reverb tail builds up. However, a reverb plate produces a very dense sound very quickly, and has no discernable early reflections. Because early reflections are fundamental in conveying a sense of room type to the listener, the plate is somewhat devoid of spatial character insomuch as it doesn't suggest any particular type of acoustic space, but in a musical context this can be an advantage, as we often use reverb as a musical effect, and not to create the illusion of a specific type of room or hall.

While the plate has no real spatial character, it certainly has a recognisable tonal character and can tend to sound rather metallic, so EQ often needs to be used to improve the end result. The decay time of a plate is also fixed, unless some form of mechanical damping is applied, so the better plates were fitted with remote-control fabric pads that could be brought into contact with the plate to reduce the decay time. Undamped, the reverb time could be as long as five seconds or more, which is too long for most routine applications.

Early spring or plate reverbs were also often used in conjunction with tape delay to help create a sense of space and size. As springs and plates generate no early reflections in the true sense of the word, adding one strong psychoacoustic cue using delay helps create a false sense of spatial identity. In a real room, this would equate to how long the sound takes to reach the first surface and then bounce back to the listener. The longer the pre-delay, the larger the space feels. Modern digital reverbs do the same by offering adjustable pre-delay times. For example, a typical vocal reverb treatment might comprise a plate reverb emulation with a decay time of 1.2 seconds or so and a pre-delay of 60-80ms.

While plates are no longer commonplace, there are some extremely good digital simulations of the plate, such as the Universal Audio *Plate 140* plug-in for the UAD1 card, and special algorithms need to be created to reproduce the rapid build-up of density that occurs in a real plate. These plate effects work particularly well on vocals, and would have been used on virtually all the classic records in the '60s and '70s. Multi-head-tape or magnetic-disc echo devices were also often used to create pseudo reverb, sometimes in conjunction with spring reverbs, but used alone these devices were unable to create the necessary density of repeats to emulate the real thing. Instead, the echo machine became an effect in its own right.

Digital Reverb

Digital reverb became a commercial reality with the EMT 250 in 1976, and early digital reverbs attempted to approximate what goes on in a real room by first using a multitap digital delay line to recreate those tightly spaced early reflections. The delay times

Two classic early digital reverb units which are still in use today: the Klark Teknik DN780 (above) and the Quantec Room Simulator (below).

would often be adjusted empirically to produce a pleasing result, rather than being a direct emulation of any particular space, and the amplitude of each tap was also adjusted to produce a natural result. Just 20 or 30 delay taps might be enough to approximate the early reflections, and by choosing different patterns and wider or narrower spacings the impression of various sizes of room or hall could be created.

As a rule, a wider early-reflection spacing is interpreted by the brain as a larger space. In order to reproduce the build-up in complexity in the decaying reverb tail, multiple re-circulating filters (usually a mixture of comb filters and all-pass filters) were used, sometimes fed from the original input, sometimes from the outputs of the tapped delay line, depending on the designer. In either case, these re-circulating filters had to be carefully tuned so that the reverb didn't ring excessively at specific frequencies, and they also included EQ-like elements to damp the high end in a way that replicated the behaviour of a real space.

Everyone had their own method of designing reverb algorithms, which is why the products from different manufacturers had, and still have, very different characters. Some of the most sought-after reverb devices didn't sound particularly natural, but happened to be musically flattering. While digital synthetic reverbs of this type rarely sound exactly like the rooms they purport to emulate, their sound has become part of popular musical culture. The most famous of the digital reverb manufacturers is probably Lexicon, who have defined the reverb sound of pop music over the past two decades, though there were numerous other early digital reverb devices such as Quantec's Room Simulator, the Ursa Major Space Station, the classic AMS RMX16, and Klark Teknik's DN780.

Another successful early digital reverb unit was the Ursa Major SST282 Space Station. Although the original hardware unit (top) is still in use in some studios, most home-studio owners are most likely to come into contact with the Space Station's distinctive sound via the smaller Seven Woods Audio SST206 Space Station (centre) or Princeton Digital's TDM plug-in recreation (bottom).

Digital reverbs are often brighter and more sparkly than anything you come across in real life, but they work perfectly in a musical context. In fact, they work so well that, when sampling or convolution reverb made it possible to 'copy' reverb from actual spaces, many people found that it didn't sound exciting enough for pop music, so turned the process to sampling pieces of reverb hardware. Perhaps the most important aspect of a good synthetic reverb is the way that it seems to become part of the original sound, rather than seeming to be an effect layered on top — which of course in reality it is! With the better units, adding more reverb increases the sense of space, but doesn't swamp the original sound and doesn't make your mix sound congested. Cheaper reverbs may sound OK in isolation, but can end up making your mix sound messy and cluttered when you try to use them on a real project. This has little to do with technical specifications — it is all down to how the reverb algorithms are designed.

Convolution

Back in the '80s when I first asked manufacturers whether it would ever be possible to sample the reverb character of a real space, the response I always got was that there would never be enough computing power available. However, computing power has continued to follow Moore's Law and today we have computers that would have been inconceivable back then. In the mid-'70s I was working for a company that hot-rodded Commodore PET computers by expanding the stock 1K memory to 8K, but now 8GB of memory is a practical option. In other words, in only 30 years or so the amount of memory you can fit into a typical desktop computer has expanded by around one million times, and clock speeds are now so high you could cook frozen chicken with them! The Atari ST, on which so many of us started, had a clock speed of just 8MHz, whereas some of today's machines run at over 3GHz.

The most famous of the digital reverb manufacturers is probably Lexicon, who defined the pop reverb sound of the day with their 224 model.

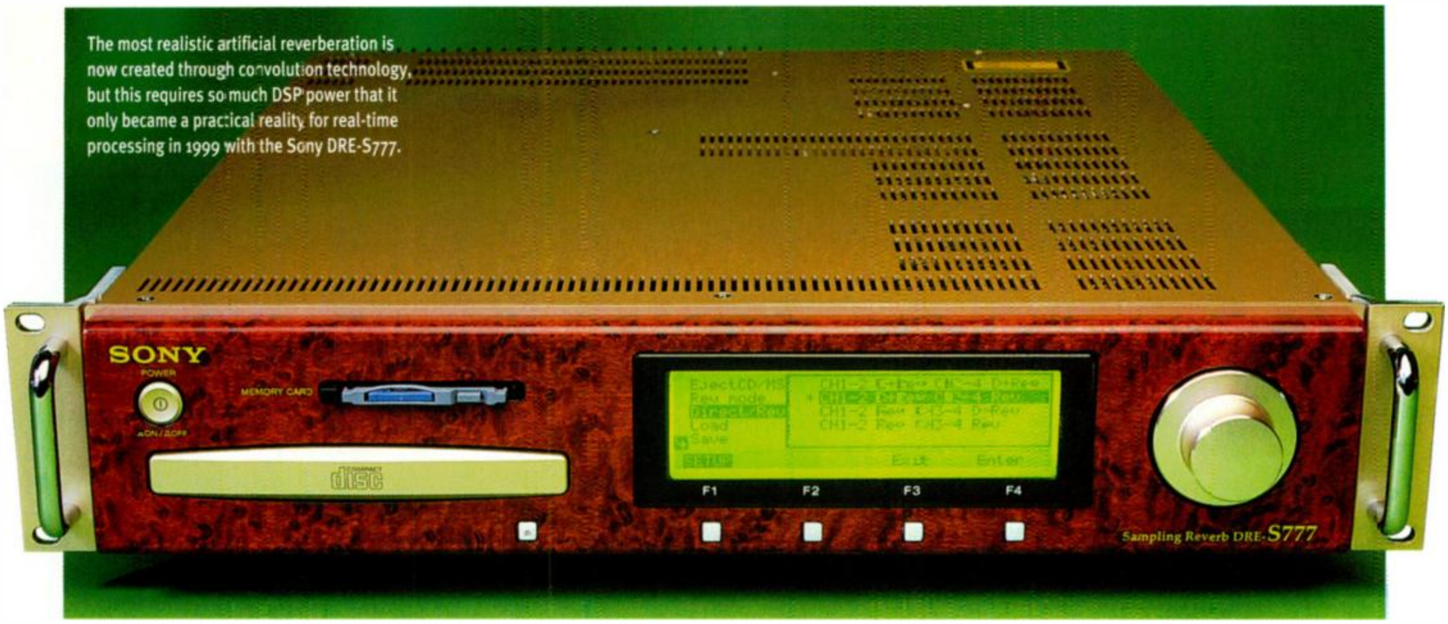
Now, of course, convolution has become a reality, and can run on just about any well-specified Mac or PC, as well as in dedicated hardware effects units. Convolution is essentially a brute-force, number-crunching means of producing reverb (or other linear delay-based treatments) which is brilliant in its simplicity. It starts with the premise that you can measure the reverb character of a room by putting up a stereo mic at the listening position, then recording the reverb created from a single spike of sound one sample in length — a starting pistol or a balloon burst gets close to this. The resulting sound picked up by each microphone is called an 'impulse response' and it can be recorded conventionally. If you then use a computer to multiply this impulse response by every sample of new audio signal, the net result is that you add the original room's reverb to that audio signal.

Nowadays, instead of trying to create a loud sound one sample in length (which isn't very easy), impulse responses tend to be recorded using a sine wave swept over the whole audio frequency range over a period of a few seconds. A mathematical process is then used to time-compress the resulting data so that it equates to what you would have got had you used a single-sample pulse. The advantage is that it's easier to reproduce a suitable signal level for measurement, and because the measurement is integrated over a few seconds it tends to be less susceptible to corruption by low levels of background noise in the room being sampled. Integrating multiple impulse-response recordings from the same source can further reduce the noise floor. Of course, to get a good measurement you need very accurate microphones and speakers, and to do a comprehensive job you may need to take different measurements at different places in the room so as to give the user a choice of reverb characteristics. Fortunately, commercial convolution reverbs come with a wide range of presets based on specific venues, and many third-party impulse responses are available on the Internet, some for sale and some free.

To capture the sonic signature of an electronic reverb unit, the same test signal can be fed through it and then mathematically processed to produce an impulse response, enabling that particular hardware setting to be recreated with almost perfect precision within a convolution reverb system. The limitations of convolution are that the amount of editing you can do to the reverb character is restricted, and also that the process is unable to capture time-varying effects such as chorus modulation. Some hardware reverbs use modulation within their algorithms, and so are impossible to 'sample' exactly using convolution, though the general characteristics of tape delay (other than wow and flutter of course) and most hardware reverb patches can be captured with great accuracy.

Which Type To Use

The choice of which reverb to use is invariably an artistic one, though convolution-based concert halls are a good choice for classical music, while churches and cathedrals will obviously suit choral or organ music. I've also found



The most realistic artificial reverberation is now created through convolution technology, but this requires so much DSP power that it only became a practical reality for real-time processing in 1999 with the Sony DRE-S777.

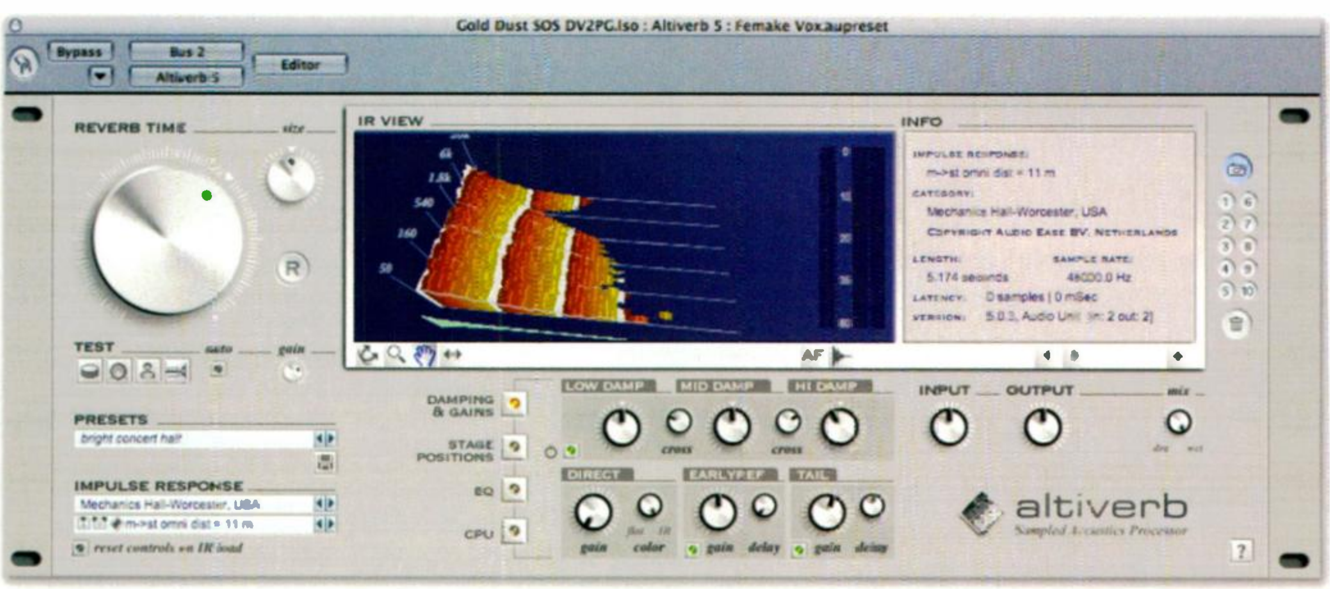
▶ that convolution reverbs can work well on folk or jazz music, where you expect to hear the music performed in a specific type of environment such as a club, but sometimes one of the library impulse responses will surprise you and work in a way that you would never have expected. For example, Apple *Logic Pro 7*'s *Space Designer* convolution reverb plug-in has some impulse responses in its library made in dense woodland, and some of these sound wonderful on acoustic guitar and vocals. Impulse responses taken in bright reflective spaces or top-notch studio drum rooms often work well for drums and percussion, but whether 'real' or synthetic treatments are best ultimately comes down to artistic choice.

Because we tend to be conditioned by the music we grew up with, synthetic digital reverb and plate reverb (either real, modelled, or convolution sampled) are still the first choices for most pop work, but fashions are always changing in music, so this situation could change. When it comes to pop vocals, I almost always end up going back to a synthetic reverb, either algorithmically generated or 'sampled' from a hardware unit that works this way. Having said that, the UAD1 *Plate 140* has such a gloriously authentic sound that I often find myself using that on vocals too. The original plates may have been low-tech, but the good ones sounded fantastic and have a very different character to most algorithmic reverbs. You get all the sparkle and density needed to create a very flattering vocal, but without the sound emulating any obvious room or hall type.

An interesting development is the system used by Wizoo's *W2* plug-in, where you can set up a convolution reverb, then get their engine to emulate it as closely as possible using algorithmically synthesized reverb. This allows more flexible editing and takes less processing power. Another approach taken by some manufacturers to minimise processing load is to use convolution for the first part of the reverb, and then to synthesize the more dense tail of the reverb. As most of the character of a room is conveyed by the early reflections, this hybrid approach can work very well.

Vocals & Drums

We've never had so much choice when it comes to high-quality reverberation, so the challenge is twofold; not only do we have to



The leaps in computer processing power during the last few years mean that reverb convolution processing has become increasingly available in the form of plug-ins such as Audio Ease *Altiverb*.

decide which type of reverb to use, but we also have to avoid getting sidetracked into trying out every one of the hundreds of presets and room samples that modern products so often come with! In my view, synthetic reverbs (or convolution samples of synthetic reverbs), with their larger-than-life sound, are still the first choice for pop vocals, though the choice of decay time is harder to pin down, as it can vary from around two seconds for some ballads to less than one second where you're after a tight, intimate sound. Furthermore, on those reverb devices that let you adjust the level of the early reflections and reverb tail separately, you have the option of using stronger early reflections to create density and space with less reverb tail, which helps avoid clutter.

As a rule, modern productions tend to use far less obvious reverb treatments than records made up to the early 1990s, so it really helps to put on some records and do some analytical listening to see what reverbs have been chosen and how heavily they've been applied. It's probably fair to say that choosing an appropriate vocal reverb is more important than any other reverb application. One factor always to keep in mind when mixing is that a heavy application of reverb tends to make a sound appear to be more distant, and that's often at odds with the concept of a lead vocal that you want to keep upfront. Shorter, brighter reverbs feel closer, while warmer, longer reverbs push sounds to the back. If you need more obvious reverb on a lead vocal, then keeping it bright and adding some pre-delay is one way to prevent the sound being pushed too far back.

Today's drum sounds are drier and tighter than the roomy sounds of the '70s and '80s, so convolution presets based on studio drum rooms, or synthetic ambience programs that focus on early reflections, tend to be very effective in producing the desired sound and sense of space without the result getting messy. For a more traditional rock drum sound, though, a synthesized reverb plate still sounds fantastic. Guitars and synths can make use of whatever suits artistically, though springs and plates are still very popular. The only word of warning here is that many synth patches come with lots of reverb to make them sound impressive for in-store demos, but in a mix you may find you need to back off the reverb a bit.

If you're into dance music, you may notice that very heavy reverb is used on some synth parts, but it is still used with care, not indiscriminately. It's also often the case that less is more, so using reverb sparingly can result in a product that's artistically better than where reverb is overused. Ultimately, the correct choice of reverb type and its application is an art which rates alongside setting level and pan settings as a major part of mixing. Knowing what is available helps, but you also need to develop your listening skills so that you can hear how the better commercial records were produced and how the use of reverb figures in those productions. SOS

Thanks to FX Rentals (+44 (0)20 8746 2121) and Klark Teknik (+44 (0)1562 741515) for supplying the Klark Teknik DN780, Quantec Room Simulator, Lexicon 224, and Ursa Major SST282 Space Station digital reverbs pictured in this article. These units can be rented from FX Rentals for around £35, £88, £35, and £59 respectively per day including VAT. Thanks also to Jim Kenniff at K-Sound Studios (+1 717 872 8509) and to Janet and Phil at Bova Sound (+1 613 228 0449) for the pictures of their EMT 140 plate reverbs.

Millennium Dual-core Pentium 4 PC Music Computer

Martin Walker

Dual-core processors hold the key to unprecedented performance with music applications. But how do systems based around Intel's Pentium D, like this one from Millennium Music, compare with their AMD-based rivals?

In *SOS* January 2006 I reviewed a dual-core PC for the first time. Scan Computers' 3XS system featured an AMD Athlon X2 4800+ with both cores running at 2.4GHz. Now it's time to check out the opposition: Millennium Music's system is based around an Intel Pentium D 840 processor, which has both cores running at a much faster 3.2GHz — although relative clock speed is no longer a reliable guide to relative performance. This is also the first system I've been sent that features PCI Express slots, which have suffered a mixed reception amongst people using PCs for music as a result of various associated problems. However, as I reported in PC Notes February 2006, the Intel 955 chip set in this Millennium PC has been pronounced free from buss-hogging, so the system should in theory achieve impressive test results.

Case For The Defence

Millennium have used a rather unorthodox case design for this computer. Antec's P180 sites the PSU — in this case a Tagan 480W model fitted with 'Silence Control Technology' and 120mm low-noise fan — in a separate chamber at the bottom, which prevents temperature-controller supplies getting noisier by having to deal with system as well as PSU heat. The case also features a magnetically sealed front 'door' to minimise drive noise levels (although, oddly, this hides the drive activity LED!), two handy front-mounted USB 2.0 ports and one Firewire port. The side panels feature three-layer aluminium/plastic/aluminium construction to dampen system noise, while Millennium have also added additional acoustic foam lining and drive-bay 'blanks' to further lower noise levels.

The Akasa Evo 120 CPU cooling system, with its large fan, is becoming a popular choice in PCs where low noise is important.

Both Scan and Millennium abandon the stock AMD/Intel CPU coolers in favour of quieter alternatives, and in fact Millennium too have chosen the Akasa Evo 120, incorporating heatpipe, heatsink, and low-noise 120mm fan, which fits a variety of AMD and Intel processors. This is obviously a popular item that performs more effectively with dual-core PCs than the previously favoured Zalman Flower cooler, and the use of 120mm fans is a growing trend because their larger blades and slower rotation speeds result in quieter operation. Millennium partly rely on the CPU cooler to push warm air from the rear of this PC as it's already almost exactly in line with the existing fan grille, but the main exit point for warm air in the P180 is via a top-mounted 120mm case fan, which is supplied with a clip-on grille/duct to prevent airflow accidentally being blocked if you place anything on top of the case.

When idling, the CPU temperature remained at about 40 degrees Centigrade, and after running Prime 95 for several hours it rose only as far as a comfortable 52 degrees Centigrade, while remaining relatively quiet. My only reservation is that although the top of the case is the perfect theoretical position to exhaust the rising hot air, I'd have personally preferred to have the top 120mm case fan mounted on the rear panel, since it's already higher here than in any case with a top-mounted PSU, and doing this and blocking off the top opening would result in even lower noise.

Express Delivery

PCI Express-equipped motherboards fall into two camps. Some recognise that the x16 graphics cards are about the only PCI Express devices currently being sold, and therefore provide five old-style PCI slots and only one PCI Express one, while others opt for a more balanced approach. This system

SOUND ON SOUND

Millennium Dual-core Pentium 4 PC **£1811**

pros

- Significantly more power than an equivalently clocked Pentium 4 Prescott PC.
- Deluxe acoustically treated case that's both quiet and stylish.
- Fast 10,000rpm system hard drive.
- Exceptionally fast 360GB RAID 0 audio setup for those requiring vast track counts or sample/video streaming.

cons

- Disappointing CPU performance relative to similarly priced AMD dual-core systems.

summary

This system from Millennium is quiet, capable, and the fastest Intel-based system I've reviewed to date. However, it's ultimately let down by the performance of Intel's Pentium D 840 processor, which now lags significantly behind similarly priced models from AMD's dual-core Athlon X2 range.

is built around an Asus P5WD2 Premium motherboard that takes the latter course — there's a x16 PCI Express slot for a graphics card, a clever 'universal' x16 format slot that can run in x2 or x4 modes, plus one general-purpose x1 PCI Express slot (which is disabled when the universal slot is used in its x4 mode) and three traditional PCI slots. This approach seems perfect for musicians who want to move forward into the PCI Express arena, but already have PCI soundcards or DSP cards that they don't want to abandon.

The case also provides another spare dummy backplate position for those inevitable extra sockets, which Millennium had used for the Akasa fan speed control. They had also fitted a budget ATI x300 graphics card with passive cooling in the main x16 slot, and an ESI Waveterminal 192x card in one of the PCI slots, leaving plenty of room for further expansion. Since the main advantages of PCI Express are for graphics cards at present, it might seem odd to partner the x16 PCI Express graphics slot with a comparatively slow graphics card, but this reduces the likelihood of buss-hogging problems, and musicians have no need for the fastest 3D graphics. Also installed on the motherboard were the previously mentioned Intel D 840 processor, plus two 1GB sticks of DDR2 800 RAM.

Four hard drives were installed and NTFS formatted: a 74GB Western Digital WD74 Raptor model with 10,000rpm spin speed (one of the fastest models available), and a RAID 0 setup comprising three Western Digital SATA II drives with a total storage of 360GB, divided into a small 'outer' 20GB Scratch partition for your current project, with the remainder devoted to Audio.

Completing the drive complement were a Samsung 16x DVD-RW burner and floppy drive, and for this particular system Millennium also supplied an ESI Waveterminal 192X audio interface, a Relisys 17-inch TFT monitor, and Steinberg *Cubase SX* 3.1 software. Providing the finishing touches were useful Millennium help files, Nero Burning ROM, Power DVD XP, and Norton's Ghost 9, to allow complete backup of both system and data drives.

Performance

The 10,000rpm system drive measured 58MB/second sustained write and 56MB/second read speeds on my standard Dskbench test, which is surprisingly similar to measurements for the more typical 7200rpm Seagate Barracuda drives used on so many music PCs. However, HD Tach proved the advantage of the 10,000rpm spindle speed, with outer to inner speeds of 72MB/second to 52MB/second (compared with about 60/30 for Barracudas). This will mean slightly faster boot-up and application loading times, although the Raptor does add about £100 to the system price.

The three RAID 0-formatted drives measured a staggering 172MB/second write and 180MB/second read speeds, and a projected track count of over 700 tracks of 16-bit/44.1kHz audio, or over 200 tracks at 24-bit/96kHz. This will benefit those who want to achieve vast audio track counts, to do lots of 192kHz recording, run an entire orchestra using multiple streamed sample libraries, or have video streaming alongside their audio tracks. Those who don't fall into these categories will probably find a single 7200rpm audio drive more than sufficient, as well as being cheaper and slightly quieter, and the more standard option of a 7200rpm 80GB system plus a 250GB audio drive would shave about £200 off the total system price. Your data is also slightly less secure spread across a triple-drive RAID 0 array, since you could lose it all should any one of the those drives ever go faulty. Millennium told me that they were interested in pushing this review system to the limits of drive performance, but that anyone buying one of their systems can choose to have the same three drives formatted as RAID 5, for 240GB of storage with significantly more security but rather less speed.

SiSosftware's Sandra measured a healthy 5180MB/second memory bandwidth, while the CPU Arithmetic and Multimedia results were almost exactly in line with the reference figures for a Pentium D 840 model. However, as always, real-world tests with audio applications are the only reliable way to establish how a music PC performs doing what it's designed to do. With Timo's *Cubase SX* Performance Test 2.0, as used on all my previous PC system tests, I got CPU overheads in Play mode of 29 percent with a conservative 23ms latency at 44.1kHz, and 34 percent when the latency had dropped to 3ms. Referring back to my previous measurements, this dual-core Pentium D 840 system running at 3.2GHz is therefore a good 35 percent faster than the single-core 3.2GHz P4 Prescott system from *SOS* January 2005, and slightly faster than the rather more expensive dual-Xeon 3.06GHz system I reviewed back in *SOS* June 2004.

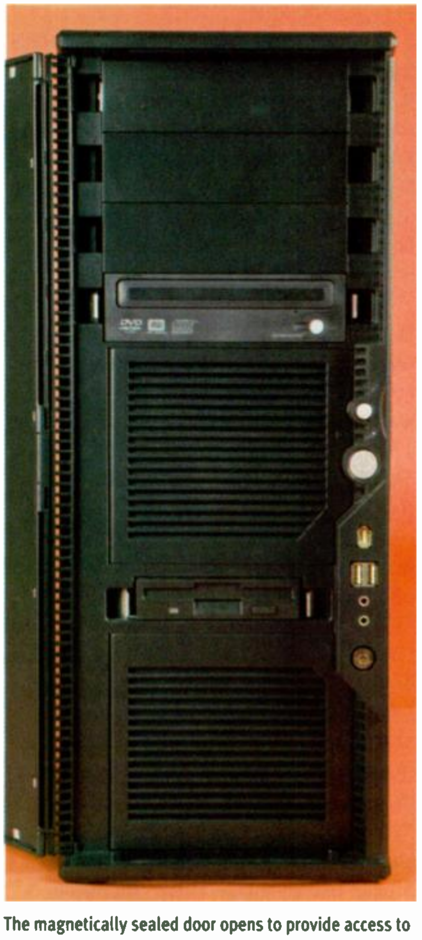
At this point I'd have been very impressed with this dual-core Pentium D system, if I hadn't recently reviewed a dual-Athlon XP X2 system. Admittedly, the Scan 3XS Athlon system had a far more expensive 4800+ processor, but you can nearly always scale processor performance in the same family from relative clock speeds, and by doing this I can extrapolate that this Pentium D 840 system turns in a *Cubase SX* performance that's of the same order as an AMD Athlon 64 X2 3800+ model.

I considered this rather disappointing, so repeated all my tests, and even removed the Echo Mia card from my own PC and installed it in the Millennium system to rule out soundcard-specific issues, but still got identical results. I then ran the *Nuendo* Thonex benchmark, which produced CPU usages of 59 percent with a 1024-sample buffer, 61 percent at 512 samples, 64 percent at 256 and 72 percent at 128 (3ms latency at 44.1kHz). Correlating these with other published tests of PC music systems using a variety of other processors, the

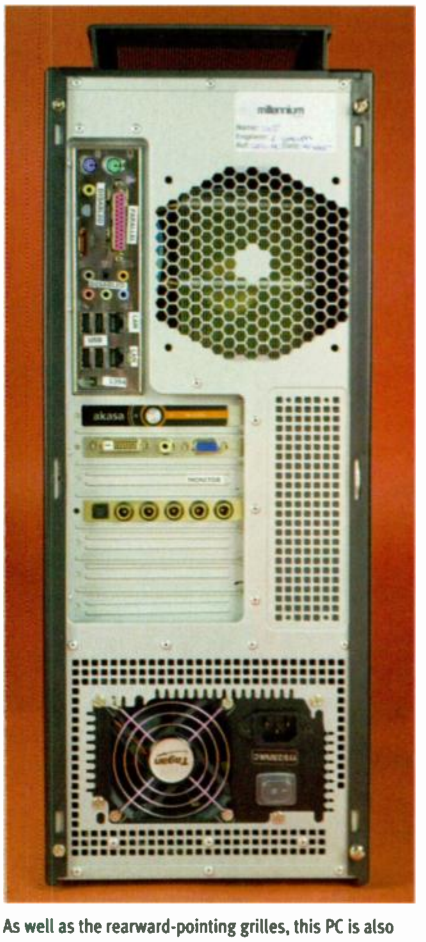
Specifications

- **Case:** Silver/black Antec P180 Advanced Super Mid Tower.
- **PSU:** Tagan TG480-U01 480W fitted with 'Silence Control Technology' and 120mm low-noise fan.
- **Motherboard:** Asus P5WD2 Premium with one LGA775 socket for Intel Pentium 4, Intel 955X chip set running at 800MHz front side buss, four 240-pin DIMM sockets supporting up to 8GB of system memory, native DDR2-800 support, three PCI slots, one PCI Express x1 slot, one Universal PCI Express x16 slot, and one PCI Express x 16 slot for graphics card.
- **Processor:** Intel Pentium D 840 dual-core, with each core running at 3.2GHz and having a 1MB cache.
- **CPU heatsink and fan:** Akasa Evo 120 incorporating heatpipe, heatsink, and 120mm fan with 15dBA noise rating.
- **System RAM:** two 1GB sticks of DDR2 800.
- **System drive:** Western Digital Raptor, 74GB, 10,000rpm, SATA II, 8MB buffer.
- **Audio drives:** three Western Digital WD1200JS, 120GB, 7200rpm, SATA II, 8MB buffer.
- **Graphics card:** ATI x300 PCI Express, passive cooling and 128MB RAM.
- **Optical drive:** Samsung Writemaster TS-H552U burner.
- **Active system ports:** PS/2 mouse and keyboard, six USB 2.0 ports, two Firewire ports, parallel port, two Gigabit LAN ports.
- **Keyboard & mouse:** Logitech cordless Internet Desktop black.
- **Installed operating system:** Windows XP Professional Edition with Service Pack 2.

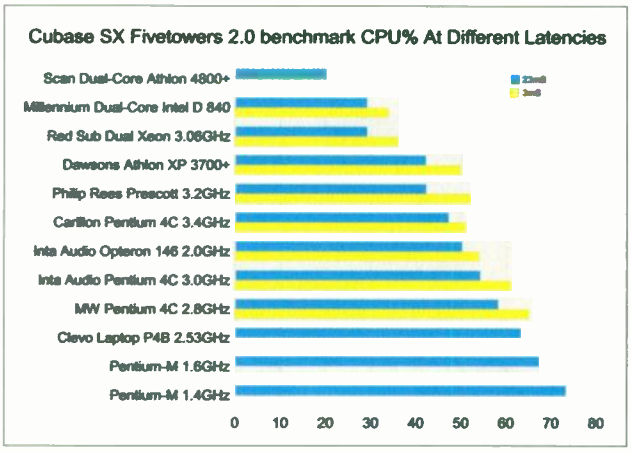
- **For this particular system:**
- **Monitor:** Relisys TL766, 17-inch diagonal TFT, silver, 1280 x 1024 native resolution.
- **Audio interface:** ESI Waveterminal 192X with version 6.0 drivers.
- **Music software:** Steinberg *Cubase SX* 3.1.



The magnetically sealed door opens to provide access to the drives and front-panel ports.



As well as the rearward-pointing grilles, this PC is also ventilated via the top of the case.



This Millennium dual-core Intel Pentium D system has similar processing power to a 3.06GHz dual-Xeon PC, and 35 percent more than the single-core Pentium 4 Prescott model clocked at the same speed, but is nevertheless eclipsed by Scan's dual-core Athlon 4800+ system at around the same overall price.

figures do seem to add up correctly for an Intel D 840, and other tests such as Sandra's CPU Multimedia also place the 840's performance somewhere between the Athlon X2 3800+ and 4000+ models.

Summing Up

Millennium Music's Intel dual-core system provides significantly more processing power than any previous Intel-based system I've reviewed, and it's quieter than the Scan AMD dual-core system reviewed in *SOS* January 2005 (although that could easily be further silenced). It also offers audio hard drive performance way beyond any other system I've tested to date. So if you're interested in a dual-core system, should you opt for AMD or Intel? Well, this is a complex question that involves quite a few decisions involving the benefits (or not) of PCI Express and whether or not you have audio DSP cards, so I've covered it in depth in this month's PC Musician feature. However, I've now reviewed one AMD and one Intel dual-core system, and it's certainly possible to compare these two.

The Millennium and Scan computers exemplify two different design approaches. This Millennium system starts with a more expensive quiet case, fitted with yet more silencing features, which in turns houses a modern PCI Express-equipped motherboard that costs £100 more than the 'tried and trusted' PCI/AGP model chosen by Scan. Both feature 2GB RAM, but once again Millennium have used more expensive materials, going for DDR2 PC5300 RAM rather than the cheaper DDR2 PC3200 used in the Scan system. Both have four drives (one system drive plus three in an audio RAID array), and the Millennium system turns in a stunning audio drive performance (albeit at the expense of slightly reduced data security), although the Scan system does offers more storage capacity. What the Scan system saved on case and motherboard was spent instead on a more powerful CPU — at current prices the Intel D 840 CPU fitted in this Millennium PC would cost about £360, compared with £570 for the AMD Athlon 64 X2 4800+ in Scan's system.

For those lured by the future promise of PCI Express, this Millennium system is quieter, more sophisticated, and uses more recent components, and it's only fair to point out that an AMD dual-core PC offering PCI Express features would be rather more expensive than the Scan system. However, if you're not yet convinced of the benefits of PCI Express, and given that the majority of musicians seem to run out of processing power before anything else, I have to say that I'd be seriously tempted by the 30 percent faster audio processing of Scan's AMD system.

Overall, Millennium have once again turned in a quiet and luxurious system with great attention to detail, but I have to admit that I'm disappointed by the performance of Intel's Pentium D range, which is lagging considerably behind that of equivalently priced AMD Athlon dual-core models. However, as always, this is just one snapshot system from Millennium's current range, and they tell me that by the time you read this they will have Athlon dual-core systems in their range. I'd be inclined to opt for one of these instead — currently, AMD seem to be the leaders of the pack. SOS

information

£ Basic system as reviewed without monitor, keyboard/mouse, music hardware or software £1811 including VAT.
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Yann Tiersen

Recording *Les Retrouvailles*

Franck Ernould

The unique musical world of Yann Tiersen was opened up to millions of people by the soundtrack to *Amelie*. His idiosyncratic working method involves toy instruments, songs built from fragmented ideas, and entire string sections created by recording himself over and over again.

"I'm not a cerebral guy," says Yann Tiersen. "When I record a song, I like to capture what's going on straight away. I don't make demos: I find it's impossible to redo something with the same energy, the same feel you had when you were inventing it. I try to retain these unique moments in the final version. A song often loses its soul when you try to refine it, to redo it well. I keep the first draft, and I only develop one version of a given song at a time.

"A song can come out of a four-minute-long guitar track where I'm just repeating the same chords; I find a bass line, then the other parts come, and the song is born. But in the final mix, that first guitar part will still be there, even if it's barely audible. Starting from a little idea like that, you have almost total freedom to build the song. The more dumb and basic the idea is, the more wild and original the song can become."

At the age of 35, Yann Tiersen is one of the most famous French recording artists. His latest album, *Les Retrouvailles* ('Reunion'), has already sold several hundred thousand copies throughout Europe. What makes this surprising is that Tiersen's style is rather marginal in today's musical world, containing no electronics, no samples, no machines and no MIDI. He plays most of the instruments himself, recording them live with almost no editing or plug-ins. Tiersen plays mainly instrumental and short songs, employs delicate and rare musical instruments, adds evocative and almost childish noises (bicycle, toy piano, music box, recorder, even a typewriter) and combines them with 'normal' rock & roll instruments. However, Tiersen is no Luddite: he writes and records his music in his own studio, uses modern production tools like *Logic Audio* and Pro Tools, and mixed the latest album on an SSL 9080K at Davout Studios in Paris, while its release was accompanied by a 'making of' DVD in 5.1 surround.

A Career On The Edge

As a teenager in the '80s, the classically trained Yann Tiersen felt more attracted by rock & roll, and put down his instruments to play electric guitar. His first album, *La Valse Des Monstres* ('Monsters' Waltz', 1995), was a compilation of themes written for theatre and animated movies. In it, Tiersen's musical style arrived fully formed. He recorded it himself with an ADAT, a few budget microphones and an analogue console. A year later, on *Rue Des Cascades* ('Cascades' Street'), Tiersen progressed: his string arrangements, which he played himself, were richer, the melodies more developed, while follow-up *Le Phare* ('Lighthouse') reached a wider audience and saw Tiersen singing for the first time, along with guests such as Claire Pichet and Dominique A, and touring France. His 2000 album *L'Absente* ('Missing') brought in new sounds like the Ondes Martenot (see box), a string quartet, an orchestral ensemble, well known English singers such as Lisa Germano and Neil Hannon, and a new sound engineer, Fabrice Laureau.

While working on this ambitious project, Yann got a phone call from French film director Jean-Pierre Jeunet, then shooting *Le Fabuleux Destin d'Amélie Poulain*. Jeunet loved Tiersen's records and had already put some songs on his rushes as temp tracks. He asked Tiersen to write some original themes for the film. "I met him, saw some sequences edited with my music, and I was amazed — it worked pretty well!" says Tiersen. "I agreed to write some more songs, even though I only had 10 days to do so." Four new songs were on the movie soundtrack, among them the famous 'Valse d'Amélie' ('Amélie's Waltz'). All the other songs came from Tiersen's previous albums, mainly the second one: a sort of 'best of Tiersen according to Jeunet'. Tiersen's music played a huge role in *Amélie*, and went a long way to create its touching and very personal atmosphere.

The film, and accompanying album, became an incredible success, with Tiersen's newfound fame allowing him to fill big halls with enthusiastic audiences... who kept on asking him to play little waltzes on the accordion! He recorded a double live album at the Cité de la Musique de Paris, with the Synaxis ensemble, and in 2003, composed his first original soundtrack, for German movie *Goodbye Lenin*. *Les Retrouvailles*, his first solo album for four years, was released in France in May 2005.

The Island

Like *Le Phare*, *Les Retrouvailles* was recorded on the island of Quessant off the coast of Brittany. Tiersen brought some instruments with him, and a mobile home studio: his condenser microphones (now Neumanns!), his Powerbook running *Logic*, a Digidesign M Box interface and a pair of loudspeakers. His girlfriend Aurélie du Boys came with him, to shoot the DVD that shows the making of the album.

"During several months, I recorded some basic elements, when they came, on guitar, piano, violin... Then Fabrice arrived here, and I presented him all these bits and pieces."

Fabrice Laureau takes up the story: "When I arrived in Quessant in June, Yann had recorded around 30 songs with two or three instruments. He knows how to record and set up the gain structure, but does not always place the microphones at the right distance. As I listened to these ideas, I gave him my opinion: I had neutral and fresh ears. I cut some parts and we added some instruments on some songs, with a close and a distant mic. Sometimes, I EQ'ed or I compressed directly. It was the first time I had ever used the Avalon VT737, and I loved it. Yann had brought his mandolin, a Pleyel piano, his violins, and several electric guitars with a little good-sounding Fender Champ amplifier."

Yann Tiersen: "An orchestra is an ensemble of people, and you can't ask 40 people to be completely involved in a track. That's why sometimes, I prefer to record myself on 40 tracks!"

At this stage, all these songs were only fragments, but the finished CD includes a track entirely recorded at Quessant, '7pm'. "I like to include a solo violin piece on every album," explains Tiersen. "Even if it's rough, I like that. But this time, I had nothing. I tried a waltz idea, and even developed a few versions of it — I never usually work like that — but nothing came. Then Fabrice and I had the idea of recording in a little church on the island. It had a very distinctive natural acoustic, with a deep, bass reverb. It kind of inspired me."

Fabrice explains the song's title: "We arranged with the vicar to have the church from six to 10 pm. We arrived, we installed the gear, Yann warmed up, and as soon as

Photo Richard Dumas

I hit Record, the bells started ringing seven o'clock. We had never thought of that! Yann kept on playing and I kept on recording. He was focused on what he was playing, the moment and the ambience inspired him, and his take was superb: he never got it better afterwards. We listened back and it was perfect. A first take, and the bells bring a lot to the song's atmosphere."

Yann's World

Yann, Fabrice and Aurélie then returned to Paris, where Yann has a studio in the basement of his house, and worked solidly for four months. "We worked song by song. I went downstairs, I played, I looked for ideas on this or this instrument, Fabrice recorded them and we listened to the results. If it was good, we kept on working in that direction; if not, we changed our perspective. We can both be stubborn sometimes, but we often agree. That's why we work together!"

"In his studio, Yann has everything at hand: his keyboards, his amplifiers, and all the childish music instruments he has collected — from a toy piano to the 'Romanian duck'!" explains Fabrice. "It's his world. We sometimes had around 100 tracks playing in *Logic*. You have to take care and to use good gear to avoid hiss." The number of tracks is needed because Yann often records all the elements that make up a string arrangement himself, one at a time. On his first albums, he was forced to do this for budgetary reasons, and it became part of his style. Most of the string sections we can hear on *Les Retrouvailles* were recorded this way, and

The Ondes Martenot

Featured on the *Les Retrouvailles* album and much more on the DVD *La Traversée* and on stage, the Ondes Martenot was one of the first electronic music instruments. It was invented by Frenchman Maurice Martenot in 1928, and inspired many classical composers from that era, including Darius Milhaud, Olivier Messiaen (whose sister-in-law, Jeanne Loriod, was a virtuoso on the instrument), André Jolivet and Arthur Honegger. The Ondes Martenot appears on some famous songs too, by Jacques Brel for example ('Les Vieux' and 'Le Plat Pays'), and in the music for films from *Mad Max 3* to *Mars Attacks*, where its mystery is perfect for highlighting dramatic tension.

The instrument has a miniature piano keyboard which is vibrato-sensitive, but it also allows the user to control the notes by sliding a ring along a metal ribbon. By changing the built-in speaker, the player can modifiy the sound signature to make it metallic or resonant in the same way that a guitar player changes his sound by using a certain type of amplifier, and a 'timbre drawer' makes it possible to change the sound's character in the same way you would edit a synth preset. Yann Tiersen has used the Ondes Martenot extensively on stage since 2001, but he lets the French virtuoso Christine Ott play it. "I always loved the instrument after I discovered in Messiaen's music," says Tiersen.

they have a distinctive feeling. "When I recorded *Goodbye Lenin*, I discovered that when you have parts played by a real orchestra, there is inertia. An orchestra is an ensemble of people, and you can't ask 40 people to be completely involved in a track. That's why sometimes, I prefer to record myself on 40 tracks! I'm not at all disappointed with orchestras, but when I do it myself, the results often suit me better."

The Joy Of Accidents

In Aurélie's DVD, entitled *La Traversée*, several sequences show the songs building progressively, but there's no foolproof method at work. "When Yann begins to stack tracks, he reacts to proposals, remarks, ideas," says Fabrice. "Sometimes it's 'Yes', sometimes it's 'No', and sometimes it's 'What about adding this or this?' It doesn't work every time: when we listen again to the track the following day, we realise we were wrong. But he's got an incredible ear, he knows what instrument to add, and all these stacked sounds never become confused or blurred, but enlighten each other. There's no routine involved, no after-the-melodica-let's-play-piano! From time to time, I'll modify the parameters on his guitar amp for more crunch or less fuzz, or I'll put a microphone further away, or if he plays bass I'll add mid and dampen the attack. Yann excels in playing all those little notes that can seem insignificant, but which build the atmosphere of the whole song."

"Except on 'La Boulange', I played the drums on the record myself," explains Yann Tiersen, "and I'm a really bad drummer. I can't make the snare or the kick sound right, so on one song I had the idea of putting my banjo on the snare and hitting my 'cello to mimic the kick. I got a pleasing and personal result, and my drums have the right sound, because I made the sound with what I could do. That's because I can't play: I didn't get what I wanted, so I got other ideas, and accidents helped me to go forward. The song is much better with my 'drums' than it would have been with a real drummer.

"Limits can help you to go further. I'd even say it's terrible to know how to play everything on a musical instrument. I play piano and violin well, but to be able to rediscover them I need to not play them for a while. These days, I have rediscovered guitars. The most important thing is to keep fresh. When I pick up an instrument, if I have already heard what comes to me 10 times before, I give up and choose another one.

"I can't play the musical saw properly, for example, but I use it nevertheless — I find it's a very evocative sound. It's the same with the trumpet: you can see me try to play one in the DVD, and I had even recorded a song with a trumpet on *L'Absente*, but I didn't keep it. I love to improvise when I work on a song. Accidents will happen, but I need that in order to go forward, and sometimes, the song comes from an accident.

"It's the same on stage. I love playing with [*guitarist*] Marc Sense because nothing's ever fixed with him. He wants to use a drill on a

Yann's Studio

The live area in Yann Tiersen's new studio, with his array of classical and not-so-classical instruments.

Photos: Franck Ernould

Asked if he bought any new gear to record his latest album, Yann Tiersen answers "Nothing, except this studio!" Until 2003, he lived in a three-room flat in Paris, with a room for his sound gear, the harpsichord and some other instruments in the living room, the microphones in his bedroom... Yann then found an interesting building a few hundred metres away: he bought it and made extensive alterations, even digging under the foundations, to create his personal studio. Around 4 x 7 metres in size and 5m high, it's acoustically treated and houses his entire collection of musical instruments: harpsichord, spinet, Rhodes piano, mandolin, violins, guitars and amplifiers, lots of percussion and toy instruments. "I record downstairs, on *Logic* or *Pro Tools*, with my Powerbook and my M Box. I'm no expert in software or plug-ins. That's more Fabrice's domain. When he comes here, Fabrice connects his RME interfaces to the control room's dedicated computer and Mackie D8B digital console. I once mixed a lot using the Mackie, but nowadays, I just use its monitoring section, its converters and its digital clock. It's a lot easier for me to mix in *Logic*." Yann uses four Avalon VT737, Joemeek and TL Audio preamps, plus Teletronix LA2A and Urei 1176 compressors. He still uses the Oktava condenser microphones he employed on his first album, although he has since bought an AKG C414 and some Neumanns.

song? Fine with me, no problem, I love that. As long as it doesn't become systematic, using unusual instruments helps me to go forward."

Reunion

After four months of hard work at home, Yann Tiersen and Fabrice Laureau booked a month at Davout Studios, a top independent studio in Paris which celebrated its 40th birthday last year. Yann wanted to add a conventional string section, a wind section, some piano parts, and record his guest singers. Fabrice enjoyed using the studio's Neumann U47s, tube and FET, and brought with him some Coles 4038 ribbon microphones: "I love them. They have the precision we enjoy on condenser microphones, without agressiveness on transients. I almost don't compress them, and the results I get are splendid."

Every day when he arrived, Fabrice transferred the *Logic* files into *Pro Tools* before mixing. "I use *Pro Tools* mainly as a tape recorder. I don't like to edit, I prefer to record things again! I hate to quantise drums or other sounds: doing that to Yann's music kills it instantly. Without time correction, his music lives, charms and touches people — groove comes from these 'imprecisions'! I hate using noise gates too, but I'm very demanding in terms of flat notes or instrument tuning."

The mix is mostly fairly intimate in its sound. "I hate endless reverbs," says Fabrice. "I'm more of the Albini school, I like using the sound of the room. Yann's recording area has a pretty high ceiling; there's some ambience if I record distant mics, but to add air, I fed the recorded sounds into Davout's huge Studio A via loudspeakers, and I recorded the ambience. On drums and on orchestras, it's a real luxury. On the album, I mainly used natural studio ambience, plus an EMT plate reverb. I must admit I discovered superb presets on Davout's Lexicon 960L. My secret weapon is the Eventide H3000. Most people know it as an harmoniser, but it houses a lot of interesting small room and ambient reverbs, very short, which allow you to 'unstick' sounds or push them further into the mix."

Yann's guest singers were Jane Birkin, Dominique A, Christophe Miossec, Liz Frazer and Stuart A Staples of the Tindersticks. They got a day each. "I invited Liz Fraser to sing on two songs," says Tiersen. "I had written the lyrics for 'Mary', but on 'Kala', I left her to improvise freely. It was incredible: she took her notebook and wrote her own words, in her own language; then she sang, and it was superb. I think there's no one who can be as creative, as inventive, as sensitive. She arrived in the morning, we knew she had to leave at 6pm, and by 2pm, we had recorded nothing! She had written some words on her notebook, but I had not a single sound in the microphone. Then, as you can see in the DVD, she asked me to play the song. Two or three takes later, it was done! A very special moment."

Fabrice Lareau

Fabrice Laureau, aka f.lor, has been Yann Tiersen's sound engineer and co-producer since 1999. He's a musician too (he plays bass, but only on his own projects), and a mastering engineer. He was born the same year as Tiersen, and his career is very atypical. Instead of studying audio engineering or becoming an assistant, he got a Ph.D in wave physics. As he didn't want to become a researcher, he quit to go on tour with his band. "I then evolved from musician to sound engineer, but I don't consider myself a 'real' sound engineer, although I have already recorded around 30 CDs. I've worked in a lot of studios, big ones and small ones, in analogue and in digital, but I don't really have a favourite studio. I know how to use a console, a microphone or a recorder, but I wouldn't be able to modify a mixer channel or outboard processors. I like the sound that engineers like Steve Albini or Mario Caldato [*who works with the Beastie Boys*] get, where you can 'hear the room' on the record. I love this kind of ambience; I use few artificial reverbs, and most often it's a plate reverb. I don't pretend to know the whole truth in the domain, but to my ears a plate reverb fits into my mixes better than a Room preset, for example."

Ten years ago, Fabrice created the Prohibited Records label with his brother Nicolas, and produced bands in Black Box Studios, in London. "I liked the results, but sometimes, it didn't sound exactly like I wanted. As we had an eight-track tape recorder, we recorded our own band, NLF Trio, with my brother and Ludovic Morillon [*now Yann Tiersen's tour drummer*], in our rehearsal room. Some people asked who had recorded it, called me, my name became known, and I ended up recording [*French singer*] Françoiz Breut's album, *Vingt à Trente Mille Jours*. That's how I met Yann — he wrote some arrangements on that album — and Dominique A, who asked me to do some mastering on a compilation box of his songs."

Photos: Franck Ernould

Fabrice Lareau, Yann's engineer, with the mobile rig he's using to record Tiersen's latest tour.

Finishing Touches

During the mix, Fabrice Laureau doesn't recreate dynamics with fader movements: "I write some volume automation in *Pro Tools*, but I try to get the nuances from the beginning. That's not because I'm lazy — I like it when things go their own way. When I mix, I rarely compress: I prefer limiting."

Davout's studio A houses an SSL 9080K console and six Prism ADA8 high-end converters. "I love the 'main' compressors on the 4000 Series, but this aspect has changed completely on the 9000," says Lareau. "I was surprised: globally speaking, the console's sound is a lot clearer. For the first time ever, I didn't recognise what I sent to the console, and I had to cut high-end, almost 3dB at around 7kHz, inserting a GML EQ. That's a lot for me: most of the time, 2 or 3 dB is the maximum, unless I want to 'destroy' a sound or exaggerate an effect."

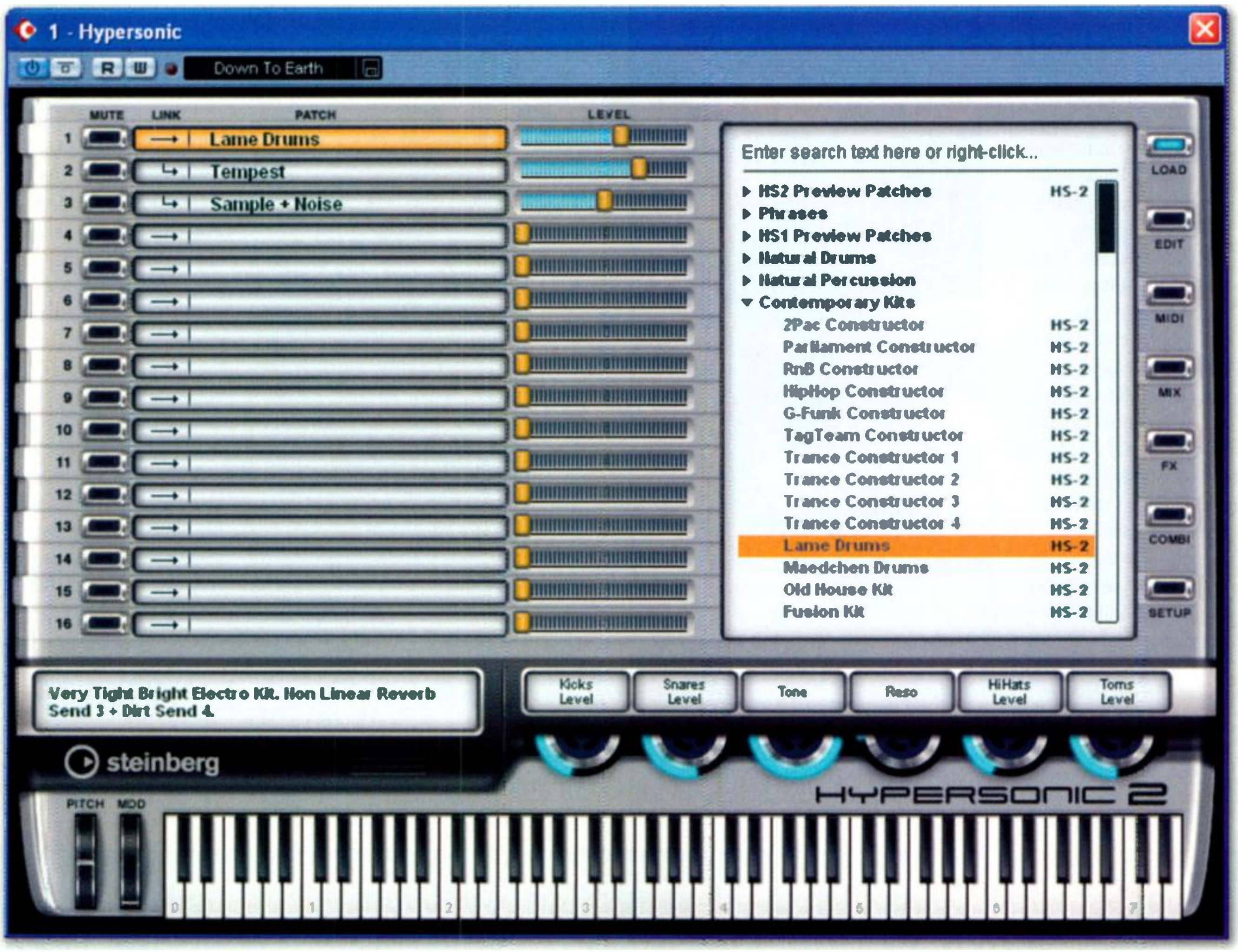
Fabrice had another surprise. He mixed the songs in Pro Tools HD and on half-inch analogue tape, but he didn't expect the results he got. "We wanted to compare the sounds during the mastering phase, and although the tape gave an nice overall effect, it lost a lot of detail compared to the digital mix. On Yann's music, analogue tape didn't work. It would probably have been different on a more rock & roll album, with more energy, but each time we compared them, we heard more detail in digital. We wasted all this analogue tape for nothing!"

Fabrice occasionally practices the art of mastering, and he's got strong ideas about it: "I do it at home, in *Pyramix* or *Logic*, with *T-Racks* and a few selected plug-ins — I know my tools well. For me, the mastering stage must produce a more listenable record. It's no use looking for the last dB of gain, just to be louder than the others. For Yann's album, we tried to work with Roger Seibel [*SAE Mastering, Arizona*] again, as we loved his work on previous albums, but he didn't find the right direction this time. Then we went to Abbey Road, with Chris Blair. We worked together, song by song, and I always understood his settings, even if he doesn't use the same tools as I do. He kept the transparency and added some subjective level, using some old Junger Audio digital compressors. We tried to burnish the brilliance we had already heard during the mix, and Chris added some low end with his vintage EMI console to balance the sound. I'm not the first to notice this brilliance: perhaps it's the Prisms? Anyway, it pleased the classical and film-music engineers who are the main customers of Studio A in Davout."

The control room in Tiersen's studio: the Mackie D8b is mainly used for monitoring and clocking these days.

Immediately after the new album's release, Yann Tiersen began an extensive European tour, playing for audiences who still know him above all for *Amélie*. Fabrice confirms: "In concert, half of the audience are just waiting for Yann to pick up his accordion! On this tour, the ambience is very electric, almost progressive music sometimes. Yann plays rock! The basis of most songs is bass, drums and electric guitar, and even the Ondes Martenot is played through a guitar amplifier. I am following the tour to record some concerts, and Aurélie is shooting everything with her digital video camera. Yann intends to release a DVD about the tour, but with a general ambience like *La Traversée* — not another concert DVD... At each show, I am recording 26 tracks on site with *Logic*, my Powerbook and my RME [*Octamic*] interfaces. It all fits in a little rack!"

Thanks to Emmanuel Plane. SOS



Steinberg Hypersonic 2

Virtual Workstation Synth For Mac & PC

The new version of Steinberg's virtual sound module brings more and better instruments, without sacrificing convenience or ease of use.

Sam Inglis

The original *Hypersonic* soft synth was much misunderstood. Some people expected it to be a vast, multi-gigabyte sample library, with every nuance of every instrument captured at a million velocity levels. Instead, it was a creative, easy-to-use workstation with a key quality that is missing from so many soft synths: immediacy. That's not to say it sounded bad, just that it met a different need from *BFD* or the Vienna Symphonic Library. It didn't offer detailed control over a million parameters, but nor did it take a minute to load every patch, or require a cutting-edge machine to play a single chord. It was, in short, the ideal sketchpad for anyone who wanted to get sounds fast.

Two years on, Steinberg and developers

SOUND ON SOUND

Steinberg *Hypersonic 2* **£280**

pros

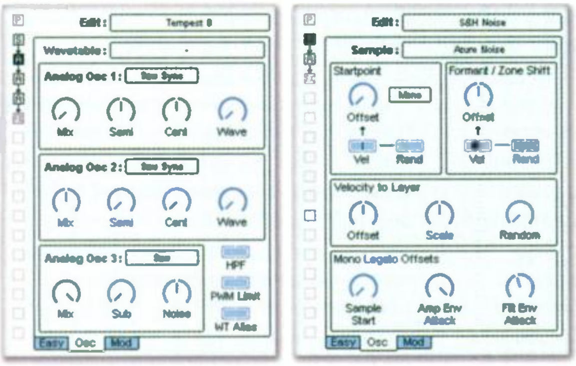
- Very immediate.
- Still economical on memory and CPU power.
- Instant patch loading makes auditioning sounds a breeze.
- Combi Chains will be useful for live work.
- Much more editable than previous versions.
- Most of the new sounds are excellent.

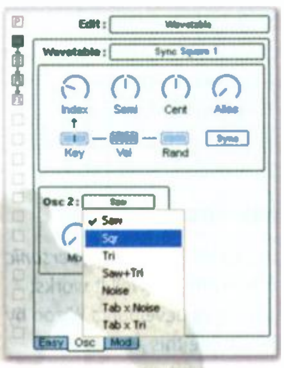
cons

- Given the sixfold increase in the size of the library, there aren't actually that many new sounds.
- Still not many useful orchestral patches.

summary

Hypersonic 2 is a worthwhile update that retains the program's immediacy and ease of use while significantly improving its functionality and adding some great new sounds.





As well as FM (not shown), *Hypersonic 2* features subtractive analogue-style synthesis, sample playback and wavetable synthesis.

Wizoo have brought out version 2. Their aim seems to be to build on version 1's strengths, whilst answering those critics who found the original a little bit superficial. In the former category, improvements include a new Hyperphrase arpeggiator and the ability to switch between multis on the fly, for seamless patch changes during live performance. To the latter end, the sound library has grown from 250MB to 1.7GB, and now features lossless data compression for better quality, plus there's greater freedom to edit patches. As before, it's authorised to a Steinberg Key using the Synchrosoft copy-protection system, but this time, no key is supplied, so anyone who doesn't already own one will need to budget for an extra £20. Dongles are always hard to love, but I didn't have any problems authorising mine.

New Sounds

Despite the sixfold increase in the size of its library, there are only about 50 percent more sounds in *Hypersonic 2* than there were in version 1. This must be partly down to the decision to abandon lossless compression, but also reflects the fact that the new sounds prioritise quality over quantity. Quite a number of them are actually 'extra large' versions of *Hypersonic 1* patches — the same instruments sampled with more velocity layers and less obvious looping. Others are wholly new, and in both cases the improved quality is readily apparent. Out of more than 1800 factory patches, everyone will have their own highlights, but mine are the Contemporary drum kits. These were impressive in version 1, but new patches like Electroquirker and Fat Snap Kit really leap out of the speakers at you. There are plenty of decent new synth sounds, more excellent Hammond patches, a very fat and funky Clavinet, very passable pianos, and the ▶

Going Hyper

I'm not sure what the difference is between a Hyperphrase and an ordinary phrase, unless it's just hype, but the upshot is that the *Hypersonic* arpeggiator has acquired the ability to play phrases as well as the usual up/down patterns. Phrases can be short melodies, rhythmic patterns, whatever you like — over 200 phrases are included, and you can import MIDI files of your own. Personally, I'm not a great user of arpeggiators, but if I was, I think I would really like the one that comes with *Hypersonic*.

pitched percussion instruments such as vibes and glockenspiels are worthy of mention. Elsewhere, there are lots of new string ensemble patches, but as with all such things, I personally can't see the point — I'd trade the lot for a single decent solo violin.

Steinberg say that you can now edit every parameter in a *Hypersonic* patch, and it's certainly much more malleable than version 1 was. The six Hyperknobs are still pre-assigned to the most important parameters or groups of parameters in a patch, but detailed editing is done by clicking the Edit button at the right. As before, each patch is presented as a series of vertical blocks containing Elements, and clicking on a block brings up that Element for editing. These Elements can either be one of four varieties of synth (a sample-playback synth, an analogue-style synth, an FM synth or a wavetable reader), or an FX unit. In each case, they present at least the most important parameters you'd expect, although the amount of control available doesn't really compare with a more free-form synth. You can't, for instance, select different algorithms in the FM Element, or easily change the keygrouping of samples in a sample-playback synth, while the architecture of the subtractive synth is fixed and fairly simple.

The biggest improvement over version 1 is that you can add and remove Elements from patches, rather than simply muting them. This doesn't make the program into *Reaktor*, but it does make it realistic to create original patches, providing you're happy to rely upon the supplied samples and wavetables. It would have been nice if the manual had been updated to cover the new editing features, though. It's also a bit odd that you can't click on an empty block to add an Element — you have to click on an existing Element and insert the new one into the chain. The new one always appears before the selected block, so the upshot is that you can add new Elements anywhere but at the end of the chain!

Final Thoughts

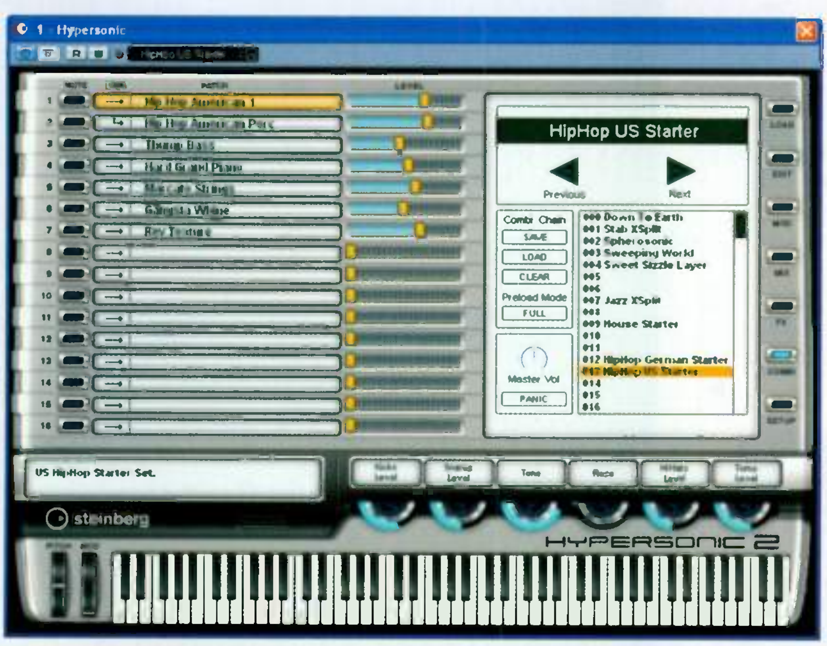
The only real concern I have about *Hypersonic 2* has nothing to do with how well it works: it's about the takeover of developers Wizoo by Digidesign. Some will see this as casting doubts on the long-term future for *Hypersonic* as a Steinberg product, especially since Digi's new free instrument plug-in *Xpand!* appears to be closely related to it. However, Steinberg are promising a 'full support programme' for *Hypersonic 2*, and it would be most unfair to judge the product that exists now on the basis of speculation about what might happen down the line. In all important respects, I like *Hypersonic 2*. I did encounter the odd quirk — for example, on a few of the new drum kits, the Hyperknobs seem to work backwards, and some aspects of the Combi handling are a bit weird — but for the most part, it is very intuitive to use, and the improvements are just what the doctor ordered. Compared to rivals such as IK's *Sampletank 2*, the sounds tend to be brasher, brighter, and more dependent on effects, which will suit some users better than others, but *Hypersonic*'s simplicity, low system load and instant patch loading will be big pluses for everyone. I still think it's a shame that there are few decent solo orchestral patches, but this seems to be true of most comparable products as well. As a sketchpad for rock and pop tracks, it's still the best way I know to get sounds together fast — and you'll be surprised at how often you keep them in the finished track! SOS

Switching Horses

One of the great features of the original *Hypersonic* was its rapid patch loading, and Steinberg have taken this several steps further in version 2. Patches load instantly even while a song is playing back in your host program, which is fantastic when you want to audition different sounds for a part you've created. Beyond that, it's now also possible to set up Combi Chains for live use. A Combi is, as in any multitimbral synth, a snapshot of all *Hypersonic* settings, including information on which patches are loaded into which slots. (*Hypersonic 2* makes it very easy to layer multiple patches for a single MIDI input, which is great.)

I was a little confused by the fact that you can't actually load and save Combis from the Combi editing page — the controls are in the Setup display. Instead, the Combi page is devoted to a list containing up to 128 numbered slots. Right-clicking on any slot allows you to assign a Combi to that step in the list, whereupon you can use the Previous and Next arrow keys to step forward or back through the list, with *Hypersonic* updating all its settings almost instantly. For extra-smooth changes, you can set *Hypersonic* to preload either the next Combi or all Combis in the chain, although I found it often worked fine without this. A nice touch is that you can leave gaps in the chain, in case you want to insert additional Combis at a later date. Sustained notes are maintained, with the original sounds, over a single change of Combi, but cut off if you change again.

Combi Chains will be a really handy feature for anyone intending to use *Hypersonic* live, and I can't think of many other workstation-type synths in software that offer a similar feature — indeed, lengthy loading times are an Achilles' heel for many rival products. However, there are a couple of things that puzzled me about the way they are implemented. An orange bar highlights the 'active' slot in the Combi Chain. Double-clicking on another slot makes that slot active instead, and loads the Combi associated with it, but single-clicking simply highlights a slot without loading the Combi or making it active. This means it's easy to let the highlighted bar get out of sync with the active slot, which can get confusing, and I'd have preferred to have the two permanently linked — there's nothing much that can actually be achieved by single-clicking on a slot in any case. I also had difficulty loading my own Combis into slots in the Combi Chain: the factory patches could be selected from the list, but the User Combis I created refused to appear in it. I asked Steinberg about this, and they said no-one else had reported it, so perhaps it's a freak occurrence on my computer.



The ability to step through a Combi Chain at the click of a button will be a handy feature for those using *Hypersonic* live.

information

- £ £279.99, upgrade from version 1 £89.99. Prices include VAT.
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Studio SOS

This month the team convert a small empty room into a professional vocal booth within the space of a single afternoon!

In order to create a bass trap for the booth, the team first built a large wooden frame (with corners reinforced using metal brackets) and moved this into position on the room's rear wall.

Paul White & Hugh Robjohns

Paul Joyner is a music technology lecturer who asked for our help with creating a vocal booth, something we've never before done from scratch. Because we were able to talk to him at the planning stage, he was able to design a room that left enough space for the acoustic treatment, which many people don't adequately consider. Typically, you end up having to cover walls with between three and six inches of treatment, meaning that the room's dimensions shrink by up to a foot in each direction! A small empty booth quickly becomes an unworkable cupboard unless you factor in the space required for the room treatment.

Building A Vocal Booth From Scratch

Paul's vocal booth was to be created in a space to the rear of his existing studio control room, and he had walled the area and lined it with plasterboard prior to our arrival. At our suggestion, he'd also fitted an off-the-shelf, double-glazed exterior glass door to the booth to provide a useful degree of sound isolation while still retaining a physical line of sight to the performer. Paul had sloped the ceiling up towards the door of the booth and had also fitted a carpet over his dual-skin chipboard floor, but that was as far as he had got before we arrived. So we were presented on arrival with a bare room measuring around five feet by five feet six inches, with a ceiling height ranging from under seven feet at the back to roughly eight at the front. As expected, singing in this bare room sounded like singing in a cupboard — very reverberant with a pronounced boxy effect and obvious ringing and resonance artefacts.

I know that some people have experimented with carpeting the walls of booths, but experience shows that this approach seldom produces usable results. Carpet only absorbs high frequencies, so the mid-range and low end carry on resonating as before, creating a boxy sound with no life at the top end. Then there are those who line the entire thing with Rockwool, but unless this is very thick it still doesn't deal with low-frequency problems adequately and can leave the room sounding very dead and oppressive. The same is often true of thin acoustic foam, which mops up everything above 300-400Hz but leaves the low end to run riot. What we needed was a mixture of absorption and reflection with enough low-frequency trapping to keep the room behaving evenly down to the lowest vocal frequencies and beyond.

Over the past few years, a number of acousticians have experimented using

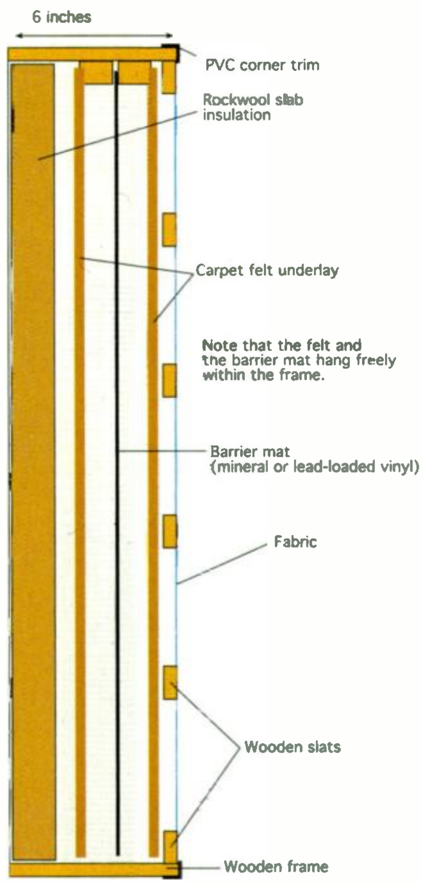
Once the wooden frame had been fixed in place, using a combination of screws and cavity-wall fixings, Paul and Hugh stacked a layer of Rockwool slabs to make up the back of the trap.

hanging sheets of barrier mat to absorb low frequencies. Barrier mat is essentially mineral- or lead-loaded vinyl, and weighs typically from 5kg to 10kg per square metre. It looks a little like linoleum, but its high mass and reasonable flexibility makes it a very useful material in all kinds of acoustics applications. The advantage of this material is that it allows a trap to be built with much less panel depth than would be required for a simple Rockwool panel, yet it still achieves similar low-end performance.

We decided to build a trap to cover most of the rear wall. We chose the back wall partly because this provided the largest single surface area for treatment, but also because it is the wall behind the singer. When using a cardioid microphone, it is reflections from behind the singer's head and body that get into the microphone and cause the most problems. Soaking up those reflections to stop them from getting back into the mic makes a huge difference. When recording in open rooms, you can clean up the sound considerably by hanging a thick duvet behind the singer, but we know that a lot of people mistakenly put the duvet in front of the singer (behind the microphone) as they intuitively feel this will help. It may help a bit, but the surface behind the singer is the most critical one.

The trap we were planning used a high-density Rockwool slab at the back, with a hanging curtain of barrier mat plus two further curtains of high-density felt carpet underlay in front. The key point here is that the felt and barrier mat are left to hang freely within the panel, and can thus move slightly to absorb low-frequency energy. We had around six inches of depth to play with, so we built a wooden frame six inches deep, six feet tall, and four feet wide. The idea was to screw the side panels together *in situ* and fasten them so the rear wall before we introduced the acoustic materials. The corners of the frame were strengthened with metal brackets obtained from the local DIY shop and battens were fixed to the inside rear edges of the side panels to facilitate fixing to the wall. Barrier mat is surprisingly heavy, so reinforcing the corners of the frame and making sure the frame was fixed very securely to the wall was important.

The spacing between the battens used to fix the side panels and the wall was set at 47 inches, which is exactly right to allow a block of Rockwool cavity wall insulation slab to be wedged between them. Another set of battens was placed inside the front edge of these side panels and set back by the thickness of a batten so that we could fix a few horizontal retaining wooden strips across the front opening prior to fitting the final cosmetic covering material. These would prevent the cover material being disturbed by the felt hanging behind it, and would also provide some protection against performers leaning on the trap. Our frame also included a couple of small horizontal wooden supports at the tops of the side panels, positioned so that a couple of lengths of batten could be slid in from the front to support our acoustic curtains.



Here you can see the overall design of the bass trap that Paul and Hugh built for Paul Joyner's vocal booth.

Once we'd fixed the frame to the wall using a mixture of cavity wall fixings and screwing directly into the wooden studding behind the plasterboard wall, we wedged the Rockwool in place, where it stayed securely without the need for any form of adhesive. If you have sensitive hands, you should use gloves to handle Rockwool, though the slab is rather less 'nasty' than loose loft insulation when it comes to shedding fibres. Wearing a face mask and safety goggles is a good idea, though, as loose fibres and dust tend to fill the air while you are handling the slabs. According to one major acoustic materials supplier, tests apparently show that Rockwool poses no direct health risk, although it is still an unpleasant irritant.

We needed to cut the last piece of Rockwool slab down to fit the available space, and my bandsaw did this very neatly. Paul also used it to cut any foam panels that needed modifying, but if you don't have a bandsaw, an electric carving knife is very effective. Even an ordinary bread knife works, but the results aren't quite as neat.

Once the Rockwool was in place, we took one of our hanger battens and fixed the top edge of the carpet felt to one side, and the barrier matt to the other using a basic hand stapler. The batten was one inch wide and established the correct spacing of the materials. We then used flat-headed tacks to reinforce the stapled joint — barrier mat is far too heavy to be supported by staples alone. We'd used a barrier mat that weighed ▶

Once the Rockwool was in place, layers of carpet underlay felt and barrier matting were hung from slats at the top of the trap, and were left to hang loose.

5kg per square metre and which had a built-in fabric reinforcement to prevent it deforming under its own weight (as some unreinforced barrier mats are prone to do). (Paul bought this mat from www.siderise.co.uk, which is a gold mine of studio building materials and information. However, although they'll ship orders for you, they don't accept credit cards so you need to send a cheque in advance. They can also provide very valuable telephone help and advice via their technical department.)

After sliding in our first 'curtain' (carpet felt to the rear, barrier mat to the front), we took a second one-inch batten and fixed another sheet of carpet felt to it. This was then inserted with the felt at the front so that it hung around one inch in front of the barrier matt. The type of carpet felt we used is the fluffy brown material around half and inch thick that smells as though it's made from recycled wet labradors! Modern foam underlays are not suitable for this application, though heavy carpet with a porous fabric backing can be used in this type of trap — as can unloved duvets! Again, handling the carpet felt generates a lot of dust, so a mask and goggles are a fairly sensible idea! From back to front, the nearly completed trap now comprised a layer of high-density Rockwool slab around two inches thick pressed firmly to the back wall, a loose hanging curtain of felt underlay, a single sheet of barrier mat, and then one further sheet of felt underlay, with one-inch gaps between each layer.

To finish off the trap, we screwed a few protective battens in place across the front, one at a carefully calibrated Average Bum Height and another at shoulder height, with a few more to protect the intervening spaces. We then stretched a sheet of muslin over the frame and stapled it all the way around. You need to take some care here to ensure the fabric is tensioned evenly and has no nasty creases in it. Don't skimp on the staples! Once this was tight, we trimmed off the surplus with a Stanley knife and used PVC right-angle corner strip, again bought from the local DIY shop, to tidy up the edges. This was mitred at the corners using a cheap and cheerful mitre saw. We used a contact adhesive applied with a mastic gun into the crease of the plastic strip to hold the trim in place, and secured the corners with masking tape until the adhesive had set.

The Rest Of The Room

The finished trap did look rather good, and at this stage the room was sounding much better, but we still had the other walls to deal with. For the ceiling, we decided to try out a panel of pyramid-profile foam from Making Waves Audio, as their UK distributor had arranged for them to send some samples over. The panel was around a metre

After another layer of carpet underlay felt had been added over the barrier matting, slats were fixed across the front of the trap, with muslin stapled over them for a neat finish.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square and four inches thick, and it fitted the ceiling space perfectly without cutting, leaving a little bare plasterboard around the edge. We didn't want to cover all the plastered surface, as some mid-range and high-frequency reflection is necessary to prevent the room sounding too oppressive. We could hear the difference as soon as we fitted this, and it looked pretty cool too!

For the walls, we used some three-inch Auralex panels, which we spaced away from the wall in order to get better low-frequency absorption. However, we couldn't afford the space to use the three-inch foam spacers provided, so we cut these into one-inch slices, again using my bandsaw. We glued one in each corner and one in the centre of each panel. Even using a one inch spacing improves the effectiveness of these panels to a worthwhile extent, and the Auralex spray glue stuck them to the walls with no problem. Two 4 x 2-foot panels were fixed vertically to one side wall, but the other had a small window in it, so we fitted just one panel to that side and then used some smaller panels cut from a thinner sheet of Making Waves Audio pyramid-profile foam to break up the remaining flat surfaces around the window and on the wall around the door. These were again glued directly to the wall using Auralex spray adhesive.

Proof Of The Pudding

To test our handiwork, we set up Paul's Rode K2 microphone in the booth and then took turns to go in and speak while the others listened via the studio monitors. As we couldn't stick foam on the glass door itself, we clipped an Auralex foam 'butterfly' baffle onto the mic stand to help absorb reflections sneaking in from the direction of the door. The difference between the original bare room we started with and the treated room was huge, as we expected it would be, but we were still surprised at just how good the result was. Even pushing the mic into one corner and standing as far away from it as possible in the other corner (about three feet away) resulted in a nice clean sound with no trace of small room boxiness. At the same time, there were enough reflective surfaces left inside the booth that the sound wasn't oppressively dead, and it was quite pleasant to sing and talk in the room.

So although our trap design was to some extent experimental, the end result was as good as we could have hoped for, and it was all pretty simple to do. We left the wooden frame unfinished, to match the other bare wood surfaces in the room, but it would be simple to paint or stain the timber to produce a panel that looked very professional indeed.

The finished vocal booth, complete with rear-wall bass trapping, side-wall and ceiling acoustic-foam treatment, and a 'butterfly' foam baffle fixed to the stand behind the mic.

During our tests, we noticed that the floor of the booth was quite noisy, with a tendency to be a little boomy in response to a tapping foot. We asked Paul what he'd done there, and he revealed that he'd made the floor from two adjacent layers of chipboard , but hadn't bonded these together. Our suggestion was to add an additional layer to increase the mass if possible, but in any event to use additional woodscrews to fix the layers together more firmly. Fitting a thick underlay beneath the booth carpet would help too. Without this, there would be a risk of foot-tapping being picked up by the microphone. A thick, heavy matting made from recycled car tires is available that would be even better in this application, and you can get this from a number of acoustics- and building-product suppliers.

Paul had already fitted a couple of XLR tie points through the wall to allow him to get headphone and mic signals into and out of the booth — although only one of each. To add to the flexibility, we felt he should also provide access for guitar cables, speaker leads and so on — after all, a vocal booth is a great place to put a guitar amp if you want to sit in the studio and record while monitoring the sound over the studio monitors. An easy solution would be to drill a hole in the wall large enough to take a length of plastic waste pipe (1.25 or 1.5 inch) through which cables (and the plugs on the end of them!) could be threaded. Chunks of foam could then be pushed into the pipe at each end to prevent serious sound leakage. Paul thought this was a good idea, because he'd already found the system he'd installed was a little inflexible when he wanted to use his tube mic — the multi-pin XLR cable is very different to a regular mic cable. For our tests we had had to put the mic PSU in the booth.

The complete job took us only four hours (part of which was taken up eating Paul's chocolate Hobnobs, drinking tea, and talking shop), so a job like this is easily within the scope of anyone who feels comfortable doing basic DIY. If there's a point to stress, it is that vocal booths require a decent thickness of acoustic treatment if they're not to end up sounding boxy. However, at the same time you need some reflective surfaces to dispel the 'padded cell' vibe. The biggest part of our project was the rear wall trap, and this cost around £120 in total, though we had enough Rockwool and barrier mat left over at the end to build a second smaller trap if required. Adding the cost of the foam panels and adhesive, the total budget for all the treatment came to around £250-300 — a small price to pay for converting an empty shell into a first-class vocal booth. SOS

Session Comments

Paul Joyner: "The advice given by Paul and Hugh from the outset was invaluable. The bass trap has made a significant difference to the look and the feel of the room, and with the additional foam added the booth sounds a lot tighter. I've now acted on the advice about the floor area, and this has already made a difference with some foot-tapping vocalists. In all, the day was very successful, and I'm now the owner of a great little space for recording vocals and guitars."

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Taken from skillfully recorded studio sessions, each guitar part has been effected, broken into a multitude of pieces, and then carefully and creatively re-assembled. The result is a DVD-ROM collection of over 400 loops that contain fresh melodic ideas with uniquely sliced and stuttered rhythms. Chopped Guitars will serve as a starting point for song ideas or add texture to existing tracks.
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BREAKBEAT JAZZ (EQUIPPED MUSIC)
If you are into that turntable culture of HipHop, then go right ahead, jump into these crate diggin sounds! Original sounding breakbeats, mixed up with vinylstyle rhodes chords, jazzy basses, FX and much, much more. 4Gb comprising 2100 Rex2 loops and 3700 Wav loops, formatted for Stylus RMX and Reason.
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Want to delve deeper into the naked, intimate sound of House? Welcome inside the hypnotic and seductive grooves of Loungin' House. Luxurious dance floor beats, mixed with sexy latin style percussion, lush chords, samples and more. With over 4Gb of material, this is the ultimate House groove toolbox for Stylus RMX and Reason users!
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URBAN JOINTZ (UEBERSCHALL)
This library is aimed at urban producers looking for street-certified modern production elements: banging drums - hard hitting basses and keys. These are real sounds from the streets, covering the entire spectrum of urban music and crafted in the styles of the hottest chart topping producers in the game.
APPLE LOOPS+REX+WAV - £59.95

NU JAZZ FUNK (EQUIPPED MUSIC)
Can't get enough of those wild, syncopated funky beats of Nu Jazz? Then enter the brazilian flavoured sounds and moods of Nu Jazz Funk. Super funky broken beats, mixed up with jazzy rhodes, moogy basses, percussion and more. This is the ultimate Nu Jazz groovebox for Stylus RMX and Reason users!
REX2+WAV - £79.95

BREAKMATIC (TEKNIKS)
Tekniks presents Breakmatic the 3rd installment to the "Matic" series. This exciting Breakbeat library will captivate your imagination and fire up your creativity. This CD is packed to the very limit with the grittiest & grimmest sounds and loops ever found and will surely not disappointment even the true vinyl junkies at heart.
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BROKEN BEATS (SAMPLELAB)
Guerilla beats & low slung breaks! Broken Beats is full of dirty mash-up breaks and deep down lo-fi beats - with over 600 heavy duty loops, this production kit is the authoritative breaks collection. Sample accurate stackable loops broken out by BPM. Produced by the same team famous for the award-winning Luscious Grooves & Discography
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EZ ROLLERS (ZERO-G)
One of the UK's premiere Drum & Bass production teams have finally made a sample library, and it rocks. This DVD is the most happening selection of D&B samples on the planet from one of the world's best D&B production teams. A killer collection of the freshest beats, basses, pads, synths, fx, vocals and weirdness!
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DRUM STYLES (ZERO-G)
An exciting and flexible range of Acoustic Drum Loops & Single Hits in styles such as funk, hip hop, jazzdance, rock, drum & bass and house together with jazz and many examples of more exotic grooves like reggae, latin and more. As a bonus there are percussion loops to add into the mix and a selection of single hit drum sounds
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DANCEPACK 2 FOR GARAGEBAND (ZERO-G)
Dance Pack 2 picks up where volume 1 left off and features a whole host of samples taken from Zero-G's award-winning back-catalogue, re-programmed into Apple Loops format. Construction Kits, Drum & Percussion Loops & Hits, Breakbeats & Groove Loops, Funk Guitar & Bass Riffs, Trance and a Vocal samples.
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HYBRIDIZER 2 (BIG FISH)
With over 6500 different pieces of sonic inspiration, Hybridizer 2 transcends all musical genres! Even more journeys of a bassic nature, distended nuclear rhythms, more manic metal guitar licks and scrapes than you ever thought possible. Thrashed and pulverized, vivisected and scourged, this library is sure to inspire!
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Sample the future

Introducing the Origin Series

Polyester Loops features synthetic sounds of all descriptions which have been expertly combined together to present a wealth of loops and complex velocity layered drumkits like nothing you have heard of before. Whether you are producing drum and bass, electro, trip hop, dig, house or film music - we hope you'll check this Cd as it really is a 'must have' title for your collection.

Soulful House Sessions encapsulates the more laid back soulful house grooves, more jackin styles, and also live and latino spirited grooves. If your making music for either bar/club settings with the more discerning ear in mind then this CD is an essential resource. Inspired by the likes of Masters At Work, Copyright, Jamie Lewis, Joey Negro, Kerri Chandler, Miguel Migs & DJ Gregory.

Combinations Reason Refill contains over 800Mb of content from the Origin Series range of award winning sample libraries, including hundreds of brand new combinator instruments and Reason patches for all serious Reason 3+ producers. Also to be found are more than 70 Instrument sounds and multisamples, 370 NNXT patches, 82 Malstrom Patches, 101 Subtractor Patches, 30 Redrum Drumkits and much more.

Cult Club Synths includes a huge collection of samples and patches created with the Access Virus, a synth known for it's power and versatility in dance production circles. There are over 100 Arp Patches, 53 Bass Patches, 88 Lead Synth sounds, and 66 Pads, all have been programmed with the dance-floor in mind, just add a bit of your favourite effects and blend them seamlessly into your track! If you produce dance music, you'll love this DVD!

Electro Breaks is a serious production tool which includes hundreds of new breakbeat and electro drumloops, breaks, heavy basslines, twisted textures, resonant synths, atmospheric pads, keys, fx, hits, kits and multisamples. Whether you are looking to make Mylo inspired Electro House, or Heavier Breaks material like Freq Nasty, Electro Breaks has a huge palette of cutting edge underground breaks and samples just for you.

Funky House Producer brings you the sounds of Ibiza, London, Paris and New York directly to your mix. Featuring the latest filtered funk loops, inspirational disco guitars, huge pumping beats, funky bass lines, fx and killer keys. Whether you are looking to capture the blissed out production of Naked and Wave Records, or perhaps something deeper – you'll find that Funky House Producer delivers on every level.

Afro-Latin Producer brings you an huge selection of authentic live played drumloops, brass samples, guitar riffs, live basslines, soft and expressive keys, live percussion loops, hits, fx, patches, FX and Vocals to merge into your tracks from some of the best session players available. Whether you are creating Latin House, Brazilian Drum and Bass, World Jazz, or something more experimental, this CD will take you on a trip of inspiration!

Hip Hop Progressions is a versatile hip hop production tool, including huge folders of live and programmed drumloops, turntable fx and scratch loops, human beatbox, funky guitar samples, live and synth basslines, hip hop keys, inspiring musical loops, hits, fx, percussion loops, strings, stabs and much more. Incorporating UK and US influences to bring you an essential sample library of hip hop samples and loops!

Acidized-Wav / EXS24 / HALion / Kontakt / NNXT / Rex2 - £39.95 each

loopmasters

Interesting Times

Marcus Ryle, Michel Doidic & Jeff Slingluff Of Line 6

Marcus Ryle (left) and Michel Doidic of Line 6.

Line 6's first product was the Axsys 212 guitar combo (left), but the one that really made their name was the Pod (middle). Their most recent are the Toneport audio interfaces, reviewed in last month's *SOS*.

Paul White

Most musicians know of Line 6 through their digital modelling guitar amps and Pod preamps, but few appreciate the lengths they go to in order to recreate the sounds of classic amps and, more recently, guitars. However, to really understand Line 6, you need to know a little about its two founders, Marcus Ryle and Michel Doidic, who previously worked together under the banner of Fast Forward Designs, where their developments helped to bring such milestone products as the Alesis ADAT into the world.

"That Call's For... Line 6"

Michel, who also enjoys playing guitar, gained a Bachelor of Science degree in electrical engineering from the University Institute of Technology in Angers. Marcus Ryle operates as the company's Senior Vice-President of Product Development, but in a company like Line 6, engineering skills are only part of the story, as musical ears are also important. Marcus has a background as a classically trained pianist, but he was soon seduced by the dark side of electronic keyboards and has been involved with a number of TV and film soundtrack projects, as well as working with professional recording artists as diverse as Barbara Streisand and Chicago.

Both Marcus and Michel worked as design engineers at Oberheim during the '80s, leaving to set up Fast Forward Designs in 1985. Their new company designed technology behind some of the most famous audio products of that era, including the Alesis ADAT range, Quadraverbs and Quadrasynths. They were also involved in Digidesign's Sample Cell, the Fostex RD8 recorder and numerous other pieces of hardware from companies such as Studer, Dynacord, and Steinberg. Eventually, they decided to move away from designing hit products for other people to doing it themselves, and as Marcus Ryle explained to me, DSP chips were just starting to get cheap enough and powerful enough to allow guitarists a taste of the sound design freedom that synthesizer players already enjoyed. In 1996, Line 6 started out on the road to guitar-amp and effects modelling with their first product, the Axsys 212 guitar combo. Apparently the name Line 6 came about because Marcus and Michel were cooking up some guitar-related ideas back in their days at Fast Forward, and to keep their endeavours under wraps, they had all guitar-related phone calls diverted to telephone line 6.

The company is now presided over by Mike Muench, who holds a Business Administration degree from Harvard University and a Bachelor of Science degree in Business Administration from the University of Southern California. Mike joined the company in 1998, prior to which he was vice president of the US Consumer Division at Apple. Could this be why almost every computer I saw at Line 6 is an Apple Mac? Like everyone else I met at Line 6, Mike also has musical connections, and he's worked professionally performing and arranging.

Today Line 6 employs approaching 250 people worldwide, which is a big step up from their original 10-strong team. Research and development is done in their Agoura Hills facility, half an hour's drive north-east of Malibu in California. Warehousing is taken care of in another California facility, but to keep costs down, all their manufacturing is done in China, where Line 6 also maintain an engineering office to liaise with their manufacturing companies. Designs for tooling and test equipment originate in the US, but the toolmaking itself is done in China using CAD blueprints created at Line 6 Agoura Hills. On my tour of their facility, the design department showed me their 'Easy Bake Oven' 3D plastic fabricator which 'grows' 3D plastic parts direct from CAD data. Using this they are able to create and test prototype parts and housings very quickly, including the housing for their new Toneport audio interface hardware.

What struck me during my tour of Line 6's various departments was just how nuts about music, especially guitars, everyone was. All the corridors bear plaques named after famous guitar players (many British I'm pleased to see — see overleaf), and after work, a number of the staff often stay behind to rehearse as a band on a small stage at the end of the cafeteria. Everyone lives and breathes the fine details of guitars, amps and effects pedals to the point that the term anorak could easily be applied (in the nicest possible way) to most of them. Over dinner, I had a long conversation with James Santiago, a very fine guitarist who is often seen demonstrating Line 6 products to their best advantage, and the conversation didn't stray far from the insides of amplifiers and effects pedals until the dessert trolley arrived.

In the design department, I was given an insight into the origins of the Tonecore pedal range, which started life as a simple cardboard mock-up to help the designers position the controls and connectors. Next, I was shown a book full of artist's sketches of what the pedals might look like. Apparently these are passed around key people to get some impression of which is likely to be best received by the target market. There are also some engineering niceties built in at the design stage, such as access to the battery compartment without the need for tools and an external power socket that can accept AC or DC power adaptors, thereby increasing the chance that the user will already have a PSU that will work with the pedal. Once a design had been settled on, CAD drawings were produced and a prototype shell grown from plastic so that all the dimensions could be checked against the circuit board and parts.

Guitars & Modelling

Before taking a look at the various R&D departments around the factory, I had the chance to talk to Marcus and Michel, and was curious to find out why they had chosen the road of guitar modelling rather than, for example, synthesizers. Marcus explained that while synthesists were already well catered for, guitar players still found it difficult to get exactly the sound they wanted and in the mid-'90s, DSP power was getting cheap enough to apply to finding a solution. While many players can set up one sound that works for them, life becomes more difficult when they need to switch between sounds, and where external effects are part of the sound, repeatability also becomes a problem. The Line 6 approach was to model not only the sound of the amplifier but also the speaker cabinet and the way it would be miked in a studio session. On top of this, the company developed numerous effects algorithms including a digital emulation of the spring reverbs used in typical guitar combos, and all this technology was put into the Axsys 212 combo. Though this was very easy to use when compared to keyboards, some guitar ▶

Fun With Hum

Lead sound designer Jeff Slingluff told me he once set up a blind listening test for some industry guitar players to see if they could tell the modelled Line 6 amp from the original. To start with, the listeners always picked out the Line 6 model even before a note was played — because the hum went away whenever the signal was switched to the Line 6 amplifier! To make things fairer, he set up a third tube amp between the two, and left it ticking over with no input so that there was background hum all the time, and after he'd done that, the results were split almost 50/50, which is exactly what you should get if the sounds are too close to tell apart.

The Amp Room, where Line 6 conduct their modelling tests, features a bewildering array of old amps and guitars.

players still felt it was too daunting, so when the next range of amplifiers was developed, the Line 6 engineers moved much closer to the look and feel of a conventional guitar combo.

It's probably fair to say that the first Line 6 product to really worm its way into the public consciousness was the original Pod, with its distinctive kidney shape and user-friendly controls. This was particularly relevant to *SOS* readers, as it simplified the process of recording electric guitar sounds and solved the problem of noise, especially when working late at night. As DSP chips became more powerful, the improved Pod XT was developed along with the later Flextone, Spider, and Vetta amplifier ranges.

Part of the appeal of products like the Pod XT, and the later Pod XT Live, is that their software can be updated and expanded. System software updates are generally available as free downloads and can be transferred into the device via USB using the Line 6 *Monkey* software, but Line 6 also saw the market potential of selling additional amplifier and effects packages known as Model Packs, which add to the repertoire of the device in much the same way as the add-on cards that synth players use to get new sounds and new presets. Currently the Vetta amplifier range is the most powerful and most sophisticated Line 6 amplifier, capable of running two amplifier models at the same time, but owners of the Pod XT or the new Toneport audio interfaces can use the optional Model Packs to upgrade their systems to take advantage of all the Vetta amplifier and effects models.

I asked Marcus and Michel whether they thought DSP was finally fast enough and cheap enough that they no longer had to compromise their algorithm designs to get the product to market at an affordable price. Marcus replied that he felt DSP power was now far less of an issue than it once was, but power consumption was still a concern in the case of battery-powered devices, such as effects pedals and portable amplification. One factor I hadn't appreciated was that although the Tonecore pedals all use exactly the same DSP engine, battery life is heavily dependent on how busy the DSP chips are: some algorithms use power at a greater rate than others. They also let me into a secret regarding these pedals — the control section, which also contains the effect algorithms, is in fact a removable module, so by fitting a different module to any pedal in the range, its function can be changed. The DSP itself is part of the main body of the pedal, but as every pedal has the same DSP engine, any module can be used in any pedal. This modularity is clearly part of a future marketing strategy.

As I understand it, the original Pods modelled the behaviour of discrete sections within a guitar amplifier to replicate the behavour of the original EQ, preamp distortion, power-amp distortion and so on. I asked Marcus whether convolution techniques were applicable to guitar amp modelling now that we have enough DSP power to handle it, and he surprised me by revealing that convolution was one of their 'secret weapons' and had been used since the very first Pod as a means of replicating the speaker cabinet and its miking arrangement. Convolution is essentially a brute force number-crunching process most often associated with 'sampled' reverb, but the longer the reverb, the more number-crunching is needed, and the more DSP power that takes up. In the case of miked cabinet modelling, the events being measured are relatively short, which is why Line 6 were able to use convolution right from the start. It also explains why they didn't offer the choice of more distant microphones, as the greater the distance, the longer the convolution period required.

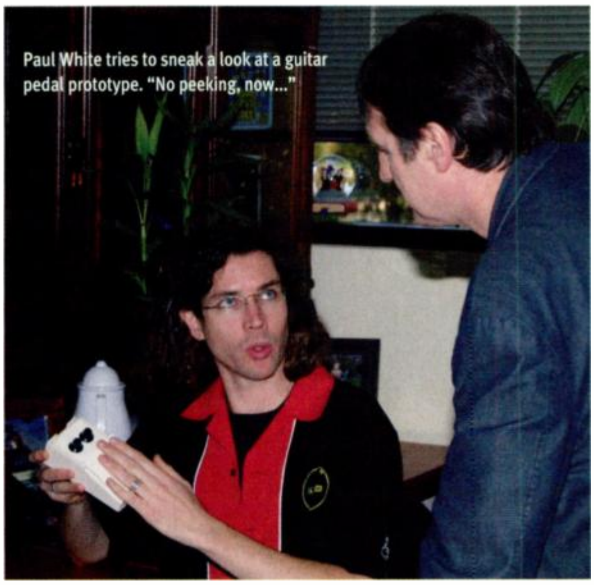
To model the cabinets, they were first set up in a typical recording studio, on the floor, with the correct mics in the designated place, as listed in the Pod or amp settings. If the manual says 'SM57 On-axis', then that's what was used to make the measurements. There's always some interaction with the room, and the engineers wanted the result to be as authentic as possible in reproducing a 'studio' sound, which is why they didn't measure the cabinets in an anechoic chamber. This set me thinking that if you did want the sound of an amplifier miked with both a close and a distant mic, it should be possible using a Line 6 amplifier (or a Pod connected to a suitable amplifier and guitar speaker) if you take the DI feed as your close mic signal and then place a conventional mic at a distance from the speaker to add in as required.

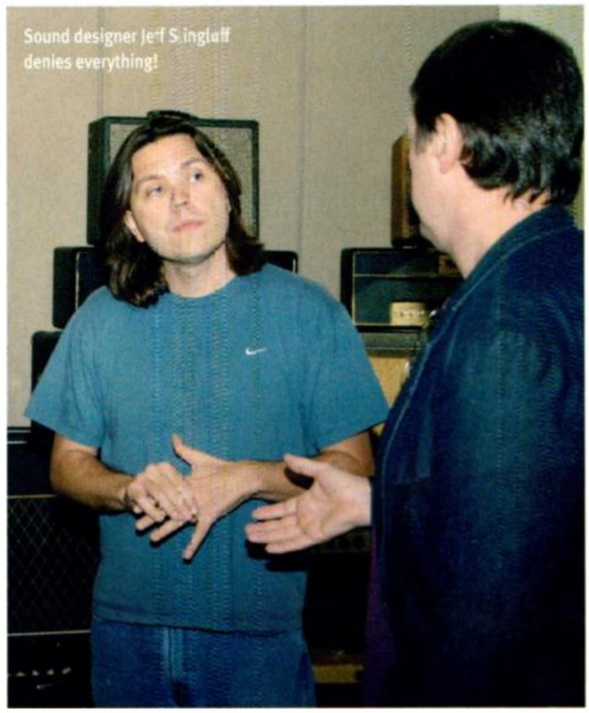
Marcus and Michel seem to have an unwritten code, so if you ask them a question and they start their reply with the word 'interesting', it means that what you have asked is probably already under active consideration, and that further probing won't get you too many answers! I discovered this when I turned my attention to Variax, which takes the modelling concept and applies it to guitars. Variax has already shown that it can effectively replicate the sounds of various classic electric and acoustic guitars and other stringed instruments. Variax uses a divided pickup arrangement, which is necessary for

Modelling Vintage Effects

One of my interests is how to capture the sound of 'classic' effects such as tape flanging and tape echo. Line 6 have already done a lot of work on tape-echo simulation, and Marcus explained that they feature vintage modelling in their new Echo Park pedal, which has switchable analogue, digital, and tape delay characteristics and is based on their popular DL4 delay processor. In the case of the analogue delay simulation, they've even captured the way the clock noise interacts with the audio, which is particularly important in recreating the sound of those old 'bucket-brigade' flanger boxes. The same approach was used in their Liqua Flange pedal, which has digital, analogue and liquid modes, the latter using two delay lines to allow the flange point to pass through zero time, as it does with tape flanging whenever one of the two tape machines overtakes the other. To try to recreate some of the randomness of tape flanging, they've also used a smoothed, random modulation generator, which can be used as an alternative to the more common modulation waveforms (there are 11 different modulation waveforms in all, including random steps). This attention to detail gives these pedals a much more organic sound than we've come to expect from conventional pedal effects and to my ears, the Liqua Flange's analogue mode gets closer to the elusive sound of the original Electric Mistress than anything I've heard before — but without the noise.

One of the corridor signs at Line 6 headquarters.


Paul White tries to sneak a look at a guitar pedal prototype. "No peeking, now..."


Sound designer Jeff Slingluff denies everything!

the accurate emulation of slanting pickups, and also allows a pitch-tracking algorithm to be used to optimise the operation of the pitch shifter used for 12-string sounds or for open tunings, which can be set up using the *Variax Workbench* software (of which more shortly).

This led me to wonder out loud whether Variax would be capable of producing some novel synthetic sounds based on the waveforms generated by the strings — after some heavy processing of course. After all, the ancient Roland GR300 guitar synth worked entirely from the string waveforms, which avoided all the tracking problems associated with most other guitar synths, and although its capabilities were extremely limited, it was still a very musical instrument. Using modern technology, wouldn't it be possible to use Variax to bring this concept up to date, especially given Marcus' interest in synthesizers? Indeed, if the six Variax string signals could be piped into a computer via USB or Firewire, it might even be possible to create software synthesizers driven from Variax. The one-word response to my ponderings was 'interesting!'.

Continuing on the guitar synthesis theme, I asked Marcus why they hadn't included a Roland-compatible output on the Variax, in order that anyone wanting to use a Roland-compatible guitar synthesizer could do so without adding a special pickup to their guitar. After all, the type of person interested in guitar synthesis would overlap with those interested in a technology instrument like Variax, and given that Variax already derives a separate output from each string, this would probably be relatively easy to do.

Their reply was that they didn't want to do this originally, because they wanted to distance Variax from sample-based technologies to avoid confusion. They needed guitarists to understand that Variax really was a guitar and that what you heard from it was the guitar strings themselves, not *samples* of guitar strings. However, now that Variax is established, my observation is apparently 'Interesting'...!

During my tour of the facility, I asked lead sound designer Jeff Slingluff whether the piezo pickups used in the Variax had a different dynamic response to traditional magnetic pickups. They're totally immune to electromagnetic pickup from monitors, transformers and mains power wiring, which eliminates the usual guitar hum problem, but don't they produce a different sound or playing feel?

Jeff explained that their modelling algorithms apply different decay times to different frequencies, so that what comes off the strings after processing matches the real thing very closely. In the case of the model that's clearly based on the behaviour of the Fender Stratocaster, they've even modelled the way the tremolo springs resonate in the rear body cavity, which adds a kind of low-level, coloured reverb to the sound. They start by modelling the pickups and their magnetic aperture, which combined with their position, affects how they pick up the sound from the strings, both singly and in combination. The pickup characteristics combined with the modelled body resonances produce the final emulation. Because the decay of different frequencies can be controlled, non-guitar instruments such as the banjo and sitar can also be recreated using only the output from the strings as their source. I asked Jeff if they deliberately modelled the effect of the magnetic string pull exerted by the pickups, as this also affects the way the note decays, and in some cases introduces atonal harmonics. He told me they hadn't done this specifically, but the way the pickups affect the decay rate of the different harmonics is automatically recorded in their measurements and so this aspect of string pull is modelled.

"*Variax Workbench* is intuitive to use and allows you to choose any pickup or pickup pair combinations and put them in any position on any of the available guitar bodies, or, indeed, position them part-way up the neck!"

Build Your Own

At a later product demonstration, I saw the *Variax Workbench* software being used, which allows the user to create guitar configurations that aren't available out of the box. Despite being very complex under the surface, *Variax Workbench* is intuitive to use and allows you to choose any pickup or pickup pair combinations and put them in any position on any of the available guitar bodies, or, indeed, position them part-way up the neck! The pickup angle can also be adjusted, and the sound updates almost instantly, so you can play as you tweak to see what works. It struck me at the time that if the graphics also provided dimensional data for the pickup placement and angle, *Variax Workbench* would make a very valuable design tool for custom guitar builders. To be truly effective in this role, it would need to include the ability to experiment with different coil taps and series/parallel wiring options, but wouldn't it be great to see how a Telecaster might sound with, for example, a Gibson P90 pickup in one position and a Rickenbacker pickup in another? In fact, my only regret was the limit of two pickups in the current version, because I wanted to try filling the entire space between the neck and bridge with pickups to see if it recreated that original *Spinal Tap* sound! *Variax Workbench* is free to owners of the Variax and either a Pod XT Live or a Vetta II.

Antiques Roadshow

A highlight of the facility tour was Jeff showing me around Line 6's vintage amplifier collection — a whole room full of them. These are not just any old examples either — they're all chosen for their sound, which leads onto an interesting aspect of modelling. Jeff explained that although you could model every single resistor and capacitor in the original circuit, it would take an enormous amount of DSP power and still might not produce the optimal result, purely because of the effect of component tolerances in the original. Most older amplifiers used resistors that had a 10-percent tolerance range or even greater, so depending on the cumulative effects of all these variations, you could get either great-sounding amplifiers or not so good ones. The valves are also very important. After Line 6 had purchased a great-sounding early Marshall amplifier that used KT66 output tubes, disaster struck when it was plugged in to do the modelling measurements. The whole thing erupted in a shower of sparks, and the entire set of output valves needed to be replaced. The original tubes were made by Mullard, and when the team tried to find a replacement set, they found them very hard to come by. According to Jeff, buying the new tubes was almost like conducting a drug deal, involving late-night meetings and bags of cash — and the four tubes they needed, though used, set them back a staggering $450 each!

While some modelling does go down to component level — and the exact details of the process are kept secret for obvious reasons — Jeff did tell me that they used fairly standard test equipment, such as Audio Precision test sets, to gain some of their data. They also need to look at the dynamic way in which amplifiers distort in response to transient sounds and also how things like power-supply sag affects the sound. Using this information, they can recreate the behaviour of the various key building blocks that comprise the signal path, and I was surprised at the lengths they go to. For example, when a power supply sags after hitting a loud power chord with the amp turned up, power supply ripple breaks through and modulates the audio to a small but significant degree. Line 6 have even replicated this so that the UK amps impose a 50Hz ripple under heavy drive conditions, whereas the US amps have 60Hz ripple!

Another thing that doesn't always come across with modelling is the way the speakers interact with a tube and transformer output stage, as a tube amp is effectively a current device whereas a traditional solid-state amplifier is a voltage device. To get around this in the current Flextone and Vetta amplifier ranges, they use solid-state power amplifier circuits that operate in current mode which, according to Jeff, gets much closer to the trouser-flapping experience of a real tube amplifier. The Pods don't have this, but then their aim is to replicate the sound you would hear over the control room monitors in a typical recording session involving a miked amplifier, not the sound you'd hear standing in front of a stack.

Currently the amps and Pods include user-adjustable noise gates (after the amp but before effects such as reverb and delay) to help clean up the noise, which is inevitable when a guitarist uses a lot of high-gain overdrive. I asked Jeff why they didn't use dynamic noise filters (sliding filters that progressively filter out the high end as the sound decays below a threshold) as I've always found these to be more benign-sounding on guitar tracks than gates. He told me they do use dynamic noise filters in the Variax guitars, and though he didn't say so, I got the impression they were considering them for future amp models or updates. I could swear he used the word 'Interesting'!

From Hardware To Software

Though Line 6 is primarily a hardware company, they have a strong software side, first aired in their Guitarport product (which strove to introduce guitarists to basic loop-composition and multitracking techniques), and now seen in the likes of *Variax Workbench* and *Gearbox*, the software that accompanies the new Toneport interfaces (shown below, and reviewed in last month's *SOS* — see www.soundonsound.com/sos/feb06/articles/toneport.htm). For me, these are the most exciting new Line 6 products.

The *Gearbox* software that accompanies the new Toneport interfaces.

Given their attractive pricing, they'd stand up on their own as practical USB audio interfaces, but what sets them apart from others is that the hardware is only a small part of the story — the *Gearbox* software is really the heart of the product. It's based on selected Pod XT amp, cabinet and effects models, and runs them on your Mac or PC computer. The interface hardware acts as the copy-protection dongle for the software, which can't be run without it. As you'd expect from a Line 6 product, *Gearbox* functions as an amp modelling plug-in, but Line 6 have cleverly developed what they call Tone Direct monitoring, so that *Gearbox* operates on the audio input before it ever gets to your audio application. The reason I say cleverly is that *Gearbox* itself has extremely low latency, and because it operates outside your host application, it isn't affected by the buffer size set for your audio-recording software. This means you can always hear your guitar in real time as you play, even if your audio software is set to a very high buffer size. Post-amp effects such as reverb and delay can be set to be recorded along with the guitar sound, or you can use them just for monitoring if you wish to add other plug-in effects after recording. *Gearbox* can also be upgraded to a full Pod XT spec or beyond by adding optional Model Packs from Line 6's web site, including all the Vetta amp models.

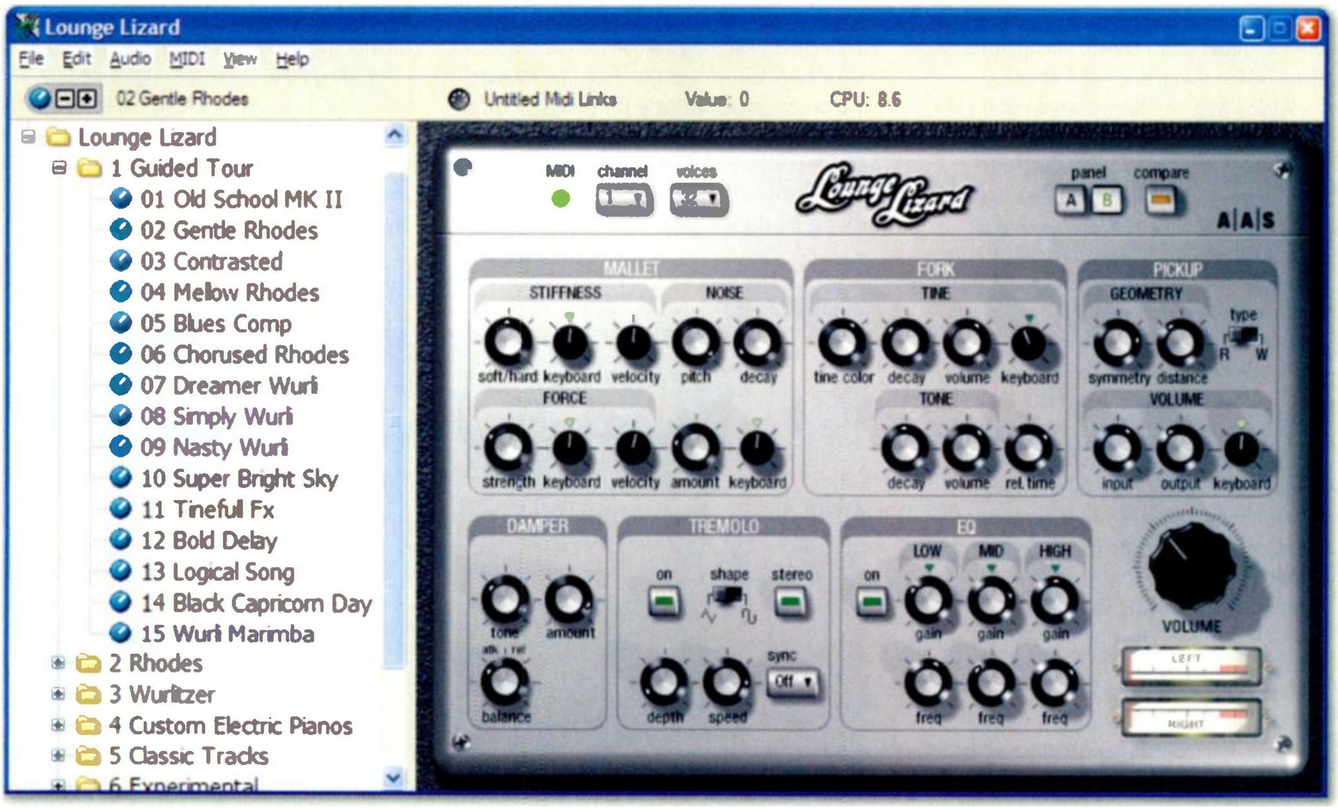
The package includes selected bass amp models taken from the Bass Pod XT and some brand-new vocal preamp models based on real-life hardware, some of them top-shelf classics (old and new), some lo-fi and some just plain quirky. These are augmented by new vocal compressor and de-esser modules. I asked Jeff Slingluff whether there was really that much difference between vocal preamps, and he replied that although they may measure similarly with steady test signals, their dynamic responses are often very different due to the use of audio transformers or time-dependent signal feedback paths in the circuitry. *Gearbox* lets you use a different signal path for your mic input via the vintage preamp models, compressors and reverbs, while recording your guitar at the same time via its own amp and effects chain. What's more, all the main effects can be used on guitar, bass or vocals, so if you need (say) a wah-wah vocal, you can have it.

Interesting *And* Friendly

Whatever they do, Line 6 want to make it musician friendly. At the time of my visit to Line 6, the Winter NAMM show was just around the corner, but nobody would confirm whether they would have anything new there to show us. Whatever may come next, I think it is safe to say that the future of Line 6 is going to be 'Interesting!'. SOS

Applied Acoustics Lounge Lizard EP3

Panel B contains the various modules that make up this unique physical string model and allow you to interact with its various parameters.



Modelled Electric Piano Instrument For Mac & PC

Most recent electric piano virtual instruments have been sample-based, but in *Lounge Lizard*, AAS continue to fly the flag for their physical modelling technology; version 3 offers an improved fork model, plus modelled electromagnetic and electrostatic pickups...

Martin Walker

Given the amount of praise that *Lounge Lizard* has already received for its accurate Fender Rhodes and Wurlitzer models from both artists and the press, you might expect AAS to rest on these laurels and move on to different projects (see *SOS* October 2002, or www.soundonsound.com/sos/oct02/articles/lounglizzard.asp for our original review of the instrument). However, the recently arrived *Lounge Lizard EP3* sets itself even higher standards in the company's search for physically modelled perfection.

At the heart of this third incarnation is the same physically modelled combination

SOUND ON SOUND

AAS *Lounge Lizard EP3* **£150**

pros

- Offers some of the most transparent and realistic virtual electric piano sounds available.
- Low CPU overhead considering the sophistication on offer.
- Excellent value for money (particularly for existing *Lounge Lizard* users).

cons

- Sounds from *Lounge Lizard 2* can't be imported into the new version.

summary

If you like electric piano sounds and have a computer, you have to buy *Lounge Lizard* — it's as simple as that!

of a mallet hitting a fork (comprising a tine and tone bar), amplified by a magnetic pickup. However, *EP3* has a new and improved Fork model, while the two pickup models now offer electromagnetic (Rhodes) and electrostatic (Wurlitzer) options for a much wider range of sounds, plus a keyboard scaling control, so that you can more easily balance levels at the bass and treble ends of the keyboard.

The Damper module provides more refined modelling of the noise of the dampers being applied to or released from the fork. The dampers now interact with the Fork parameters, and there's also a new Balance control that alters whether you hear the damper noise in the attack portion of the instrument, in the release phase, or in a mixture of the two. Finally, there's a completely new three-band semi-parametric EQ module to replace the more basic bass/treble controls of *Lounge Lizard 2*.

Like AAS's *String Studio*, (see *SOS* August 2005 or www.soundonsound.com/sos/aug05/articles/aasStringStudio.htm) *Lounge Lizard EP3* now has two switched front Panels (A and B) instead of one, which means that each one can be narrower, and AAS have taken advantage of this to add variable width to their preset Browser (just drag the divider between the two halves of the interface), so you can always read each preset name in full without horizontal scrolling (hooray!). The same reorganisation also results in the MIDI channel, polyphony, and activity controls moving to the main display from the Toolbar, so there's now enough space on the latter to display long preset names in full (hooray again!). Panel B houses the sound-generating components, and will be fairly recognisable to existing users. However, Panel A is ▶

The Bundled Library

AAS have put a lot more effort into the detail of their bundled library, with a 'Guided Tour' section at the top that quickly showcases the range of available sounds, followed by folders dedicated to classic Rhodes and Wurlitzer sounds, and a range of custom electric pianos that push the engine to more extreme tonal values and dynamics. Next up is a set of sounds which attempt to emulate electric piano sounds on a variety of classic tracks, and with names such as 'Dreamer', 'Riders On The Storm', and 'No Quarter', you should soon be able to work out the original sources of inspiration! The Experimental folder uses the engine to produce more unexpected sounds such as organs, vibes, bells, and chimes, with more extreme effects settings, and the collection is rounded up with two sets of signature sounds from Christian Halten (www.skylife.de) and yours truly. Here are some of my library favourites:

- **'Chorused Rhodes': there's a lot of Rhodes presets on offer, including 'Mellow', 'Tinefull', 'Tineless', 'Old School', and 'Fully Serviced', but this is my favourite, for the delicacy of its tine sound and the 'bark' when you dig in.**
- **'Chiff Bass': this offers a mellow mid range for chords, and rounded bass octaves with a 'chiffy' attack ideal for jazz workouts.**
- **'Dreamer': this cutting sound captures the Wurlitzer from the famous Supertramp track perfectly.**
- **'No Quarter': an atmospheric rendition of the watery phased piano from the classic Led Zeppelin track from *Houses Of The Holy*.**
- **'Nasty Electro 2': electric pianos don't always need to offer tones of scintillating beauty, and this one can be mellow when played quietly, yet cut through almost any mix when required.**
- **'Meditation Bells': with shimmering oriental overtones, this provides proof that the new *Lounge Lizard* engine can be pushed in more extreme directions.**

completely new, and its contents are discussed in the box on the right.

In Use

Lounge Lizard EP3 runs as a stand-alone application, and also supports VSTi, DXi, Audio Units and RTAS formats. I'm pleased to say that AAS have finally bowed to user demand and provided a 15-day grace period before you have to supply the response to the unique challenge generated after installation. This should go down well with all their users, especially those who have non-Internet-capable dual-boot setups and those who run into initial authorisation problems.

AAS have learned the value of a set of 'Guided Tour' presets from their popular *Ultra Analog* and *String Studio* instruments in quickly showcasing the strengths of new products. *Lounge Lizard 2* was already an impressive beast, but from the first moment you play one of the presets in the new version you can hear the engine improvements — there's a greater transparency of sound in the upper harmonics thanks to the new Fork model, while the greatly enhanced effects section adds further polish to each instrument.

CPU overhead is also surprisingly modest for a physically modelled instrument, and even my elderly 2.8GHz Pentium 4 Northwood processor managed to produce the maximum 32 notes of polyphony with CPU usage peaks of just 30 percent — some sampled electric pianos consume this without offering any of the subtlety and expression of *Lounge Lizard*.

I did miss the pitch-tracking of the mallet noise (which I've used in the past to create tuned percussion, organs, and vibes), although there's no denying that for more realistic electric pianos the mallet noise pitch shouldn't change across the keyboard — hopefully AAS will provide a tracking on/off switch in a future update for those of us who like to derail the engine in search of exotica.

The only other downside to the new engine is that because of its modifications and enhancements it can only import instruments in its own LX3 format, and not those from previous versions of *Lounge Lizard*. However, those who rely on existing instruments from the older version of the program can simply leave that installed alongside with no conflicts (as there are no samples in *Lounge Lizard*, the install size is only around 12MB).

As in previous versions, MIDI Links let you automate any combination of panel controls in real time using an external controller, but in *EP3* there's a folder of supplied examples that provide insight into what's possible, since apart from obvious links such as mod-wheel control of Tremolo and other effects, there are also various pickup-related links that control pickup symmetry, distance, and output level simultaneously in varying amounts and relative directions. These provide lots of performance options, and surprisingly, there's no 'zipper' noise except when changing the effects mix controls.

Final Thoughts

When the original *Lounge Lizard* came out, it was regarded as extremely clever, not to say handy if you didn't have a real electric piano to hand. By version 2 the heavy processor overhead had plummeted and the sound had become more refined, and various big-name players were beginning to regard it as more convenient, even if they did have the 'real thing' as well.

With *Lounge Lizard EP3*, I predict that many musicians may well desire it more than the acoustic instruments it models — it just feels so good to play, you forget within a few seconds that you're not playing a real acoustic instrument, and there are hundreds of variations and refinements available that don't involve a technician manhandling 73 tines and pickups! *Lounge Lizard EP3* is quite simply the new benchmark against which all electric piano sounds must be judged. SOS

Panel A

While the Tremolo section from *Lounge Lizard 2* remains unchanged on Panel B, its Wah and Phaser modules have now been incorporated into a much more flexible multi-effects section on Panel A comprising three modules. The first two are identical, offering a mono and stereo chorus and a flanger, plus other effects like vibrato, ping-pong, digital and tape delays, a phaser, auto and standard wah, a notch filter, and distortion, while the third is a versatile reverb with nine algorithms ranging from small rooms to large halls.

The reverb is always the final effect in the chain, but the other two modules can have their order swapped or be run in parallel, offering you some versatile new possibilities such as having simultaneous clean and distorted sounds, or complex multiple chorus or flanging effects. The Rate controls can either be set to manual or sync'ed to a VST host application, or to the new Clock module, which features manual or tap-tempo adjustment. There's even a tiny Padlock icon at top left of the entire multi-effects section to lock the effects settings, even when you're changing presets. Overall, the multi-effects are the icing on the cake, offering sophistication and flexibility.

Band members will be pleased to find a new keyboard module with global tuning and transpose functions, and there's even support for different tuning temperaments by loading in Scala-format micro-tuning files. One 'Concert Stretch' file is included (and you can download 3000 more from www.xs4all.nl/~huygensf/scala), or you can switch back to equal temperament and use the new Stretch control to manually create your own octave interval either side of the chosen reference note, from narrower to wider with a neutral position in the middle. Keyboard tuning module settings are independent of presets, though, and there are no reset or default settings (even between sessions), so take care.

Finally, there's a new Recorder module to capture your performances, plus various smaller improvements such as stereo instead of mono output-level meters.

Panel A contains a host of more traditional controls to fine-tune your sounds, including multi-effects.

Test Spec

- 2.8GHz Intel Pentium 4C PC with Hyperthreading, an Asus P4P800 Deluxe motherboard with an Intel 865PE chip set and an 800MHz Front Side Buss, running Windows XP (SP2) with 1GB of DDR400 RAM.
- Steinberg *Cubase SX* v3.0.2 & Cakewalk *Sonar 5*.

information

- **£** £149.99 including VAT; upgrade from version 2, $49 (on-line download only).
- **T** SCV London +44 (0)20 8418 0778.
- **F** +44 (0)20 8418 0624.
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Raxx System Wall-mounted Rack Hardware

Paul White

Free-standing studio racks are very useful and can hold a lot of heavy equipment, but sometimes it can be advantageous to make use of some of that spare wall space. That's where Raxx comes in, a modular system that can be used to fix rack mounting gear to any strong wall using two sturdy metal uprights to which side arms of varying lengths can be fitted to accommodate equipment of different depths. The available side arms are 1U or 2U high and have conventional rack fixing points on the front, while their rear flanges fix to the wall uprights by means of M6 cap-headed screws. The small starter kit reviewed here comprises a pair of 6U uprights and six 1U side arms, complete with fixings and M6 bolts. Additional side arms can be purchased if necessary in both 1U and 2U sizes and in lengths of 250, 400, and 600mm. Optional components include CD racks and half rack mounting trays.

When I first heard about this system, I was worried that it might struggle to hold heavier pieces of equipment, but as soon as I examined it, I realised such fears were without foundation. The components are manufactured from heavy-gauge steel with a durable paint coating, and all the fixing points are really solid. In fact the only limitation of the system is the purely practical one that you can't get around the back to plug things in and out, so you need to unscrew your units from the Raxx system before you can change the wiring around. You also need to have a strong wall to fix onto, because the installed system will only ever be as strong as the wall you're screwing into. Overall, though, this unique modular system is eminently practical. SOS

Sound On Sound

pros
- Makes the best use of available wall space.
- Very strong.
- Modular.

cons
- No easy access to the rear of the units once fitted to the rack.

summary

Raxx is a great little concept for situations where floor-standing racks may not be appropriate — as long as your walls can take the weight!

information

£ Starter Kit comprising two 6U uprights and six 1U 250mm side arms, £46.60 including VAT.
T MTR +44 (0)1923 234050.
W www.mtraudio.com

The ART T8 mounted in one of the six slots of the Raxx Starter Kit.

ART T8 Transformer Isolator

Paul White

The ART T8 is a simple 1U rackmount box containing eight transformers which you can use to break ground loops. As there is no active support circuitry required, there is no need for a power supply. Each of the channels is identical, and all inputs and outputs are available on balanced XLRs and TRS jacks, plus unbalanced RCA phonos. As supplied, the XLRs are on the rear panel, while the jacks and phonos are on the front, but you can undo four screws to reverse the rack ears if you want, bringing the XLRs to the front.

Technically, the signal path has a 10Hz-50kHz response, flat within ±0.5dB (measured with a +4dBu signal). Distortion is typically 0.01 percent at 1kHz, with an operating level of +18dBu and a crosstalk figure better than 90dB. How much signal is lost through the transformer depends on the load that each transformer is driving, and ranges from less than half a decibel into 100kΩ to 5.5dB when driving 600Ω — I found insertion losses for typical studio line-level inputs to be mostly unnoticeable. A CMRR (Common Mode Rejection Ratio) of 60dB is quoted for the balanced inputs. The balanced connections of the T8 can be used with unbalanced connectors where required, but it is invariably better to use balanced connections where possible. In a typical setup, unbalanced electronic instruments will feed the jack inputs, while balanced jack or XLR cables carry the outputs of the T8 to the mixer or audio interface.

In practice the T8 sounded surprisingly transparent, and the only very very subtle difference that I could detect was that the bass end actually seemed slightly more solid via the T8! I find it remarkable that such a low-cost transformer-isolation system can work so effectively while retaining a commendably high degree of audio transparency. I have no doubt that more sophisticated transformers can provide a better technical specification, but I'm equally convinced that any audio deterioration produced here will be far less of a concern than the quality of the output electronics in much budget studio gear. SOS

Sound On Sound

pros
- Clean audio path.
- Full transformer isolation.
- Choice of connectors.

cons
- None at the price.

summary

Eight channels of effective and clean-sounding transformer isolation at an affordable UK price. A very practical solution to ground-loop problems with unbalanced equipment.

information

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NOTHING ADDED. NOTHING TAKEN AWAY.

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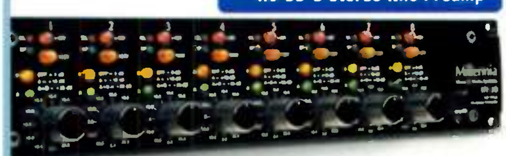

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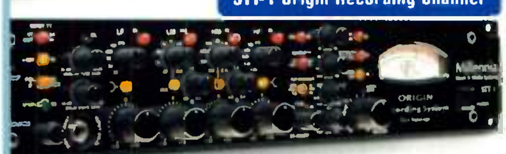
STT-1 Origin Recording Channel



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Sound & Vision

Making A Living From Music For Picture: Part 4

Hilgrove Kenrick

Success! You've been invited to pitch for a job. Surely, that Hollywood career beckons? Well, no. This is just a pitch — there are others competing for the job. We look at how to maximise your chances of winning the commission...

By the time you've finished reading this part of this series, I hope that you'll have a clearer idea of just how nerve-frazzling life as media composer can be, how you should go about putting together a showreel to attract work, and what sort of issues you should consider when adapting or buying a studio setup so you can work as a media composer. This month, I'll sign off by considering what to do when your showreel finally produces a positive result, and you are offered the opportunity to pitch for a job.

Keeping It Brief

The process of pitching, in my experience, often starts with a phone call, email, or letter. "Got your showreel, we'd like you to try for the job, now show us what you can do, as soon as you can — will tomorrow sometime be OK?" seems to be the essential content of most of these communications. Budget is rarely discussed, and if you should ever broach the subject, you'll usually be met with a reply along these lines: "Ah... yes... that's a bit up in the air at the moment, actually...". Don't expect more than this, because in my experience, that's as firm a commitment as you're likely to get. And... it's not very firm, is it?

Of course, before you can pitch, you need to know something about what the client wants in terms of music, but again, don't expect this to be given to you routinely. Those commissioning music for picture frequently don't know what they want in musical terms, and you may be offered no specific guidance on the music whatsoever. What you should get, though, is what's known as a brief, possibly from the

Understanding The Brief

As with any close-knit trade, music for picture has its own hard-to-understand jargon, and when you're establishing yourself, it can be tricky to understand what directors are asking you for. Here, I've explained three terms which you will hear used endlessly, and which, at first, utterly baffled me.

STINGS

These are short bursts of music. Sound effects are often also classed as stings (a door slam for example), but in musical terms, a sting can be as much as 10 seconds or so. A classic example is *Who Wants To Be A Millionaire*, and the little ditty that plays when a contestant gets a question right (or wrong). A sting often incorporates a statement of a theme, or simply functions to accentuate the beginning or end of a section of underscore. And if you're wondering what that is...

UNDERSCORE

Sometimes referred to as a bed, the simplest description of underscore is when you use music in a scene but don't want to hear it! Don't be fooled into thinking you can just lower the volume of a busy piece of music, either. It's a balance between giving the impression that something is happening without getting in the way. Thinking back to *Who Wants To Be A Millionaire* again, the continual music during the actual game play could be classed as underscore; it's ever-present and greatly contributes to the atmosphere, while never getting in the way of the host asking questions or the player answering.

THEMES

Well, all right, this one is pretty self-explanatory, but it deserves a bit of coverage. Whilst many composers think automatically of melodies when they're asked to produce themes, you shouldn't feel constrained in this way with music for picture. A 'theme' could be rhythmic and not in the slightest bit melodious, or could even be a sequence of chords with no obvious tune to it. Personally, I try to find my themes, in whatever form they may take, whilst pitching — after all, you can restate a theme in so many different ways, be it with differing orchestration, tempo or genre. Once you've come up with something, it's a great building block around which to base the rest of the score if the job comes your way. Sometimes, no theme shows itself, and you're left with a mish-mash of ideas. If that's the case, fret not. If you get the job, you can always go theme-hunting later — as long as you're quick!

director or even one of the producers (more on who these people are and how they relate to you, and the project, in a moment). The brief is supposed to be a description of what the project is all about and where you fit in — how much music they want, any special concepts or crazy schemes they have come up with on the back of a fag packet, and maybe, just maybe, something vague about the musical styles or genres they're hoping for.

Now comes the complicated bit, as you need to either follow such advice as the brief does contain to the letter, or bin it and do what you think is right. Both of these routes have their own advantages and pitfalls, and the reality is that you'll probably do a bit of each to try and please all the execs at the same time as satisfying your artistic cravings!

The biggest complication here is musical terminology and description — to me, R&B still means Otis Redding or Ray Charles; but to someone else it might be R Kelly, or whatever her name is [*shurely some mistake? — Ed*]. Furthermore, you must walk a narrow tightrope between what directors say they want, and what they *actually* want. The trick here is being able to work that out, describe it in terms they understand, and then deliver it. Being able to hum it, play it or describe it in words of less than one syllable is crucial; it's that or they'll laugh you out of town, or give the job to someone else!

Why consider all these complicated factors, and do all of this complex psychological second-guessing, before going through the process of pitching? Well, simply because you need to know what you're pitching for and what is being aksed of you — after all, if you produce the most awesome Aretha Franklin clone when the execs were actually thinking of Beyoncé Knowles, then your pitch is probably doomed!

The Pecking Order

First things first — who will you be talking to? TV programmes and films have producers, directors, script editors, production staff, secretaries, runners — can you tell them all apart? Fortunately, the hierarchy is quite simple. As the man doing the music, you're the lowest of the low in terms of almost everyone's priorities. While this may be depressing, it also lends the situation a certain clarity, and means there's only a few people who you absolutely must be able to recognise and work with.

THE EXECUTIVE PRODUCER(S)

These guys are the *grand fromages*; without their say-so, thou shalt not pass go, nor collect £200 on the way. They answer only to the Shareholders, Broadcast Commissioners or their gods, in that order. An executive producer is the one person in the world you would really like to do without (yes, even more so than your mother-in-law) but, inevitably, is the one that will never, ever go away. Upset them at your peril.

THE PRODUCER

Producers are the *demi-fromages*; they live and breathe the project from the day it's born until the day of transmission/general release, and then they'll be off elsewhere. Don't ever talk about the next project or the last, as all they can see is the leviathan they're struggling with. The buck stops with them, and everything has to pass through them. If they ask you a question, be ready with an instant answer, and if they speak, listen. Upset them, and you'll be out on your ear before a crotchet rest has elapsed.

THE DIRECTOR

Finally, someone friendly! The director is your best buddy; the project will be his baby, and he will want it to have a good, wholesome arty upbringing. This is the person who will buy you a pint (or 17) and talk your ears off about that clever tilt and pan move he did for the Organic Offal Monthly documentary on the Shopping Channel. Directors are also the only people on a music-for-picture project that talk any artistic sense. Upset them, and they'll probably just curl up into a ball and cry, but remember that everything you plan together has to be signed off from above.

EVERYONE ELSE

This group will differ from project to project, may include a production manager, someone (or everyone's) PA, associate producers (nobodies with vested interests), camera operators (great for cribbing cigarettes), editors, soundies, sparkies... the list goes on.

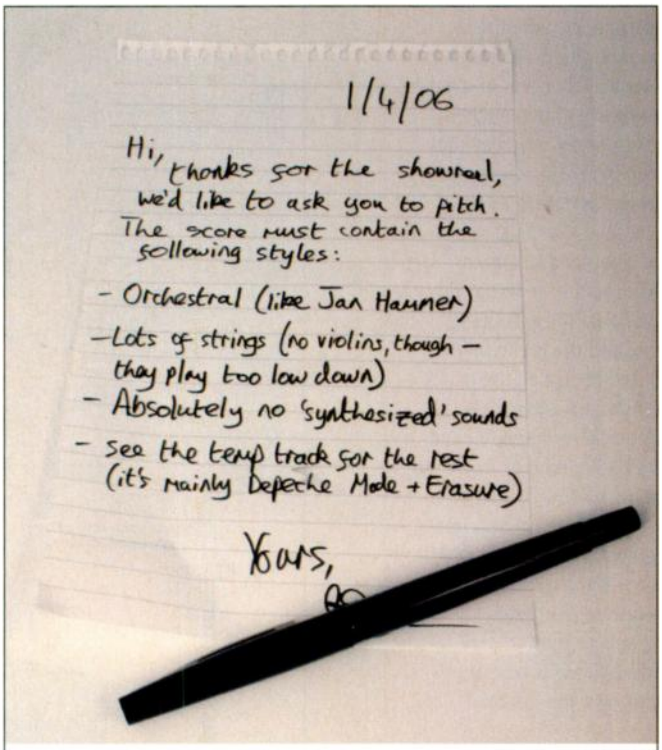
YOU

So why, you ask, are you at the bottom of the pile? Well, you were probably chosen last, you have a creative input when everyone above you wants complete creative control, and you cost a fair bit of money. Know thy place!

The thing to remember is that whilst in most cases the director has the creative say-so over a TV or film project, the likelihood is that the producer (or producers) will be able to override him — so the director can wish for the moon on a stick, but the producer may well tell him he'll have to make do with a lightbulb (and no lolly). The complicated bit can be juggling them both to keep them simultaneously happy.

Perfect Pitch

So on we come to the pitch itself. The invite will be delivered into your hands in one of two forms, depending on how advanced the production is; remember that music is usually bottom of the pile in terms of decision-making, so more often than not it is the last thing considered. If you're in early, it'll just be a brief. The chances are the brief will be a simple description of the project and, at best, a list of vague adjectives which probably encompass every musical style you have ever heard of, and several that you haven't. As I've outlined already, these may well be contradictory and/or not what the production team *really* want at all (see above for an example), so it's down to you to interpret them as you see fit. I was involved in one pitch last year which featured the words 'orchestral', 'epic', 'ethnic', 'big' and 'Hollywood-esque' (oh, how they all love that one!). So I did my best brass-belting, cymbal-crashing, timpani-thumping Hollywood score impression (which is pretty damn good, if I say so myself). And the result? Not a multisampled sausage; the job went to another guy. A month or so later, I watched the finished program and nearly died laughing — the principal component of the score was a gently pulsing synthesizer. It was great, but it certainly wasn't what the brief had described!



OK, so this (admittedly fictitious) brief may seem silly — but believe me, it's been heavily inspired by dafter, more self-contradictory ones I've experienced in real life. And if you think no self-respecting media mogul would send you a written communication on tatty notepaper like this, think again — I had an invitation to pitch and a brief once that had been scrawled on the blank bits of a postcard of Brighton. And when I'd finished with it, I had to give it back, as the producer concerned had written another important person's phone number on the other side of the card...

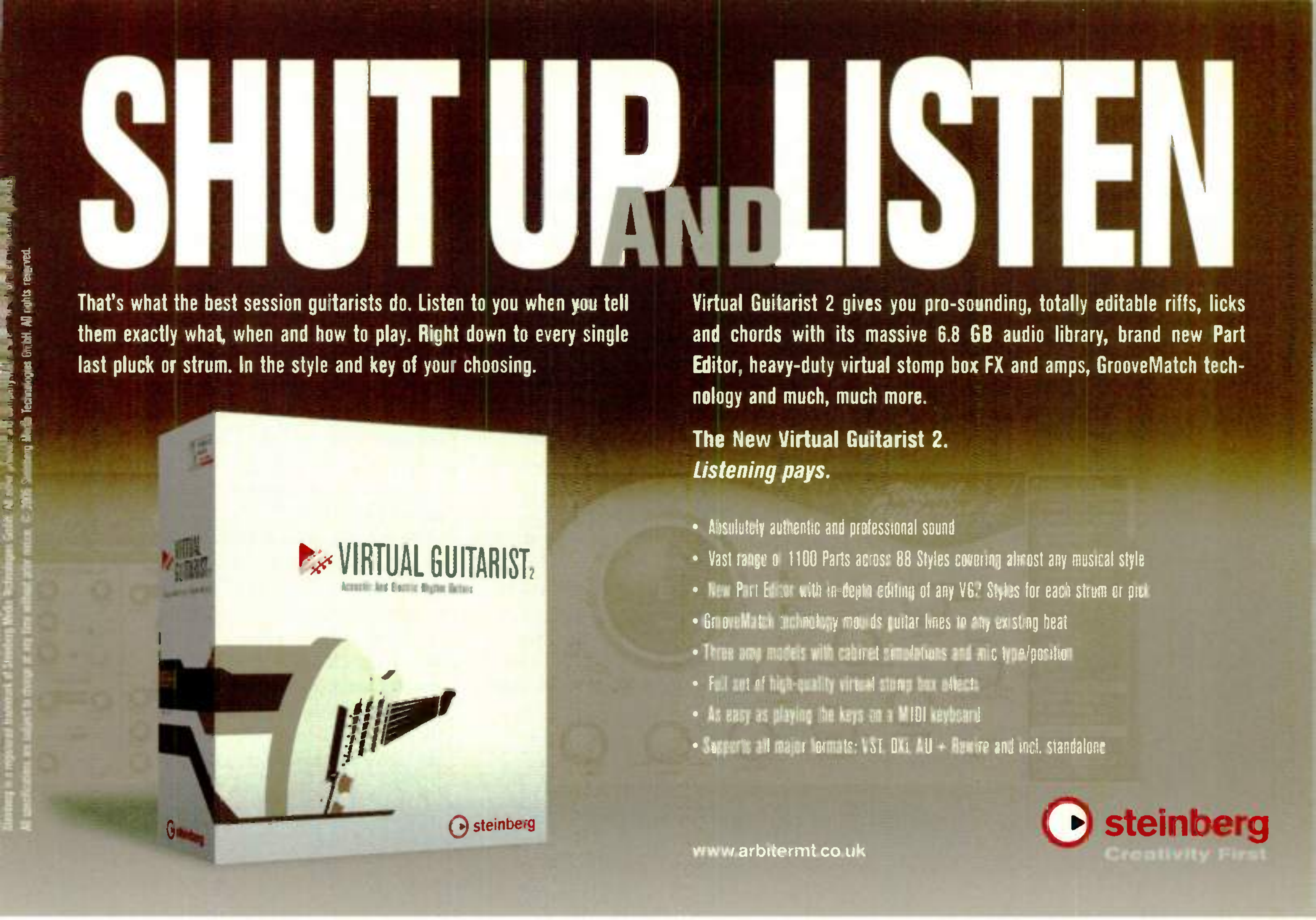
The other possibility is when there is a completed draft edit of the program. This is likely to have a 'temp track' dubbed on to it. As the title implies, this is temporary music laid over to help to guide the editing process, and also to reveal the plans of the execs or director for the style of music, and roughly where it should start or end. This will probably be attached to the same sort of vague brief as those that arrive with no video. This temp track can be either your saviour or downfall; you ignore it at your peril, but if you follow it too closely, you'll probably find that it wasn't what they wanted at all. Try to take it *and* leave it — carefully analyse what works well in the temp and what falls down, then extrapolate from the good bits.

Now you have a brief, and, hopefully, a deadline for pitching. I say hopefully, as if it's an open-ended invitation, someone is probably having a laugh at your expense! That deadline is *sacrosanct* — almost more so than a production one. After all, in the world of normal work, your boss may well haul you over the coals if you're late for a day at work, but if you're late for the interview, you probably won't get the job in the first place.

Fire up the computer and turn off your phone — this is going to be the best music you have written in your entire life (again!). Once you're on the production, you can go easy on yourself (OK, that's a lie, but it helps to think this way), but this time it has to be all or nothing. No duff notes, no glaring production errors, no dodgy General MIDI samples; only the best will do.

Getting It Right

How do you go about producing the pitch? First, let us assume that all you get is a few scribbled notes, and you have to construct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the score and a style from scratch with only those notes to base it on. Once you've decided on that style — whether you follow such information as you can derive from the pitch, or throw it out and choose something competely different — you must stick to it. Once this is done, you can think of the pitch as a glorified showreel — about six minutes long and showcasing the very best of what you can do. The fundamental difference this time is that it will all adhere to the one very specific basic style that you've decided to adopt.

Within that five or six minutes, and your chosen basic style, you must somehow illustrate every moment, every emotion called for in the brief. If you know there will be a car chase, you need a chase theme, if you know there will be a funeral, you need to stir the emotions, and if you know there's a love scene, you have to stir... other things. I cannot stress enough that quality is paramount over quantity. Don't go shoving 78 key and tempo changes in alongside 14 styles; make sure that what is on there is *exactly* what you think they're asking for. You'd be far better served by offering them a theme (stated in a couple of different styles or tempos), a couple of variations and a rousing ending rather than literally ticking off an exhaustive list, if one was provided in the brief. Go for a big opening, then slow it down and show that you can call forth slow and tender sentiments — and then puncture their ear drums with a soaring finale, or whatever variation of that principle the brief calls for.

If you've received a draft edit of the project to work to, you'll probably have a temp track to base your work around. The question is: do you plagiarise it wholesale, pinch the odd idea from it, or ignore it completely? As ever, the answer is a blend of each; with a bit of luck, you may have an idea (or be able to work out) why they chose that music for the temp track in particular. Was it in the nearest CD case along the shelf, or was there some specific idea in mind? If it's standard library fare, then the chances are it was simply what was lying around. If, on the other hand, it consists of individual popular or classical tracks, they may well have been chosen very specifically. If you're careful, you may well start spotting reasons why they've chosen the temp track — the instruments and orchestration may seem wrong initially, and fly in the face of the brief, but sometimes, if you step back and listen again, you may see that the mood or atmosphere is spot on nonetheless. If so, find it, extract the underlying theme or idea, and work with that.

If the material has arrived as a draft edit, then you have one other option — you could pretend you've already got the job and score straight to picture. This is a tricky decision, as you may have very little time in which to deliver, and it's a more complicated way to proceed, but if you can hit the mark *and* get it to the production team on time, you may just score a double whammy, proving that you can work accurately (and very quickly),

Further Education

There are many education courses run by colleges and private companies that claim to be able to ready you for the mad world of media composerdom, but in my experience, sorting the wheat from the chaff is a difficult process in itself. The best courses, in my opinion, are those that concern themselves with

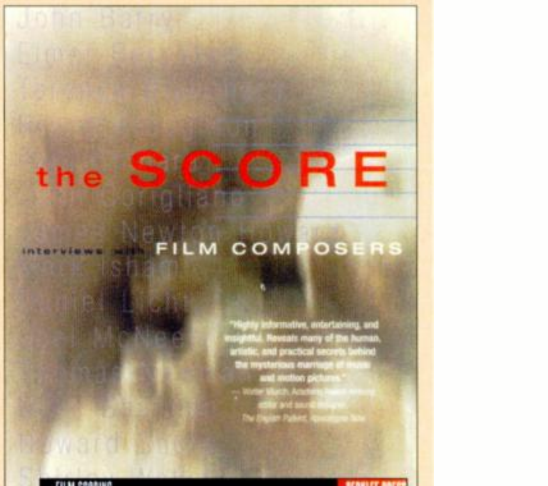
practical technique: the actual doing rather than the theory. One such that I can heartily recommend, having had personal experience of it, is Music for the Media (see www.musicforthemedia.co.uk), the course tutored by Emmy Award winner Guy Michelmore, along with a crowd of other experienced composers. The chosen approach of this course is deeply practical, with a range of modules covering writing techniques for everything from corporate videos to film projects. It's a great place to start, or even to hone your existing skills.

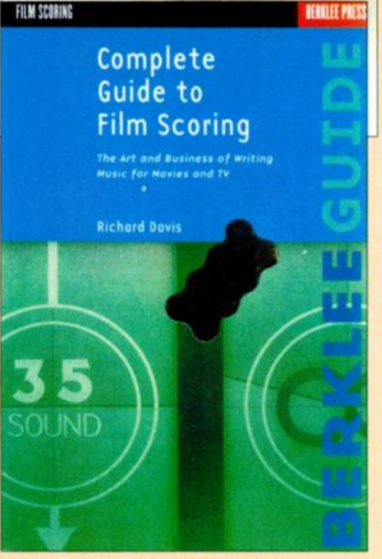
If you'd rather gain useful background by reading in your own time, there are plenty of books which offer a deeper insight or analysis of the art of music for picture. Some are weighty tomes that dissect existing scores and explain how they were constructed in detail, while others offer wildly differing advice on how to make millions in your first month or two. Beware of US books, whose advice is not always applicable to the business as it is here in the UK, and *vice versa*. There are also some older books that are interesting but dreadfully out of date in terms of actual technique and technology! On the right are a couple that I think are worth mentioning.

THE SCORE
This book (right), by Michael Schelle, is a very easy read, and features interviews with many film composers, including John Barry, James Newton Howard and Howard Shore. It's a captivating book that will both inspire and confuse, as the outspoken interviewees frequently offer conflicting advice on the best way to do things — but that can make it all the more interesting!

COMPLETE GUIDE TO FILM SCORING
This book by Richard Davis is again very easy to work through, and offers some very practical advice. Just don't get too hooked in to the fees, royalties and copyright issues described, as they differ from country to country.

THE REEL WORLD
Jeff Rona's book is very accessible, and utterly practical. This is one of my personal favourites, although again, not all the business advice given holds true on the UK market.





and also giving the team something they can drop straight into their edit and listen to it as you intended, locked to picture. The biggest drawback of this approach is the margin for error — or rather the lack of it. Mess it up, miss the hit points, deliver locked music that actually doesn't lock, and you will well and truly have blown your chances.

Now I'm going to raise the ire of the audiophiles (as well as the Editor In Chief of this magazine) by daring to suggest, for the second time in this series, that you master your completed pitch to within an inch of its life before you send it. I know it's not the done thing (and you certainly shouldn't do it when you're delivering your finished score), but in the same way that you wanted your showreel to grab prospective clients by the neck, this time, while you have their attention, you don't want to lose it — so make it loud.

One final thought on the pitch. How much time do you have to spare? There is never any better time to get some extra practice if you can fit it all in. Thus, if you can supply several pitches at once, then so much the better — again, as with the showreel, don't provide an hour or two of music, as it'll never get listened to, but two or three renditions in different interpretations of the brief are no bad thing. Don't go overboard, and whatever you do, never compromise quality for the sake of quantity. You're better off proving your genius with one polished track that misses the mark than with several that come close but are a little ragged.

Finally, I must warn you that it's not uncommon for you to be asked to pitch for a job even though another composer has long since been selected. In fact, the lucky winner may have already started work. This may seem most unfair (and from the composer's perspective, it is), but in the commissioning world, it's seen as sensibly 'keeping options open', and nothing you can do will ever prevent it happening.

"I must warn you that it's not uncommon for you to be asked to pitch for a job even though another composer has long since been selected."

Next Month

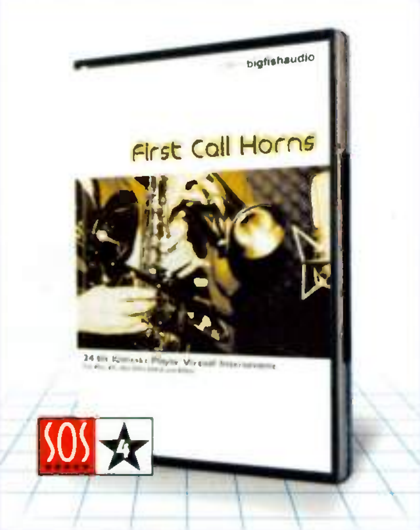
And on that downbeat note, I'll take my leave. If you think that's miserablist, tough — that's what this job's like sometimes, which, of course, is why I tell so many people not to bother trying their luck at it. I hope that if you've read this far, you have a pretty good idea of the trials and tribulations that await you if you stubbornly pursue this course. If your mind *is* made up, then stop reading this series and spend the next 30 or so days using your nice shiny new equipment to bash your showreel into something you're proud of. In next month's instalment of this series, established American media composer Bill Lacey will take over the reins, and tackle the complex issues of how to conduct yourself if you actually *win* your pitch... SOS

Big Fish Audio First Call Horns

Gigastudio/Kontakt

Sampled pop horns were popular during the '80s and '90s, but recently orchestral brass libraries have dominated the market, leaving those needing pop, rock, and jazz brass samples out in the cold. Big Fish Audio aim to raise the temperature of these poor shivering souls by releasing the defiantly pop-orientated *First Call Horns*. This 1.5GB library's dry studio acoustic can be sweetened by adding a splash of *Kontakt*'s built-in reverb. Most instruments come in a choice of solo or stereo section formats, the latter (which sound like unison duos) producing a lusher, broader sound than the mono solo instruments. None of the samples are looped.

Pop brass needs good saxophone samples, and Big Fish's baritone sax is the star of the show — big, honky, and in your face, perfect for dirty low-end riffs or for punctuating Tamla



bass lines. The alto sax's intimate, breathy quiet notes are also particularly attractive. On the minus side, there's a big, jarringly obvious change in timbre between the alto and tenor saxes' quiet and loud notes, and the soprano and baritone instruments are single dynamic. However, being able to keyswitch instantly between vibrato and non-vibrato notes restores some expressive potential.

The solo trumpet's large array of attention-grabbing big-band mannerisms (long and short falls, 'doits', leery shakes, vicious razor-sharp staccato stabs, and so on) provides a stark contrast to the inhibited delivery of most orchestral players. The use of Harmon, cup, and plunger mutes takes us deeper into Miles/Louis Armstrong jazz territory. If you want subtlety, the mellow-sounding flugelhorn and stopped French horn employ a tasteful, understated vibrato, but if you don't, the trombone's vibrato notes are satisfyingly over the top. One nice, unexpected extra is the no-frills piccolo trumpet, which belts out some good, pure-sounding loud sustains.

The Improvs & Riffs section of the library is a big selling point, boasting over 1000 jazzy solo performances played by trumpet and trombone (with and without mutes), flugelhorn, soprano, and tenor sax. The blowing is fluid, free, inventive, and highly varied, featuring bebop-style lines, wasp-in-a-bottle note flurries, extended melodies, melancholic soft phrases, screaming high notes, and a few effects. No keys or tempos are listed, but with a lot of patience and a little retuning, this gigantic grab-bag of horn improvisations will yield great musical results.

First Call Horns' instruments work together to create a realistic, versatile pop/rock brass-section sound. The samples are less well suited to replicating the subtle inflections, tonal detail, and smooth legato transitions of an expressive solo melody line — that requires a more forensic sampling approach. But if you're in the market for fat, powerful brass chord pads, swells, stabs, and falls, as well as a tremendous stock of jazzy licks, you could do a lot worse than this title, available now from your local Big Fishmonger. *Dave Stewart*

***Kontakt* Instrument or *Gigastudio* DVD-ROM, £165 including VAT.**
Time + Space +44 (0)1837 55200
www.timespace.com www.bigfishaudio.com

Best Service Galaxy Steinway 5.1

Kompakt Instrument

This 7.8GB piano library from Best Service features a Steinway model 'D' recorded in Belgium's top-notch Galaxy Studios and presented in both 16- and 24-bit versions. Release and 'resonance' samples (of which more in a moment) are included, and the instrument can output in a variety of multi-channel formats including 5.1 surround.

Let's cut to the chase — the piano sounds great. Packing up to eleven dynamic layers under its gleaming black bonnet, and sampled at tone intervals, it 'speaks' clearly and precisely while retaining a subtle lyrical quality. Dynamic transitions remain smooth right across its seven-octave range; the beautifully transparent high notes are responsive to the most delicate of touches and the bass notes can deliver weight and power. Note-off realism is enhanced by a set of newly minted, well-programmed release samples. Classic sounding without being too 'steely' or overbearing, the instrument is subtle and expressive enough for jazz and classical music, and strong and bright enough to cut it in a pop mix. No sampled piano could ever replace a real grand, but this one does a commendable imitation at less than one percent of the price!

Reproducing the effect of a piano's sustain pedal is a conundrum for samplists; Best Service tackle the problem by supplying a 'resonance' sample (consisting of the sympathetic overtones of other undamped strings) for each note. Recorded at one dynamic only, these ethereal samples remain silent until you hold down notes and press the pedal, at which point they fade quickly into the mix — the resulting drifty, reverb-like wash is extremely pleasant and inspiring for the player, but isn't entirely lifelike.

On to the big selling point: 5.1 surround capability. There are six-channel multis combining front left and right (close mics), rear left and right (ambient room mics), centre (mono mix), and LFE (low-frequency mono mix) presentations. These multis are highly CPU-intensive, but you can reduce the strain by using just the front and rear mikings, which still feels like being in the hall with the piano. Alternatively, you can stick with stereo by using only the front mics.

Surround compatibility is all very well, but there are other, more important considerations: is it a superior, well-maintained instrument? Is it well-tuned? Is the room acoustic good? Was it miked sensibly? Does it play well over MIDI? In the case of *Galaxy Steinway 5.1*, the answer to all these questions is yes, leaving me with no alternative but to recommend it highly. However, you should bear in mind that judging piano samples is a subjective process which depends on the player's experience, expectations, taste, and touch, so do check out the on-line demos before parting with your cash! *Dave Stewart*

***Kompakt* Instrument, £166 including VAT.**
Time + Space +44 (0)1837 55200 www.timespace.com www.bestservice.de

Big Fish Audio
London Solo Strings

Gigastudio/Kontakt Instrument

Originally released in 2004 in 16-bit *Gigastudio 2* format, this title is now newly available as a 24-bit, 3.75GB *Kontakt* Instrument. Big Fish Audio won't say whether the samples were actually recorded in London, but the library does feature reverb-free, high-quality studio recordings of solo violin, viola, cello, and double bass playing a wide range of deliveries including tremolo, pizzicatos, trills, harmonics, crescendos, sul tasto, sul ponticello, and bow ricochets.

The solo violin's sustains have a tasteful, sweet vibrato which reaches full effect after about a second. The player also puts a slight volume fade on the front of each note, but while this expressive style sounds fine in isolation it doesn't work very well for legato melodies. I tried the old trick of layering short notes over the sustains to add some attack, but the effect wasn't entirely convincing. The lack of straight sustained notes with a fast attack is arguably the library's greatest weakness.

Matching the violin's expressive delivery, the viola and cello sound lovely on very slow-moving passages, while the double bass's brusque, no-nonsense attack copes better with faster lines. I really enjoyed the sound of the double bass — big, authoritative, and with a lot of character. Its pizzicatos have a very nice round, mellow timbre and great tuning, and work equally well for orchestral and jazz styles.

The library includes no grace notes or col legno bow hits, but I did find a few fast chromatic scales and some great slides lurking undocumented in the effects section. Unusually, there are no staccato samples; short notes come in the form of marcatos (which the viola plays rather languidly) and spiccatos, both supplied in separate up- and down-bow versions. The most emphatic short-note style is the violin and cello's martelé (hammered) delivery. Dynamics are a strong point: the main articulations use three dynamic layers, and all instruments play excellent, strong crescendos of three different lengths. There are also a large number of programs with mod wheel control, useful for subtle swells and tonal shadings.

No samples have been looped; durations are on the short side, with sustained notes lasting between five and seven seconds and the violin's harmonics expiring after barely a second. However, the lack of loops does allow the players' vibrato to develop naturally from start to finish, while the short note durations guarantee that any feelings of fear and suspense generated by the string tremolos will be mercifully short-lived!

Although *London Solo Strings* lacks certain contemporary features like release samples or a monophonic legato mode, it covers a lot of musical ground in depth and boasts a sound quality second to none. The pricing is reasonable, and composers will appreciate this pragmatic set's ability to do its job efficiently with minimum setup time. *Dave Stewart*

***Kontakt* Instrument or *Gigastudio* v2 DVD-ROM, £215 including VAT.**
Time + Space +44 (0)1837 55200
www.timespace.com www.bigfishaudio.com

Tascam Larry Seyer Acoustic Drums *Gigastudio*

This title is a collaboration between US producer Larry Seyer and drummer Pat Mastelotto. Both have impressive CVs: Larry's recording projects occupy several sheets of foolscap, while ex-Mister Mister Mr. Mastelotto played on XTC's *Oranges & Lemons* and shared King Crimson drumming duties with Bill Bruford. *Acoustic Drums* features some pretty intense sampling: each snare-drum head was divided into eight zones, which were struck twice at five different dynamics with drum sticks, hot rods, and mallets, employing straight hits, rim shots, and 'crushes' (short press rolls), both with and without damping. Applying this to eight snares, six kicks, four toms, ten hi-hats and 40 cymbals has generated 7.36GB samples.

The snare drums all come across as well-recorded, classic rock snares. My favourite is the Ayotte model: precise, big, and beefy with just the right amount of snare buzz. A Montineri snare constructed from surgical steel (good for heavy metal?) has a harder, more aggressive attack, while the lower-pitched tones of the Slingerland and Sonor models work well for old-school funk. There's also a menu of tasty cross-stick samples for soulful ballads.

The kick drums range from the traditional pop/rock dry thud to more open, sustaining hits better suited to jazz and acoustic settings. Tuned quite deep, the library's set of Yamaha toms sound big and resonant. Played with mallets, they resemble timpani — a fabulous sound. Cymbal stand-outs include ear-catching cup chimes and splashes, rivet rides, China crashes, and mallet rolls. My only criticism concerns the hi-hats. Although it's handy to use the mod wheel to switch between tight and loose hits, I would have preferred the tight samples to be the default set.

As well as offering these elements separately, the library combines them into 115 very usable kits — these sound very good right out of the box, and the well-designed, modular programming system makes it easy to substitute a component (say, the kick). Reduced, two-octave General MIDI versions of all kits are also provided, along with 100 MIDI files covering a variety of styles.

Thirteen excellent 'room colours' for the *Gigapulse* reverb are included, fashioned in two-, five- and seven-channel configurations. The rooms range in size from a beach hut to the Taj Mahal, and feature some enjoyable 'trash' micro-spaces which sound like the samples are going through a Martian transistor radio.

If you need a solid and varied collection of acoustic drum-kit sounds, they don't come any better than this title. Despite the fact that there are no sustained drum rolls, licks, or brush hits, this set is a programmer's dream. Congratulations to all concerned (including programmer Stephen Orsak) for this great-sounding, versatile, and dynamically responsive library. *Dave Stewart*

***Gigastudio* v3.1 DVD-ROM set, £259 including VAT.**
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Creating A Drum Editor In Logic

Some users of Steinberg's *Cubase* complain that *Logic* has no specialised drum editor. Little do they know that *Logic* lets you actually build your own.

Stephen Bennett

To a new user, *Logic* can often seem to be packed with mysterious and arcane features. As time has passed, Apple have simplified the program and made certain features seemingly redundant and only apparently retained for backwards compatibility. One of these features is the Hyper Edit window — I've met quite experienced *Logic* users who have never even opened this window. Another is the Mapped Instrument object from the Environment, and some *Logic* users have never looked in the Environment at all! Combining these two features rectifies a common criticism of *Logic* often made by those who use, or have recently moved from, *Cubase*, namely that *Logic* doesn't have a dedicated drum editor. So this article describes how to create one in *Logic*.

A specialised drum editor can be really useful especially if you are a 'mouse' drummer who prefers to enter your percussive data with your finger rather than using sticks or the keyboard. It can also be a boon to those who enter drum data with a keyboard or specialised percussion controller, as it makes it easier to visualise previously recorded data on an on-screen grid. It's particularly useful for those wanting to emulate such systems as Page R on the Fairlight CMI, and it's also similar to, though more flexible than, the step sequencer in *Ultrabeat*. As with all *Logic*'s editing windows, data recorded here can be modified elsewhere.

Current Versions

Mac OS X: Apple *Logic Pro* v7.1.1
Mac OS 9: Emagic *Logic Pro* v6.4.2
PC: Emagic *Logic Audio Platinum* v5.5.1

Setting Up A Mapped Instrument

To build the drum editor you first need to create and set up a Mapped Instrument in the Environment. I always like to create a new Layer when creating something like a drum editor, so first open the Environment and click and hold over the Layer name area. You can then double-click on the default name and call it something useful, such as Drum Inst. (*Logic* limits the number of characters in these text boxes.)

On the new Layer, create a Mapped Instrument from the New menu. The Instrument will open, but close it for now. Make sure the box next to the Icon field in the Parameters is ticked, otherwise the object won't appear in the Arrange page. You can rename the object to something useful by either using the Text tool or by double-clicking in the name field in the Parameters box. In this case I've called it Ultrabeat Mapped Drums, because I'm going to use *Logic*'s *Ultrabeat* drum plug-in to actually produce the sounds in this example — to do this I've cabled the Mapped Instrument object to an Instrument Audio object containing *Ultrabeat*. To create the cable between the Environment Layers, you click and hold on the little arrow at the top right of the Mapped Instrument while ▶

To cable between Environment Layers, hold down the Alt key while clicking on an object's cable output. This brings up a hierarchical menu containing all available objects to which it can be cabled.


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and larger
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Available in 2, 4 & 5 octaves
Unsurpassed MIDI controlling technology from the true pioneers, Novation, is now available in its most condensed format to date. With its funky Xpression pad and stick, the lightweight ReMOTE LE is the only choice for a musician wanting to trigger samples, lay down basslines or control a sequencer with true style.
Two Groups of 18 assignable knobs and buttons, (alongside the dedicated Transport controls) are factory preset in 16 onboard Templates, enabling the ReMOTE LE to work with most popular software applications straight out the box. Add to that many more Templates available for free online, along with overlays and a groovy Template Editor for building your own, and you have an unbeatable package!
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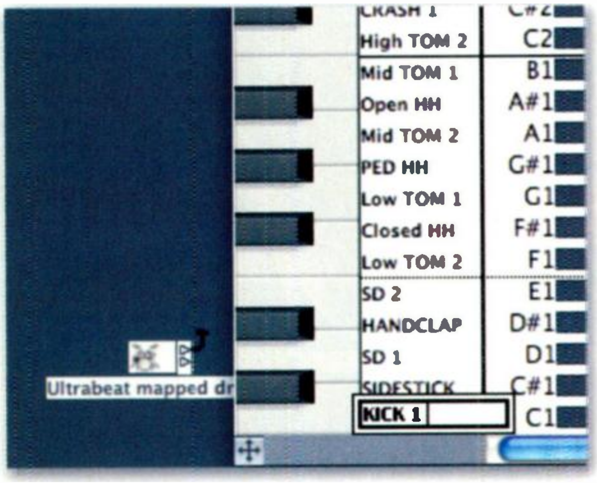


It's worthwhile labelling the keys in the Mapped Instrument's setup window to reflect which drum sound is triggered from each. The advantage of doing this is that then the Hyper Edit window can be configured automatically to match.

holding down the Alt key to access a hierarchical menu where you can select the requisite Instrument Audio object — *Logic* will ask you if you want to remove the channel's port setting, so click on Remove. You'll now notice that the Mapped Instrument has a stubby cable coming from the arrow. This indicates that it's been cabled 'through' Environment Layers to the required Instrument.

If you double-click on the Mapped Instrument, a window will open where you can actually play the loaded *Ultrabeat* sounds by clicking on the keyboard on the left. You can then label each key with the actual drum sounds that are loaded into *Ultrabeat* by double-clicking on the Input name column. The Initialise menu allows you to set the drum names to various defaults that may speed up this process in some cases. This stage is an important one in the setting up of the drum editor, so it's worthwhile talking the time to enter the names correctly. You can create many Mapped Instruments for the different drum sets which you use in your music and cable them to whichever Environment objects you want to provide the actual sounds.

Building The Drum Editor

If you select the Ultrabeat Mapped Drums object from the Arrange page, you'll notice that your previously created Environment Layer is now a pull-down menu on the Arrange window's Track column. Put *Logic* into Record Pause mode by clicking on the Transport bar Pause button followed by Record — just don't press Play! From your controller keyboard play in each note to which you have assigned drum sounds. In our example, we'd play in all the notes from C1 to B2, which play the sounds Kick 1 to Ride 2. Now click on the Stop button to bring *Logic* out of Record Pause mode. Notes will have been created at the Song Position Line for all the drum sounds.

Make sure the sequence is selected and open the Hyper Edit window from the main Windows menu. In the Hyper Edit window select Create Hyper Set, and then choose Clear Hyper Set from the Hyper menu. You'll be left with a single Volume definition in the right-hand side of the Hyper Edit window, which you should select and then delete using the Delete Event Definition menu item.

Once you have a clear Hyper Edit window, select the Multi Create Event Definition menu item from the Hyper menu. Click on All when *Logic* asks you to create the event definitions. A Hyper Set will now be created comprising event definitions bearing the names and note numbers of the drum sounds derived from the Mapped Instrument. Double-click the MIDI controls label and enter a more useful name, such as UBeat Drum Ed in this case. You can now delete the small MIDI sequence on the Arrange page which you used to create the Hyper Set.

Have Your Say!

If you want to suggest changes or improvements to *Logic*, then here's your chance! The Apple development team are inviting *SOS* readers to send in their suggestions of what they'd most like added or changed in *Logic*. Email your top five suggestions (in order of preference) to logicnotes@soundonsound.com, and we'll forward your lists on to the *Logic* team. We'll be asking them for feedback on which changes users deem most important and how these might be addressed.

Logic v7.2 News

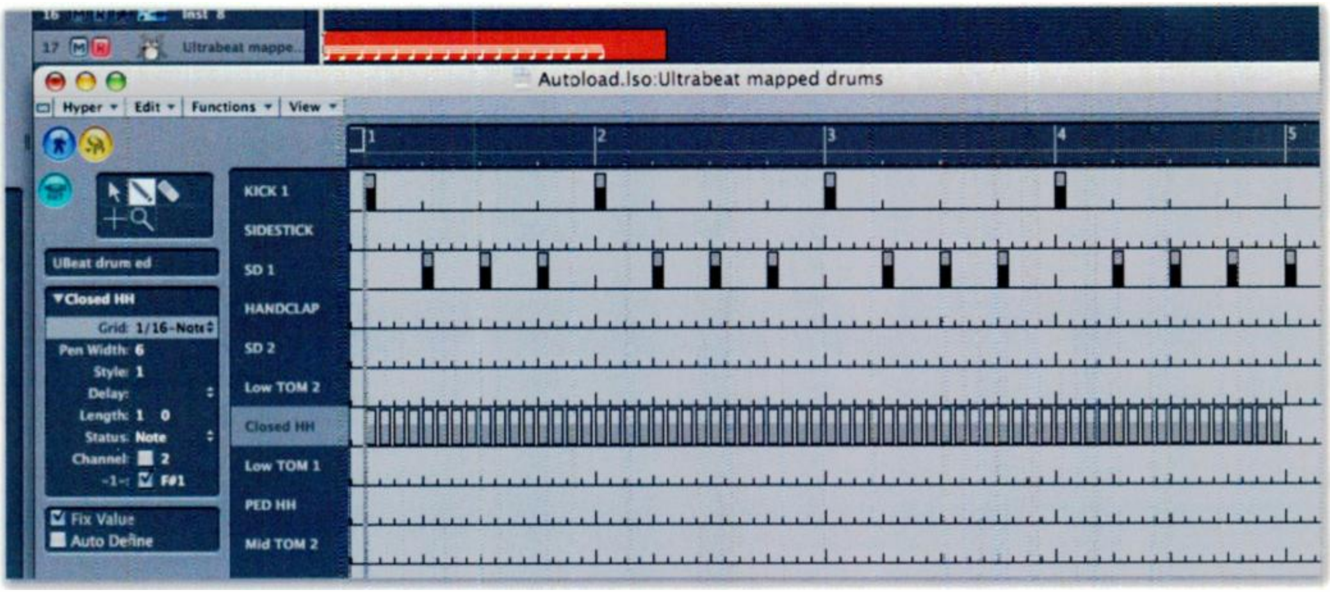
The latest *Logic*-related news from the NAMM show in California is that Apple were showing an Universal Binary form of *Logic Pro*, which runs on the latest Intel-based Macs as well as on PowerPC-based machines. This new software version is due to become available for a small fee in March as v7.2, and will also include a number of other refinements including improved support for hardware control surfaces, multi-channel software instruments, and Propellerhead Software's Rewire protocol.

Using Your New Editor

The Hyper Set becomes the 'matrix' upon which you can now draw your drum parts. Say you wanted to create a four-bar loop using the drum editor. First select the Ultrabeat Mapped Drums object from the Arrange page and use the Pencil tool to create an empty MIDI sequence, dragging it so it's four bars long. Select the sequencer and open the Hyper Edit window, and then choose the Ubeat Drum Ed Hyper Set from the pull-down menu under the Toolbox. The Pencil tool can now draw in notes for each drum sound using the different event definition lanes, and the Eraser tool can delete individual notes.

Each event definition can have it's own quantise grid, selected from the Grid pull-down menu in the Parameters box — this menu contains all of *Logic*'s usual quantise values, as well as any you've created yourself. You could, for example, set the Kick drum to eighth note or the hi-hat to ▶

It makes sense to set up your Multi Mapped Instruments on their own separate Layer, because this makes it much quicker to select as a Track Instrument in the Arrange window.



Creating patterns in the completed drum editor Hyper Set.

1/16th note, and then drag the pencil horizontally across the channel to quickly create notes. If you want to create a series of fixed-velocity notes, first create a single note at the velocity you want, then tick the Fix Value box underneath the Parameters and draw in new notes — they will all be at the same velocity value as your 'seed' note.

One special feature of the Hyper Edit window is particularly useful for hi-hat parts. A real drummer will only play one type of hi-hat hit at a time, as even the best players cannot play open, closed, and pedal hi-hats simultaneously! If you click to the left of the drum instrument name, a small 'hi-hat mode' icon is created, and this only allows one note at a time to be output from that particular event definition lane.

There are other Hyper Edit window features which come in useful when it's being used as a drum editor. Let's say you want to create a snare roll rising in volume, first click and drag using the Pencil tool to create a series of snare-drum notes at the desired Grid resolution (or play in a roll roughly from the keyboard and quantise it as necessary). Untick the Fix Value box and select the Crosshairs tool. Click at the bottom of the first note and drag the mouse to the top of the last note. You'll see a line attached to the mouse pointer. Clicking at the top of the last note will draw a line over the notes and create an upward velocity ramp. You can edit individual note velocities using the Arrow or Pencil tools if you select only one note — selecting a whole range of notes will cause the range of velocities to be moved up and down with the mouse. As you can imagine, drawing in drum parts in the editor makes it easy to build up complex drum patterns.

Of course, you can enter data into the drum editor in all the usual ways. If you are looping around a four-bar part in the Arrange page to create a sequence, you'll want to activate the Merge New Recording With Selected Region item in the Recording submenu of the Song Settings menu. This way the notes from each loop-through are placed into the same MIDI sequence in the Arrange window. You can use the Functions menu to quantise any selected note data or use the Transform menu to put back a bit of random variation into strictly drawn drum parts — the Humanise preset is particularly useful here, though you may want to set the Position Condition to Thru and only randomise velocities.

So that's all you need to know to get started with matrix-style drum editing in *Logic*. If you like this way of working, then you'll be pleased to know that the drum editor works in conjunction with all the other MIDI editors in *Logic*, and it can be used to record and edit any data, not just drum and percussive parts. SOS

Logic Tips

Mac OS X Tiger's Aggregate Device was one of the most eagerly anticipated features of the upgrade from Panther, as it allowed several hardware audio interfaces to appear as one combined interface within *Logic* and any other program. I've been using it with MOTU 896 Firewire and Aleisis X25 USB interfaces in order to access the latter's extra I/O pair for a send-return loop to a reverb — any latency is not really a problem, as it just adds a little pre-delay. I usually work with the Software Monitoring Audio Preference switched on so I can use monitor-only plug-in effects when recording vocals, the low-latency settings being perfectly usable on a Powermac G5. I've just bought a Line 6 Toneport guitar effects processor, which also doubles as a USB audio interface, so I created an Aggregate Device with the Toneport and the MOTU 896. My plan was to record guitar by selecting the Toneport inputs for recording while monitoring everything through the MOTU 896's outputs, as these are the ones connected to my monitoring system. This worked, except that the latency produced in this combination was quite unacceptable. As I don't need any of *Logic*'s effects while recording guitar, I switched off Software Monitoring, only to find that the audio from the Toneport could then only be routed through it's own outputs! It seems that *Logic* requires Software Monitoring to be On for the Aggregate Device to work. I can't find any mention of this anywhere in Apple's literature on the subject, so it's worth bearing in mind if you wish to use this feature.

Logic Express is a powerful and cost/CPU-efficient version of *Logic*, and many people have found that the feature list it provides is quite enough for their requirements. However, as you might expect given the price difference between it and *Logic Pro*, it has some limitations. Fortunately, some of these can be overcome with a bit of lateral thinking. An example of this is the *Logic Express* restriction on the number of physical audio inputs you can use to twelve, irrespective of how many you actually have on your audio interface. However, Apple's *Garageband* will allow you to use as many inputs as your audio interface has. So if you create a Song in *Garageband* with as many inputs as you wish to use, and then import it into *Logic Express*, the extra inputs become available alongside the 'allowed' 12.

Speaking of *Garageband*, given its higher CPU load you sometimes get Songs which won't run because the program complains that it's running out of CPU power. However, minimising the program into the Dock can sometimes get the Song to run nonetheless — it seems *Garageband*'s main resource hog is its nice graphics, so let's hope *Logic* doesn't inherit that particular feature!

Working With Video In *Pro Tools*

Once you've got your video and sync audio files into a *Pro Tools* Session, what do you do with them? Our short series on sound for picture concludes by explaining the ins and outs of dubbing, sound effects and Foley.

Mike Thornton

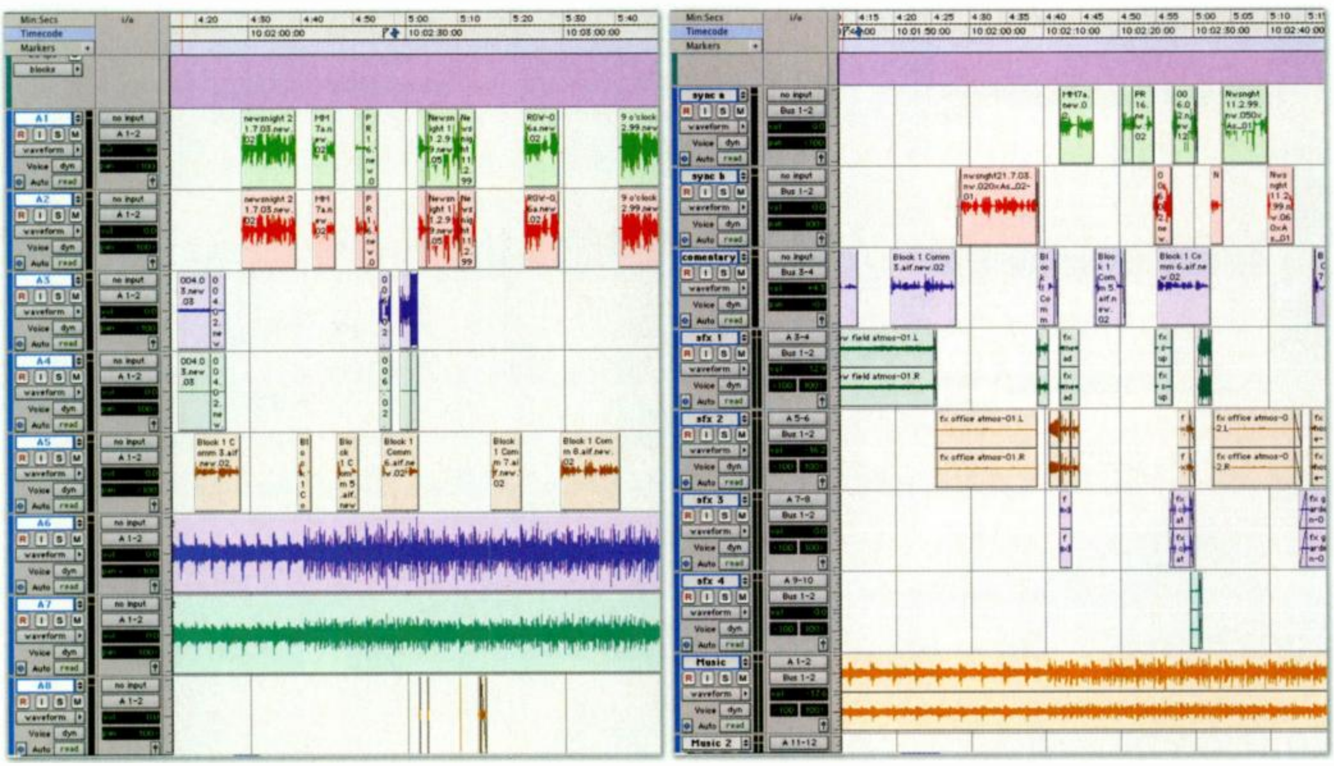
Over the last couple of months, I've explained how your Pro Tools system needs to be set up to work to picture, and how to get audio and video into a Session using an OMF or other source file. In this month's workshop we are going to look at what happens in the audio post-production workflow once we have imported the OMF into our Session, and the typical steps you might go through to get from this OMF to finished, mixed programme audio. We left off last month at the point where we had created a video track and imported the OMF file into our working Session. So now what?

Getting Organised

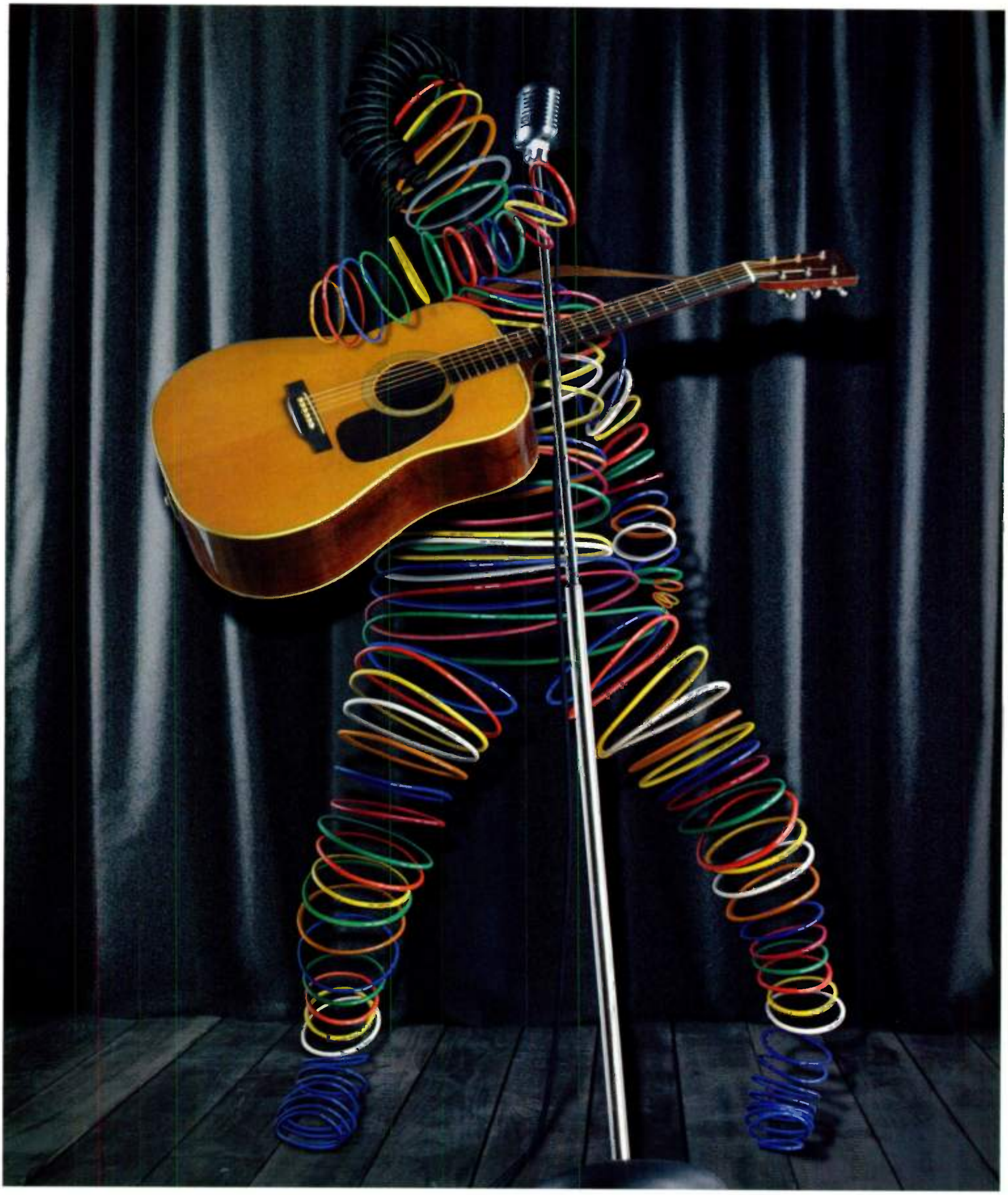
Take a look at the two screens below. Both are screenshots from the Edit window of a TV audio post-production Session. The first is how the Session looked immediately after importing the OMF file into it and the second is from the completed programme. Notice that the former consists of just mono tracks, like a *Pro Tools* Session from the days before we had stereo tracks. The track names are generic (A1, A2, A3 and so on) and those tracks that appear stereo don't always contain stereo material, as with A3 and A4 in the middle of the window: there is audio on A3 that isn't on A4.

To get everything in order, I look at how the tracks have been laid up by the video editor and start to rearrange them to suit my way of working. I create a batch of stereo tracks and name all the tracks to help navigation. I knew from the video editor that in this case tracks A1 to A4 were original sync sound. That is sound that was acquired 'in sync' with the original pictures. Note this may, or may not, be any use. The sync sound may have the director talking all the way through the shot, it may have been badly recorded, if there was no specialist sound recordist on the shoot, or it may simply be unusable because of unseen extraneous noise like a lorry passing just out of shot. In TV the director will rarely allow time to wait for a 'quiet window'. In music work it's 'we will fix it in the mix'; in audio post it is 'we will fix it in the dub'! In this example most of the sync sound was 'dual mono' where the same signal was recorded on both tracks. However some scenes were 'twin mono' where a different mono signal was recorded onto each track — in this case, the camera mic on one track and a personal radio mic on the other.

Having rearranged my tracks, I start working through the Session looking at what ▶



Before and after: the sync audio as imported from the OMF file (left), and after it has been reorganised (right).



has been laid up by the video editor. Some choices are easy. For example, I move all the music onto stereo tracks and label them Music, Music 2 and so on. I do the same with any Commentary track too. I also create some stereo sound effects tracks ready for the extra sound effects I will add later, either to replace poor sync sound or to cover non-existent sync sound.

I check each region in the Session to find out what it is and how useful it is. If the sync sound is 'dual mono', I get rid of one half to save cluttering up the session. If it is 'twin mono' then I listen to each track separately and decide which one to use; alternatively, if the shots are cut so there are both wide shots and close-ups, I may choose to keep both tracks and mix between them to suit. Even if the sync sound is unusable I will listen to it to help me select a suitable sound effect to replace it with.

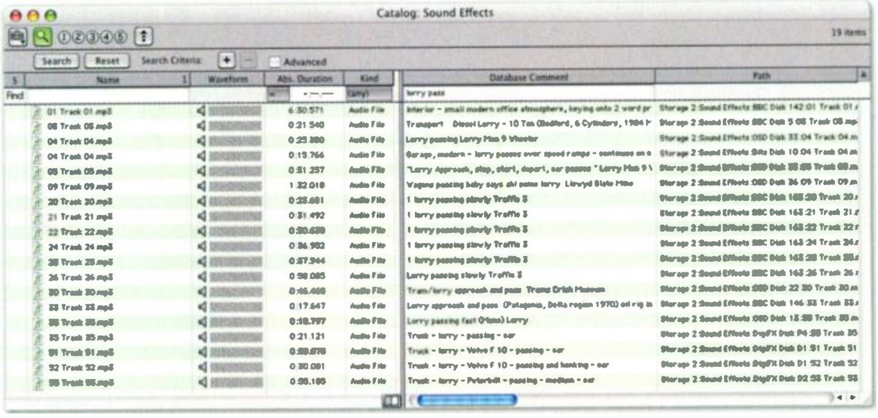
The opening shot of this example had a lorry coming into shot from the background and passing from right to left, followed by a shot of a combine harvester. The sync sound was in mono, probably from the camera mic, and I chose to replace most of it with stereo effects from my library. The only sync sound I kept in this section was a close-up of the combine harvester coming into shot at the end. I needed a 'new' lorry and a combine-harvester effect and so went to the library to find suitable candidates.

Post-production work is thus a matter of systematically working through the Session adding and replacing sound effects, massaging the transitions with cross fades and so on, and then mixing the effects in with the sync sound, commentary and music to produce the finished programme.

Sound Effects

Post-production facilities have libraries of sound effects. These will be a combination of commercially available libraries from companies like the BBC, DigiEffects, Sound Ideas and the like, and their own specially recorded sound effects. A growing third source of sound effects is on-line downloadable libraries like Sonomic and Sound-Effects-Library.com (there are some 200-plus of my own sound effects available on the latter!). These sites allow you to search for a particular sound effect, audition it, buy it and download it to use it in your project.

In the early days sound effects were stored on vinyl and quarter-inch tape and you searched through books for a particular sound effect, then went and got the tape or disc off the shelf and played the effect from that. Then CDs came along, first with paper directories and then searchable databases, but now the most common storage technique is to keep them on hard drives, either as MP3s or WAV

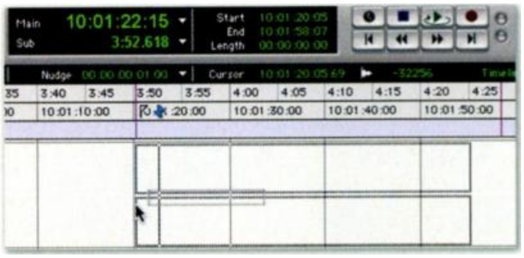


In my Digibase database of sound effects, I've used the 'Database Comment' field to enter information on each file that can be searched. Here are the 19 entries that are found by searching under 'lorry pass'.

files, and use dedicated software such as *Soundminer* or *Mtools* from Gallery Software to search, audition and import them directly into *Pro Tools*.

Alternatively you can do as I do and use Digibase Pro from within Pro Tools TDM. This enables you to search and audition your sound effects library without ever leaving *Pro Tools*, but isn't available on LE systems. I have all my sound effects (over 20,000 and growing!) on one drive and I have used Digibase Pro to catalogue them all, so from the Catalog section of the Workspace window I can select my Sound Effects catalogue and open it in another window.

From this window I can search for a suitable sound effect, for example for the opening shot in the Session described here. Click on the magnifying glass button on the Catalog window and 'Find' row will appear towards the top of it. You can do a search in any of the fields. On my system, the sound effect details are in the 'Database Comment' field, so I enter a suitable set of keywords in this field (in this case I chose 'lorry pass'), hit Enter and *Pro Tools* searches my catalogue of 20,000-plus sound effects and comes up with 19 items. Now I can audition any of these possible candidates in two ways: clicking and holding on the 'speaker' icon for that file plays that item from the start, whereas clicking and holding anywhere in the Waveform section plays the file from that point. Once you have identified a suitable file, you can drag it from the Catalog window across into a suitable track on the Edit window. Note that as you drag it around the Session, the video will scroll backwards and forwards, so helping you place or 'spot' the sound effect more accurately even though the region is still an 'outline'. Once you let go, *Pro Tools* will then automatically import and convert the file in the background. Whilst it is doing this it shows the region in light blue. Once the conversion process is finished, this will change to a normal region with name and waveform. At this point, I rename the region with an appropriate name for the Session.



Files can be dragged straight from Digibase into the Edit window. The larger grey outline shows where the region will be created when I drop the file.

Thanks to Digibase Pro I have been able to search for, audition and import a sound effect into my *Pro Tools* Session all without leaving *Pro Tools* or my seat! With sample CDs, I would have to had to switch out of *Pro Tools* into a separate sound-effects database, in my case *Filemaker Pro*, search for an effect, get the list of 19 effects, then get up and pull the appropriate CDs from the shelves, listen to

Useful Links

Sound-Effects-Library.com
W www.sfx-gallery.co.uk/

Sonomic
W www.sonomic.com/

Soundminer
W www.soundminer.com/

Mtools
W www.mtools.info/

***Doctor Who* story**
W www.digidesign.com/news/details.cfm?story_id=2949

'You Are Surrounded' series in *SOS*
W www.soundonsound.com/sos/aug01/articles/surroundsound1.asp

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each track to decide which one was right, and finally load that into *Pro Tools*, and it would have been so much slower.

Digibase And *LE*

You can use Digibase with *Pro Tools LE* but you won't be able to make or use searchable catalogues. You can, however, search filenames from the Workspace window, so providing your sound-effect filenames have suitable key words in them, you can search for files that way. Failing that, use a separate database like *Filemaker Pro* to handle the search and then navigate your way to the correct location from within *Pro Tools* and drag the chosen files from the Workspace window into the Edit window. Apple's iTunes makes an excellent free ripping software package that puts each CD into a nice convenient folder for you.

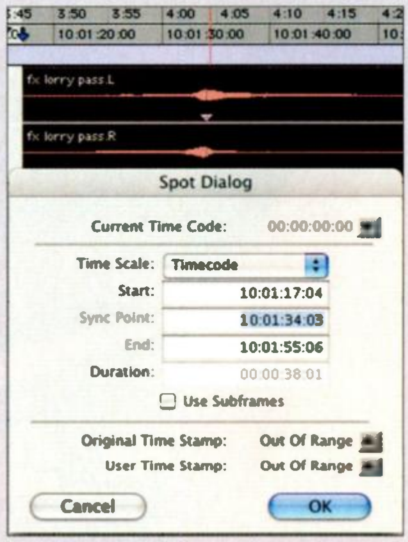
M&E Tracks

In audio post-production it is usually necessary to provide two mixes: a main mix, which is a mix of the all the elements for the programme, and an 'M&E' mix. M&E stands for Music and Effects and is similar to an instrumental mix for a vocal album project. The M&E track is created to make foreign-language repurposing as easy as possible and so is a partial mix, which contains all the music and effects elements but not the dialogue or any commentary. When it comes to producing a foreign-language version, you have a mix of all the universal elements and all that is required is to add the dialogue and commentary elements in the desired language. Moving material onto specific effects and music tracks makes it much easier to create these M&E mixes.

Spotting

An alternative way of placing sound effects within a *Pro Tools* Session is to establish the point at which you want the sound effect to play and use the Spot feature in *Pro Tools* to locate the effect in the correct place. Using the example of the truck passing, as covered in the main text, you would first put a 'Sync point' on the region at the point the lorry passes using Command+',' (Mac) or Ctrl+',' (Windows) — in the screenshot, the sync point is the triangle on the bottom edge of the region. Remember you can only have one sync point per region. Now place your cursor at the point on the timeline you want to line up this sync point with (in this case it's a lorry passing in the video). Choose Spot editing mode from the toolbar, click on the time code in the Counter to highlight it and copy it into the clipboard using Command+C (Mac) or Ctrl+C (Windows). Trying to drag the region will now open up the Spot window. Paste the timecode value from the clipboard into the Sync Point field and Click OK. The sync point of the region will jump (spot) to the timecode point you identified with the cursor, and the sound effect of the lorry should work 'in sync' with the video.

Above: I've added a 'sync point' to the effect region at the significant point (in this case, where the lorry passes the listener). This can then be used to 'spot' the region to a timecode value.



With Spot editing mode active, moving a region opens the Spot dialogue.

Going Deeper

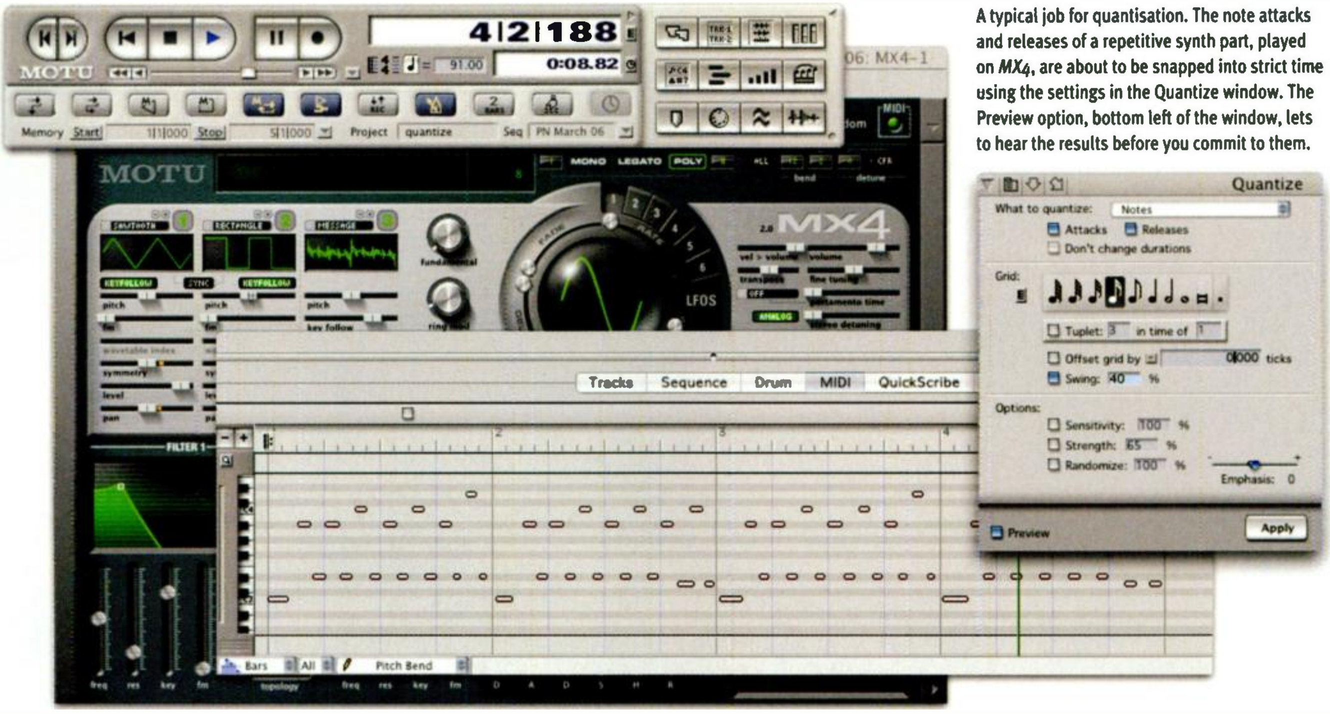
In this article, I have outlined some of the basic procedures that go into the audio post-production of nearly all the TV programmes, adverts, and films you will watch. If you want to go into greater detail, I recommend again the book *Pro Tools For Video, Film & Multimedia* by Ashley Shepherd (Muska Lipman: ISBN 1-59200-069-X). There are also regular articles on the Digidesign web site, including an interesting one about the post-production of *Doctor Who* in the Users section of Digidesign's UK site. Digidesign's Digizine also has articles and a regular feature called 'Post Scripts'.

Next month I am going to concentrate on recording and mixing in surround for both music and broadcast post-production using *Pro Tools*. If you feel like doing some homework, I recommend the excellent series of articles called 'You Are Surrounded' published in *Sound On Sound* from August 2001 to April 2002 (see 'Useful Links' box). SOS

Foley

If you watch the credits from a film you may have wondered what Foley is all about. Foley is the name given for recreating sounds needed to match the action in shot and is done by a person 'acting' rather than assembled from pre-recorded sounds. It is named after Jack Foley, who invented the process when movies were changing from silent to 'talkies'. Foley would be called in to create all the action sounds for adding to a silent movie by watching the movie and re-enacting the actors' movements.

Although I tend to use pre-recorded sounds, there was a sequence in a project recently where I needed to add the sound of people eating crackers. I couldn't find a suitable effect in my library, so I miked myself up and played the video from within *Pro Tools* whilst recording myself first breaking and then eating the crackers, trying to keep in sync with the action on the video track in *Pro Tools*. I then went back and tidied up the crumbs and the sync errors and all was well! It reminds me of the time when I was working for a commercial radio station and they wanted to make pancakes on air for Shrove Tuesday one year. The station no longer had a kitchen, which was how it had been done in the past, so my solution was to go home the night before and make pancakes in my own kitchen. I must have looked very strange beating the batter and then tossing the pancakes whilst recording it all, wearing headphones of course. The following morning I played the sound effects in as the presenter 'acted out' the process and no-one was any the wiser!



A typical job for quantisation. The note attacks and releases of a repetitive synth part, played on *MX4*, are about to be snapped into strict time using the settings in the Quantize window. The Preview option, bottom left of the window, lets to hear the results before you commit to them.

Feel By Numbers

Using Quantising In *Digital Performer*

Quantising is all about turning your sloppy playing into rigid, computerised perfection, right? Well, no — at least, it doesn't have to be. This month we check out the finer points of quantising in *DP* and look at what it can do for your music.

Robin Bigwood

Quantising — a really distinctive and powerful technique with a nerdy name — has been around since the dawn of MIDI sequencing. It's a process that takes MIDI notes or other events and 'snaps' them into line with a user-configured rhythmic grid, often resulting in inhumanly rhythmic playback that can be exhilarating or mind-numbingly lifeless, depending on the musical context and the skill with which it's applied. But quantising doesn't have to make everything you do sound like Kraftwerk, especially when you have *DP* as your sequencing platform. Used wisely, it can allow you to actually enhance the 'feel' aspect of your sequences, while easily taking care of more obvious errors.

As you might expect, *DP*'s quantisation features are extensive, sophisticated, and crop up in all sorts of places — they're by no means a 'one size fits all' solution. In fact, in true *DP* fashion, there's a staggering degree of flexibility. So rather than exhaustively going through every single feature one by one, I thought we'd look at how you might best use quantise in a range of typical contexts, to achieve a variety of effects.

We Are The Robots

It might not be big and clever, but sometimes really strict, mathematical quantising is just what's required, especially in more 'electronic' musical styles. As you'd expect, it's simple to achieve in *Digital Performer*.

Probably the best way to apply this sort of quantisation is as a 'destructive edit' that actually changes the MIDI data you've recorded into a track. In the case of MIDI notes, they'll be moved to coincide with strict rhythmic sub-divisions of the bar (or 'measure', in MOTU-speak). Here's a typical course of action for this type of quantising, which assumes you already have a MIDI track that was recorded with reference to *DP*'s metronome:

1. First, select where you want to quantise. You could do this using any normal selection technique in *DP* — selecting phrases in the Tracks window or dragging over notes in the Sequence Editor or Graphic Editor, for example, or even by hitting Apple-A (Select All).
2. Then hit Apple-0 (zero) or choose Quantize from the Region menu, to bring up the Quantize window (see screen above) and decide what you want to quantise. We'll assume, for now, that it's just MIDI notes, so choose Notes in the 'What to quantize' pop-up.
3. The rest of the window is dedicated to how the notes will be quantised. The first thing to decide is the basic grid value, and for that you need to consider what the shortest note value you played was. Figure out your smallest sub-division of the beat

and choose a rhythmic value to match.

4. In order for your notes to be 'snapped' into time, you'll need Attacks checked in the top part of the window, but it's up to you whether you also have the ends of notes quantised. For the most natural effect, you should probably choose just Attacks, with or without the 'Don't change durations' option, which causes the note releases to move relative to the attacks. Choosing to have Releases quantised too, however, can create a very mathematical, almost step-time effect, so experiment.
5. Make sure that nothing else in the window is checked and click Apply. Your MIDI track is quantised according to the settings you made.

If you find it difficult to predict how the settings in step three and four will affect the musical outcome, use the Preview option. To see what it does, run through the same steps as I've just described, but first set up *DP*'s Memory Cycle loop markers around the section you're quantising. When you reach step three, play your sequence and let Memory Cycle loop round and round. In the Quantize window, check the Preview box and you should instantly hear the effect of the grid value you've chosen, and of the Attacks, Releases and 'Don't change durations' options.

I'm Going In

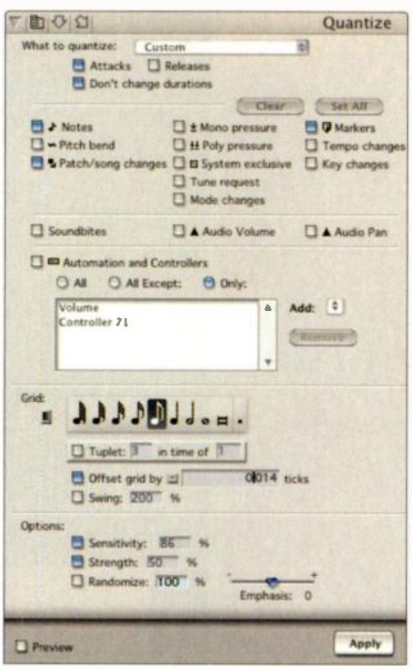
Preview can help you zone in on useful settings, but when you're already certain what kind of musical effect you want, there's a way to cut out the entire note-selection and quantise-application rigmarole and just quantise as you record. This is called Input Quantize, and because it relates to the actual recording process, rather than a pre-recorded region, it's found in the Studio menu, or can be called up with the Ctrl-Shift-I shortcut.

All the same quantisation options are offered in the Input Quantize window, but there's an additional 'Enable...' check-box, which allows you to keep the window open during extended recording sessions and simply enable the function as and when you need it.

I find that Input Quantize comes into its own when I'm building up drum parts using the Memory Cycle function (allied with Overdub record mode). It means that every new 'pass' goes in perfectly in time. The alternative — having to select every new drum you record and apply the Region menu quantise — is particularly tedious and best avoided! For more about this approach to recording, see the Memory Cycle section in July 2005's *Performer* Notes.

Quantisation Tips

- **Tidy Up Markers: Quantising isn't just for MIDI notes — it's possible to quantise almost any events in *DP*, including Markers. I find this exceptionally useful if, for example, I'm working on a piece and have just written some markers 'on the fly', by hitting Control-M during playback. Markers placed like this don't automatically align with beats or bars, but you can quickly snap them into place by hitting Apple-A to select all of your sequence, then Apple-0 (zero) to open the Quantize window. Now choose Custom from the 'What to quantize' sub-menu, then deselect everything except Markers. Choose a suitably coarse grid (maybe half notes, or minims), hit Apply, and you're done!**
- **Control Controllers: If you can quantise Markers, why not do the same with controller data? This opens up the very real possibility of quantising controllers relating to filter cutoff and resonance in a synth MIDI track, say, and producing rhythmic, almost sample-and-hold effects. The 'What to quantize' pop-up in the Quantize window can be set to Controllers, which quantises all controllers, or Custom, which allows you to choose exactly which ones to quantise.**
- **Customise The Grid: The quantise Grid doesn't have to be tied to musical rhythmic values. By clicking the little Time Format button near the word 'Grid', you can quantise to absolute time values, SMPTE time intervals, or even audio samples. This is potentially very useful when matching music to rhythmic cuts in film.**
- **Plug-in To Quantising: The Quantize window's Preview function is a tremendous help in refining settings, but if you're the sort of person who can never make their mind up and you want to be able to quantise and then change your mind at some point down the line, *Realtime Quantize* is for you. This is a MIDI plug-in that you select in a MIDI track's plug-in slot in the Mixing Board. It offers exactly the same features as the Region menu 'offline' Quantize, but it quantises in real time, only during sequence playback, leaving you free to change its settings at any time. If you ever decide you need a destructive quantise after all, just select the data in the track the plug-in is applied to and choose 'Capture Realtime MIDI Effect' from the Region menu.**
- **It's Not Just For MIDI: As of *DP* 4.5, quantising has not been just for MIDI events. You can quantise beat-detected audio too. *DP* gives you three options: to quantise beats within soundbites, soundbites themselves, or beats and soundbites. It's also not particularly worried about whether you recorded your audio to a click or not. Clever stuff, which you can read about at length in *Performer* Notes from *SOS* February, March and April 2005.**



Who said quantise was just for MIDI notes? *DP* lets you quantise every kind of sequence event (except loops, actually) via the Quantize window's 'Custom' view.

Shuffle Rhythms

Many musical styles make use of compound rhythm — that is, a fundamental rhythmic 'feel' whereby each beat is split into threes or sixes rather than twos and fours, as in a typical rock shuffle. Some *DP* users might deal with this by entering a compound time signature (6/8, for example) in the Conductor Track, which would result in your basic beat being a dotted crotchet (quarter-note). That's a perfectly good approach, since Quantize's grid has a dotted note option, far right. But if you're happier just sticking to so-called 'simple' time signatures like 4/4 you can still easily work with — and quantise — compound beats.

In the Quantize (or Input Quantize) window, take a look at the Tuplet section below the grid section (see screen, left). You have to enable it by clicking its check-box, and then configure it by typing numbers into its two boxes (or by clicking and vertically dragging in the boxes). If you'd recorded a shuffle-rhythm track using a 4/4 time signature and crotchet (quarter-note) metronome beat, you could quantise the track by selecting a quaver (eighth-note) in the grid section and '3 in the time of 2' in the Tuplet section. You're just telling it that you need three notes in the space of two, and since two quavers add up to a crotchet (quarter-note), you end up with a quaver triplet grid. But the flexible nature of the Tuplet section means that you could just as easily quantize quintuplets or septuplets, or tuplets so complicated that no-one's thought up a name for them yet!

Swinging

Musically speaking, Swing describes a rhythm based on a division of the beat

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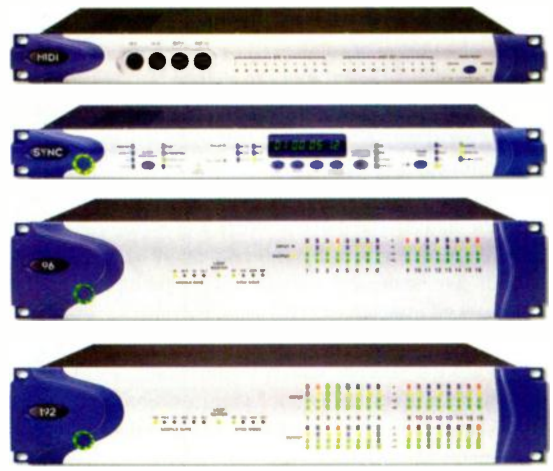

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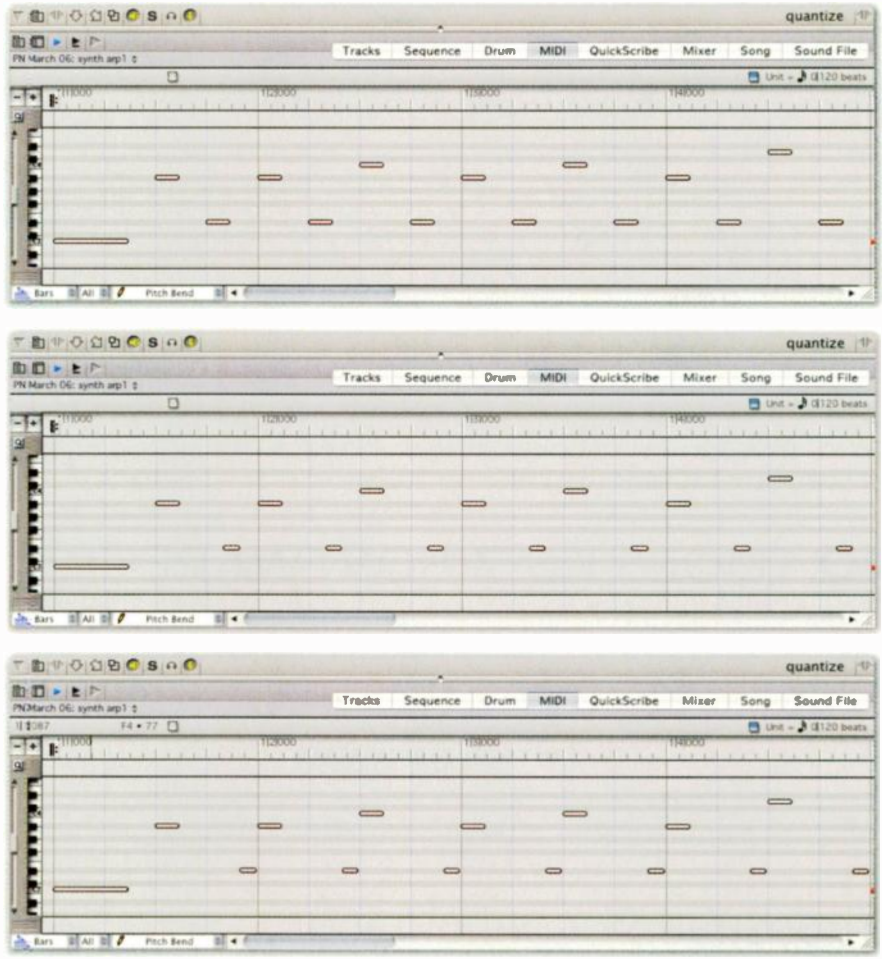
into long-short pairs — think of the most clichéd jazz ride-cymbal parts — and this, again, results in a compound beat.

However, any decent drummer or percussionist knows that swing is a very flexible thing. For example, the rhythmic feel of a quaver or semi-quaver (eighth- or sixteenth-note) hi-hat part can change dramatically with the introduction of subtle long-short variations in each pair of hits. And for some musical styles — such as garage and two-step — the precise degree of swing used can make or break a groove.

If you'd like to swing in the safety of your own home, and without any danger of long-term psychological damage, try this:

1. Record a 'straight' quaver (eighth-note) hi-hat pattern, with your favourite drum sound source, into a MIDI track, using the metronome and a 4/4 time signature.
2. Set up a Memory Cycle loop around this and play back your sequence at a reasonably fast tempo — say 130bpm.
3. Select the notes you just recorded and hit Apple-0 [zero] to open the Quantize window. Check the Preview box so that you can immediately hear the changes your settings will make.
4. Start off by choosing a quaver (eighth-note) in the Grid, making sure that nothing else in the Quantize window is enabled. You should hear your hi-hats snapped perfectly into time.
5. Click the Swing checkbox. If the percentage value is set to the default 100 percent you should hear your hi-hats snap into a swung compound-beat rhythm (see screens, right).
6. Now you can try different Swing amounts. The easiest way to dial these in is by clicking and dragging up or down in the percentage value box.

For many typical 'straight' musical styles, the introduction of, say, 20-50 percent of swing can really help to loosen up hi-hat and other continuous rhythmic parts, especially if you're able to complement it with a little dynamic variation, such as making the 'swung' hits or notes (ie. those 'between' beats) a little quieter. You could do this during the recording stage, or with subsequent editing of note velocity in the Sequence or Graphic editors.



By dialling in increasing amounts of Swing when you quantise you can turn a 'straight' rhythm into a compound-beat shuffle, or even more pronounced long-short pairs. These screengrabs show a straight rhythm (top) being treated with 100 percent (middle) and 200 percent (bottom) Swing.

Maintaining Feel

Very often, when using quantise, the problem facing you is tracks that are a bit too loose when unquantised, but which lose all their feel when quantised strictly. Something that can come to the rescue in this situation is the Strength option. This is found towards the bottom of the Quantize window, and by default is unchecked, so you need to click it on and dial in a value to start working with it.

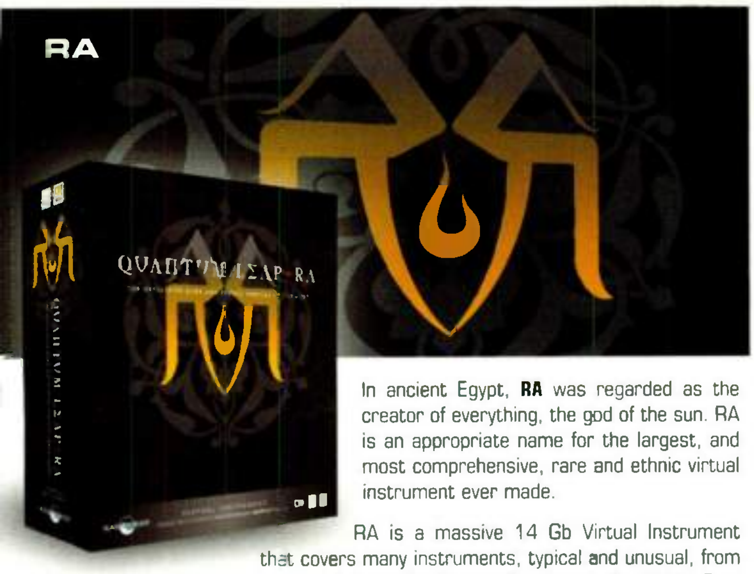
The basic idea of Strength is this: if the option is unchecked, or turned on with a value of 100 percent, any events (such as MIDI notes) moved when quantise is applied are yanked from their original positions and repositioned precisely, according to the Grid, Tuplet and Swing settings. But when Strength is set to a value of less than 100 percent, any events moved end up between their original position and their fully quantised position. So if Strength is set to 50 percent, for example, quantising moves out-of-time notes only halfway towards their precise, mathematically correct positions. This opens up the possibility of much more gentle and less destructive quantising, eminently suitable for more soulful tracks that nevertheless need some tidying up. You may need to experiment with Strength settings a little every time you use it, as the results can vary depending on the material you're quantising (see screens overleaf).

Another subtle quantise concept is Sensitivity. This is possibly an 'experts only' quantise option, but it can do some clever things. To understand Sensitivity, it helps to first think about how quantise actually

Get Smart With Quantising That Scores

The sharp-eyed menu-watchers amongst you may have spotted Smart Quantize in the Region menu. If you thought this might be some sort of 'clever' quantise option that could vary its settings dynamically, depending on the rhythmic content in your tracks, you'd be right, but it's not quite as useful as you might imagine. It actually performs a specific function — namely, preparing MIDI data for export to third-party scoring packages such as *Sibelius*. These packages sometimes transcribe what might seem like perfectly simple MIDI sequencer files in the most horrendously complicated (and unreadable) way, so Smart Quantize is there solely to make *DP* sequences more notation-friendly. It can be a time saver for general use too, but its lack of controls and configurability make its effects unpredictable.

RA



In ancient Egypt, **RA** was regarded as the creator of everything, the god of the sun. RA is an appropriate name for the largest, and most comprehensive, rare and ethnic virtual instrument ever made.

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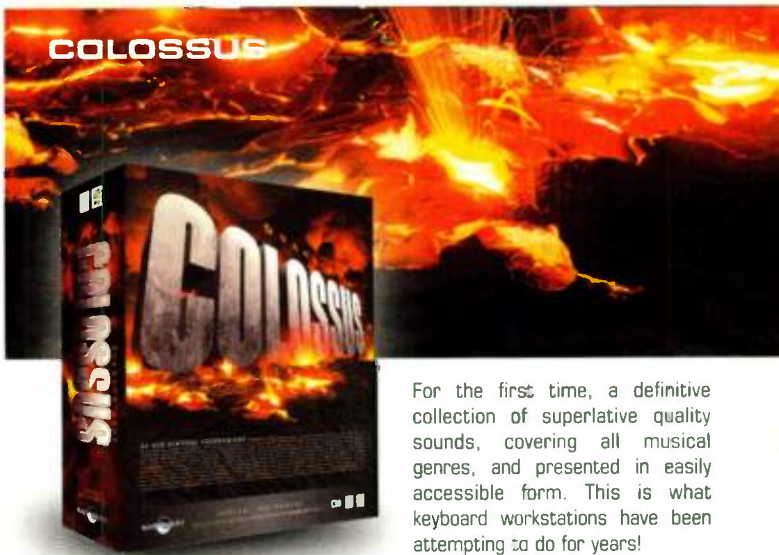
One thing is certain; there has never been a rare and ethnic collection produced that matches the quantity and quality contained in Quantum Leap RA. Includes special version of NATIVE INSTRUMENTS Kompakt sampler.

COLOSSUS



For the first time, a definitive collection of superlative quality sounds, covering all musical genres, and presented in easily accessible form. This is what keyboard workstations have been attempting to do for years!

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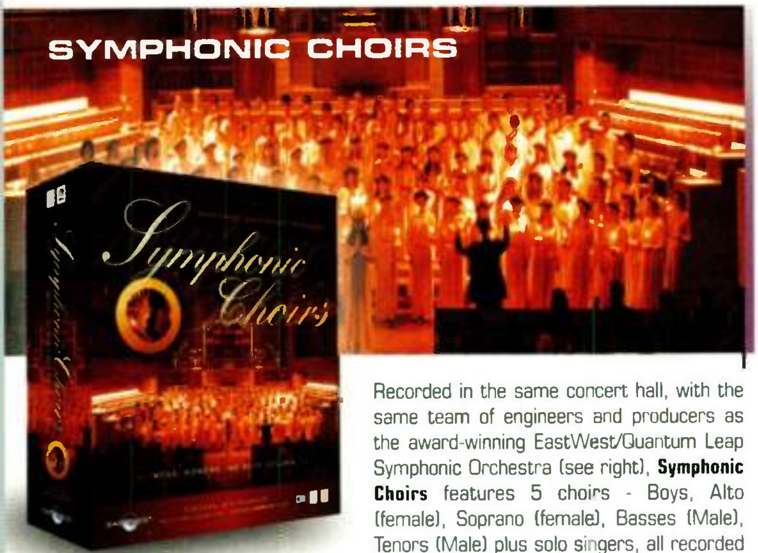
SYMPHONIC CHOIRS



Recorded in the same concert hall, with the same team of engineers and producers as the award-winning EastWest/Quantum Leap Symphonic Orchestra (see right), **Symphonic Choirs** features 5 choirs - Boys, Alto (female), Soprano (female), Basses (Male), Tenors (Male) plus solo singers, all recorded in position and chromatically sampled with multiple dynamics (non-vibrato, light-vibrato and heavy-vibrato).

3 simultaneous stereo mic set-ups were used (close, stage and hall) which can be mixed in any combination to control tone and ambience. All choirs were recorded in stunning 24bit by Grammy-winning classical recording engineer Prof. Keith O. Johnson.

The included word-building utility means you simply type the words you want the choir to sing into your Mac or PC!

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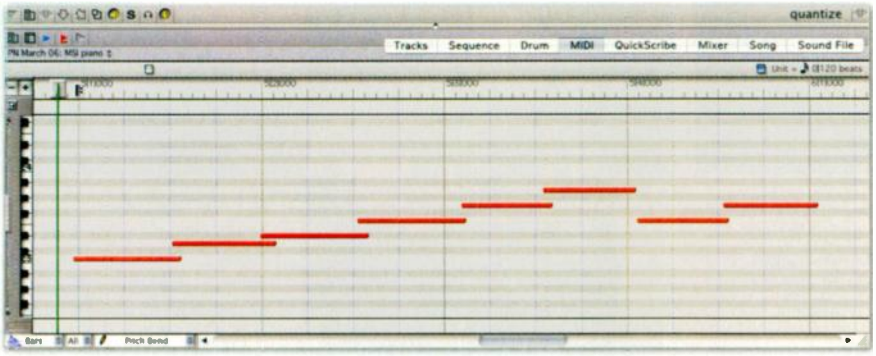
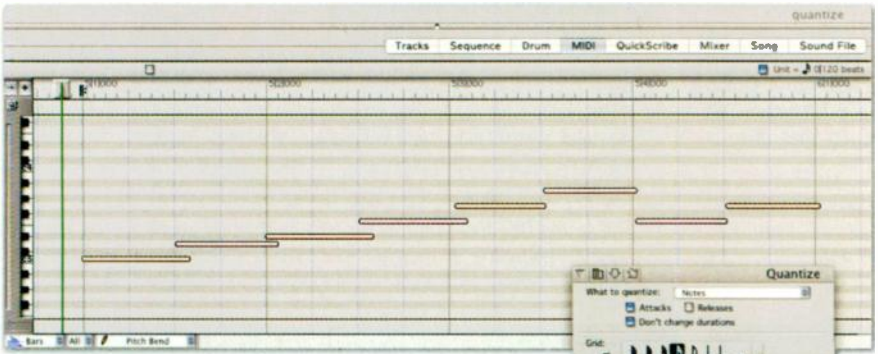
QUANTISING TIPS





The Strength option allows you to improve sloppy timing without having to align notes exactly with the quantise grid. It's a great way of tightening up your rhythm without destroying its 'feel'. In these grabs we see the result of quantising a sloppily played piano part (top) with 65 percent strength (bottom).

works. Let's say you choose a grid setting of a crotchet (quarter-note). Assuming that you're not using any other quantise features such as Swing or Strength, *DP* first finds any events which lie up to a quaver (eighth-note) before or after each beat, and then 'snaps' them directly on to the beat. To look at it another way, quantise casts a crotchet-sized 'net' around each beat, to catch those events that don't align exactly with it.

The Sensitivity option, then, allows you to shrink the size of this net, so that events that lie 'in the gaps' of the quantise grid are never included in the quantise action, while events that lie nearer to the rhythmic divisions of the grid are quantised as normal.

Why on earth would you need this? Well, for one thing, it allows you to have rhythmic accuracy where you need it, while allowing 'free' rhythm in other places. As an example, imagine you have a synth solo track. For 16 bars there's a strong downbeat at the beginning of each 4/4 bar (that needs to coincide with a precisely placed drum hit on another track) and a loose, free-flowing solo for the rest of the bar. You could do with a quick solution that quantised all your downbeats but left the rest of the bar untouched. As you might guess, this is possible using the Sensitivity option — and here's how you'd do it. First, select the section, then in the Quantize window choose a semibreve (whole-note) grid, turn on the Sensitivity option, and dial in a sensitivity of, say, 10 percent.

By doing this, you're narrowing the quantise 'net' to just 10 percent of the length of each semibreve — five percent before it and five percent after — which, of course, aligns perfectly with your 4/4 bars on their downbeats. Now only the notes very close to your downbeats are quantised, and *DP* ignores the rest of each bar.

It's also possible to set the Sensitivity value to a negative number, in which case the quantise 'net' is aligned not around the divisions of the grid, but around a point halfway between them. This is actually very useful, because you can use it to make quantise catch only notes that are really out of time and some way from the rhythmic grid divisions you've chosen, while ignoring ones that are already aligning with the grid or are close to it. You might have deliberately played a snare drum track, for example, with each hit fashionably behind the beat. By quantising this track with negative Sensitivity you'd leave most hits where they were, quantising only those that had strayed too far out.

Playing with Sensitivity might not be something you'll do every day, but getting under its skin and knowing how to use it to your advantage can be very handy every now and then.

Creating Feel!

As the possibilities afforded by the Sensitivity parameter show, quantising doesn't have to be about making music soulless. Two extra parameters ensure that it doesn't have to be 'square' at all.

The first is the 'Offset grid' option, which simply moves quantise's rhythmic grid ahead of or behind the rhythmic grid used by your sequence as a whole. Clicking the little button next to the value field selects positive or negative, and the amount of offset is entered in beats and ticks. Likely candidates for offset quantising include individual drums, to modify the 'feel' and groove of a pattern, or perhaps one or more of several arpeggiated synth tracks, to achieve precise delay-like effects.

Finally, there's Randomize. This is an impostor in the quantise camp, as it can only detract from the rhythmic precision promised by the other features. However, a few percent of Randomize can loosen up an otherwise rigid quantise, and by using some Emphasis you can decide whether you want your Randomized placement mostly ahead of the beat (negative emphasis) or behind it (positive emphasis). Higher values of Randomize, which cause notes to be placed all over the shop — even changing their order — are the *DP* equivalent of opening a picture of a friend in *Photoshop* and using Liquify, Pinch and Spherize on it. We've all done it, and it's funny at times, but not particularly useful all that often! SOS

Digital Performer News

Just when I thought *Performer* Notes was wrapped up for another month, MOTU announced the imminent availability of audio and MIDI interface drivers for the new Intel-based Macs revealed at Macworld San Francisco. As for *Digital Performer, Mach Five, MX4* and *Symphonic Instrument*, MOTU say they are "currently qualifying... software products for use with the new Intel iMac and Macbook Pro". Watch this space...

In the meantime, users of those quaint, old-fashioned IBM and Motorola Macs can take advantage of a MOTU software update that has already been released. *Symphonic Instrument* has gone to version 1.1, gaining new features that include stand-alone operation, disk streaming, 64 parts per instance, and individual part outputs. This is a major update, which owners of version 1.0 of the instrument can download for free from www.motu.com.


-30
-20
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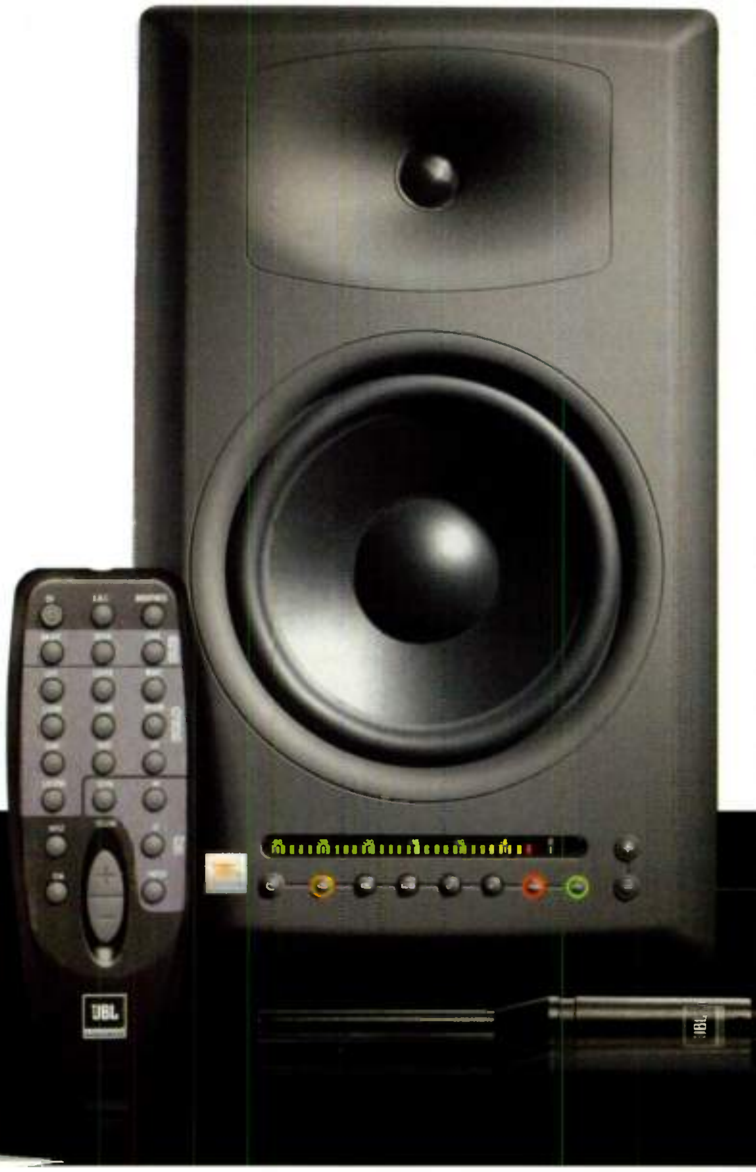
SOLO
RMC
EQ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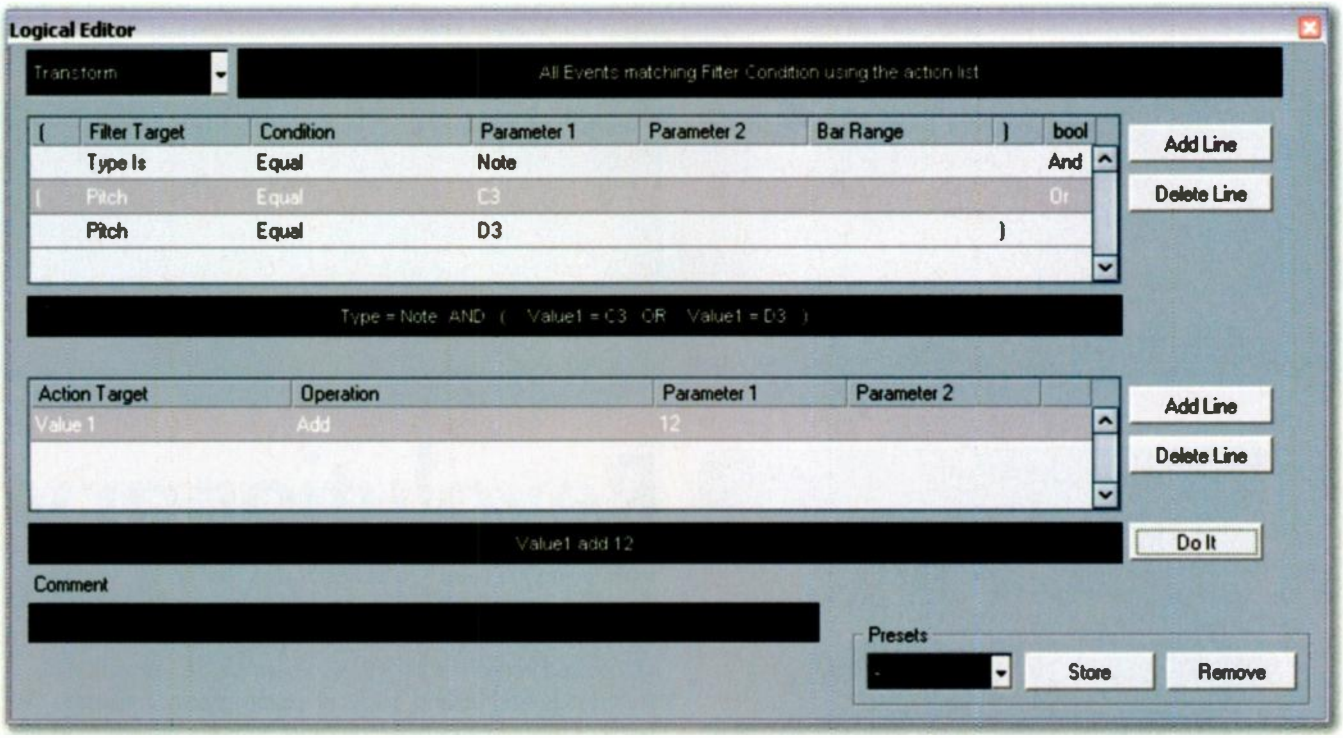
Using The Logical Editor

In Cubase SX & SL



The Logical Editor supports simple Boolean operators to facilitate the building of more complex expressions in the Filter Condition List.

This month we delve into the Boolean operations in the Logical Editor, offer some tips for entering numeric values in the Transport Panel, and take a first look at *Cubase* running on dual-processor, dual-core systems.

Mark Wherry

In January's edition of *Cubase* Technique we began an in-depth look at *Cubase's* Logical Editor, a powerful way of processing MIDI data using logical operations. Although we were able to carry out some fairly advanced functions when combining Logical Editor Presets with Key Commands and Macros, the actual conditions we were using to select certain MIDI Events in the Filter Condition List were fairly straightforward. However, in order to offer more complexity in how each line of the Filter Condition List is used to build an expression, you can use Boolean algebra to build sets of conditions and express how one set relates to another.

If the thought of reading about Boolean algebra in *Sound On Sound* is somewhat unappealing, don't worry: the amount you need to know to make use of the Logical Editor is minimal. However, should you want to learn more about this subject, an easy-to-read book I'd recommend that doesn't require a degree in discrete mathematics is Charles Petzold's *Code* (ISBN: 0735611319), which covers this and many other related topics.

Tell Me The Truth

At a basic level, Boolean algebra is all about truth: is a given condition true or not? In the case of the Logical Editor, consider the condition 'type is note': if you have a MIDI Part selected containing both MIDI note and controller Events, this condition will only be true for Events that are notes, so only the notes would be selected. Now consider adding a second condition, as we did previously: 'pitch is C3'. Because we want to select only the notes that equal C3, we need both conditions to be true, which requires the use of the AND operator from Boolean algebra. So the complete condition would be 'type is note AND pitch is C3'.

At this point, you might have noticed the Filter Condition List has a 'bool' column for Boolean operators, and by default *Cubase* has added an AND after the first condition. The Boolean operator used to join conditions can either be AND or OR, and you toggle between these by clicking on the operator you want to change in the Filter Condition List. When using the OR operator to associate two conditions, an expression is true if either one of the conditions is true, which we'll look at in more detail in just a moment.

The Filter Condition List also supports sets, so you can use open and close brackets to indicate a set within an overall expression via two columns, one on either ▶

side of the Filter Condition List, that enable you to specify the state of the open and closed brackets by clicking in the relevant space and choosing brackets from a pop-up menu. By default, one set of brackets encompasses the entire expression, and you'll notice an open bracket (the left column) in the first condition and a closed bracket (the right column) in the second condition. This is actually explained more clearly in the status line below the Filter Condition List, which for the 'find notes that are C3' example will read: '(Type = Note AND Value1 = C3)', clearly showing the use of Boolean algebra.

To explain this a little more practically, let's say we want to build a condition that selects notes that are either C3 or D3. Since this condition has three expressions ('type is note', 'pitch equals C3' and 'pitch equals D3'), we need three Lines in the Filter Condition List, and the crucial thing here is how these expressions are grouped with Boolean operators. There are two possible input expressions: a note that's C3 and a note that D3, so you could express this in the Filter Condition List in four rows with two sets of brackets: (Type Is Equal Note AND Pitch Is Equal C3) OR (Type Is Equal Note AND Pitch Is Equal D3). If either expression is true for an Event, that Event will be processed.

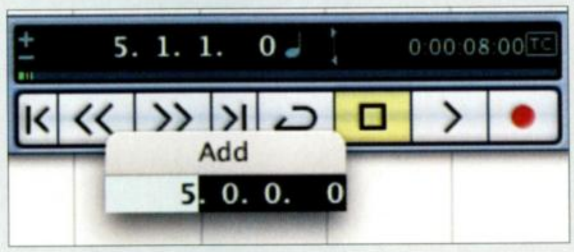
However, I mentioned we can do this in three lines, and since in both of the two expressions described above there is the Line 'Type Is Equal Note', we can factor this out so we end up with just three lines: Type Is Equal Note AND (Pitch Is Equal C3 OR Pitch Is Equal D3). Pretty neat. And another nice thing is that if you make a mistake when setting up the grouping of a condition with brackets, the status line below the Filter Condition List will read 'Syntax Error', instead of displaying the full condition.

Transport Trickery

A tip that we previously covered in *Cubase* Technique was that to enter the position of the locators on the Project Cursor numerically, you can press either Shift+L, Shift+R, or Shift+P to trigger the Input Left Locator, Input Right Locator or Input Position Key Commands respectively, enter the time (using the left and right cursor keys to navigate between the different parts of the location value) and press Return. However, once you have triggered one of the Input transport commands, you can press '+' on the numeric keypad to open the 'Add' text input window, enabling you to enter a time value that will be added to the current time.

For example, if the Project Cursor is at 5.1.1.0 and you press Shift+P (Input Position), followed by '+', type in 5 and press return, the Project Cursor will be set to 10.1.1.0. Although you can't press the '-' key on the number keypad in the same way to enter a negative value, you can enter a negative value in the Add window that appears when you press the '+' key instead. This method works for the Left and Right Locators as well, and while the Secondary Time Display doesn't have an 'Input' Key Command, if you are editing this Time Display you can still use the addition function we've been describing.

The only caveat to the Add feature is that it doesn't work correctly when the Project is playing back and you use the Input Position command to modify the Primary Time Display. You can carry out the instructions as described, but there will be no effect on the position of the Project Cursor, although you can make additions to the Left and Right Locators — and the Secondary Time Display — and this will work as expected. Interestingly, the addition command will work for the Primary Time Display if you drag the values in the Add window with the mouse instead of entering them with the keyboard, but this is far less convenient. Hopefully this will be fixed in a future version.



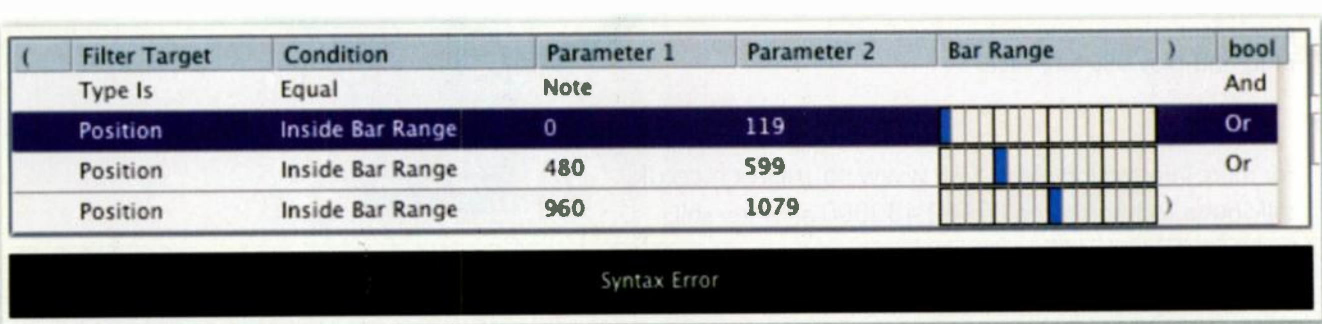
The Add window enables you to add an amount of time to the current position of the Project Cursor and Left and Right Locators if you press '+' on the numeric keypad when editing a time value in the Transport Panel.

Easy Accents

Let's look at another example where the Boolean operators can come in handy, and the idea for this one came from the same composer who needed to readjust velocity layers in one of the previous column's examples. Say you have a repeated semiquaver sequence in a 4/4 bar and you want to accent the first semiquaver beat of every crotchet (beats 1/16, 5/16, 9/16 and 13/16); normally you'd select the notes falling on these beats and increase the velocity for these selected notes. But it can be really tedious to select these notes manually — especially if the basic rhythm of the sequence continues for many bars — and while you could investigate the groove quantise options to make this process easier, building a Logical Editor preset to select the appropriate notes is much more flexible because you can do more to the selected notes once they're selected than just change the velocity.

Start with the 'Init' preset in the Logical Editor window again, giving you a Filter Condition List Line with 'Type Is Equal Note', and set the Function to Select. Remove the brackets from the first Line in the Filter Condition List by clicking the open and close bracket columns and selecting 'All Off' and add four lines (for each beat/area of the bar we want to specify), setting the Filter Target to Position and the Condition to 'Inside Bar Range'. Next, put a single open bracket before the second Line and a closed bracket at the end of the fifth Line, and set the bool column to the second, third and fourth Lines to OR by selecting the appropriate Line and clicking on AND in the bool column to toggle it to OR. The 'Inside Bar Range' Condition allows the bar positions you specify to apply to every bar, as opposed to specifying absolute positions, such as bar one to bar two.

Notice how the Bar Range column becomes active, giving you a visual



(	Filter Target	Condition	Parameter 1	Parameter 2	Bar Range	)	bool
	Type Is	Equal	Note				And
	Position	Inside Bar Range	0	119			Or
	Position	Inside Bar Range	480	599			Or
	Position	Inside Bar Range	960	1079		)	

If you make a mistake with Boolean operators in the Filter Condition List, the status line below will report a 'Syntax Error' instead of displaying the full expression. Can you spot the mistake here?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Rewiring Reason

Using *Reason* With Major Audio Host Applications

***Reason* makes an ideal 'synth rack' for use with a variety of host software. We take you through the steps needed to Rewire it to eight popular programs.**

REWIRE

Craig Anderton

Propellerheads' Rewire is a connection protocol that allows two or more software applications to work together like one integrated program. That sounds simple enough, but it means a lot to today's computer-based musician.

For example, imagine you've created an amazing rhythm track in *Reason*, but want to add some vocals and guitar as overdubs. Because *Reason* doesn't record linear digital audio tracks, if Rewire didn't exist you'd need to export the *Reason* stuff as audio, import it into a MIDI + Audio sequencer and do your best to match the program tempo with the *Reason* file's existing tempo as you did your overdubs. If you then wanted to make a change in a *Reason* instrument, you'd have to export the *Reason* file, then import and so on all over again.

With Rewire, you can use *Reason* as the client (also called the synth application or slave) with a compatible host (also called the mixer application) such as *Pro Tools*, *Sonar*, *Digital Performer* or Ableton *Live*. Both programs will follow the same tempo while you lay down your audio tracks. There's a misconception that Rewire is a CPU-intensive protocol, but that's not the case. Rewire itself is quite efficient; what causes the hit to your CPU is running two audio/soft-synth programs at the same time. As long as your computer can handle running these two programs at the same time, Rewiring them together shouldn't cause any problems.

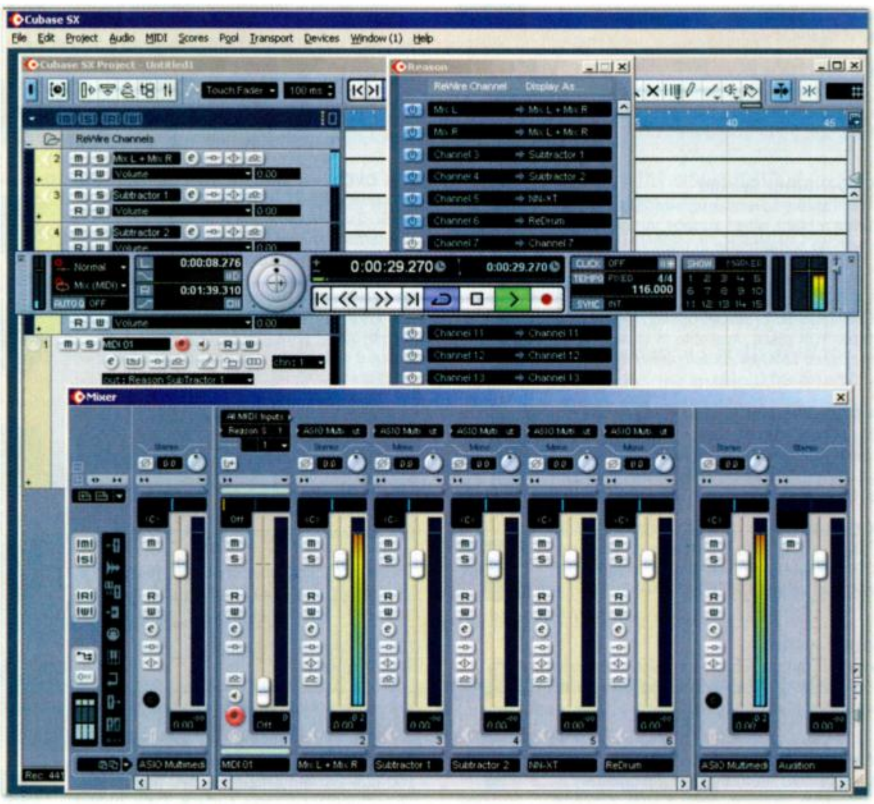
Rewire Basics

Any Rewire-compatible application is either a host, a client, or both (but not simultaneously — you can't Rewire a client into a host, then Rewire that into another host). There are four main aspects to Rewire:

- The client's audio outputs stream into the host's mixer.
- The host and client transports are linked, so that starting or stopping either one starts or stops the other.
- Setting loop points in either application affects both applications.
- Both applications can share the same audio interface.

The original version of Rewire allowed streaming of up to 64 individual channels into the host's mixer; Rewire 2 ups that to 256. You may have the option to choose only the master mixed (stereo) outs, all available outs, or your choice of outs. If you choose all available outs, instruments can Rewire into the host's channels individually, and be processed individually. (With Rewire 2, it's also possible to stream 4,080 individual MIDI channels — 255 MIDI buses with 16 channel per buss — from one application to another.)

Another aspect of Rewire is that programs ▶

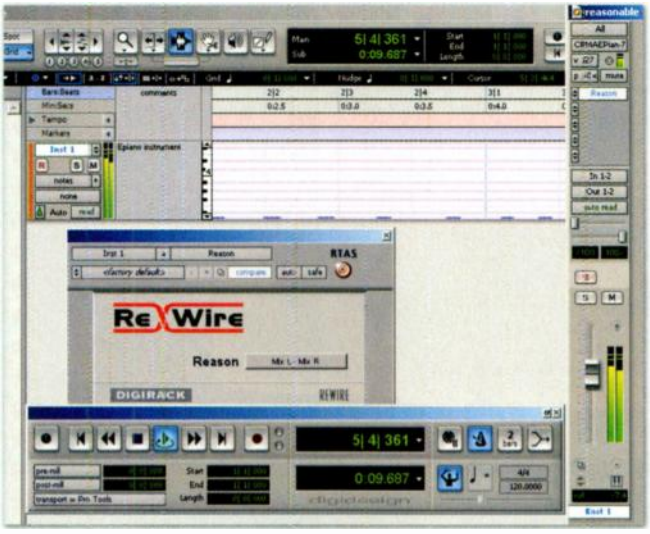


Cubase SX's *Reason* device panel is toward the upper centre; the mixed output and four other channels are active. *Reason*'s selected instrument outs show up in the Arrangement window as part of a *Reason* folder track, as well as in the mixer. The *SX* MIDI track drives whichever instrument is selected as its output.

must be opened in a particular order, usually host first, then *Reason*; however, some programs will automatically launch *Reason* if you select it as a client. You close programs in the reverse order: *Reason* first, then the host. Now let's look at how to Rewire *Reason* into various host programs.

Steinberg *Cubase SX3*

1. Go to the Devices menu, which lists various Rewire-compatible applications.
2. Select '*Reason*' and a Rewire panel appears with *Reason*'s 64 available channels (see screen shot on previous page). Click the buttons toward the left to enable the channels you want (you can rename channels in the right column, so names that are relevant show up in *Cubase*'s mixer).
3. Open *Reason*.
4. The channels you activated in step two appear in the mixer. In the Arrangement window, they're in a folder track called 'Rewire Channels'.
5. Create a MIDI track, and in the track's 'Output' drop-down menu select the *Reason* instrument you want to trigger via MIDI. Now any MIDI data you play into *Cubase*, or record into the associated *Cubase* track, will trigger that instrument.



The Instrument track introduced in *Pro Tools 7* contains MIDI data and outputs audio from the Instrument. It's also the ideal place to insert *Reason* outputs.

Digidesign *Pro Tools 7*

Pro Tools 7 introduced the Instrument track, which simplifies Rewiring, compared to previous versions (see screen shot above).

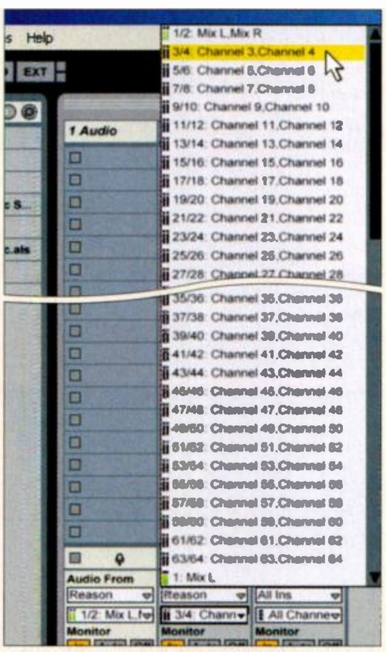
1. Go Track / New and select a Stereo Instrument Track.
2. Click on 'Create', which inserts the Instrument track into your session.
3. In the Mix window, go View / Mix Window and tick 'Instruments'.
4. In the Instrument track's insert section, go Multi-Channel Plug-In / Instrument / *Reason*.
5. *Reason* will launch automatically. A Rewire window appears where you can choose the desired *Reason* output.
6. To send MIDI data to a *Reason* device, first, in the Mix Window Instrument track's topmost field, choose the desired MIDI input that's providing the MIDI data.
7. Choose the destination *Reason* instrument in the field below the MIDI input field. All available destinations will be named, along with associated MIDI channels.

You can now record MIDI data into the Instrument track, which will feed the destination you selected in step seven above. Also note that you can create multiple Instrument tracks and insert *Reason* in each one, while choosing different outputs from *Reason* for these tracks.

Ableton *Live 5*

***Reason*'s outputs flow into the *Live* mixer channels as if they are inputs you're recording.**

1. **If you have empty tracks available for *Reason*'s outs, that's fine. Otherwise, go Edit / Insert Track and insert the required number of tracks (remember, *Live*'s tracks are stereo).**
2. **Open *Reason*.**
3. **Once *Reason* is open, click on *Live*'s 'Audio From' field to see a pop-up menu showing all available Rewire clients. Select '*Reason*'.**
4. **Next, click on the field just below the 'Audio From' field, to reveal a pop-up menu showing all *Reason*'s available outputs. Choose the desired output for the current track. Note that you can create multiple tracks, select *Reason* in the 'Audio From' field, then choose several outputs from *Reason*.**
5. **Click on the 'In' button in the Monitor section for each Rewired track, so that you can hear the Rewired signal. Note that the channel meters will indicate *Reason*'s levels. You can also process *Reason*'s signals through *Live*'s effects, as well as sending them to any aux effects via the send controls.**
6. **Assuming that the 'MIDI From' parameter is set to receive data from your MIDI controller, set 'MIDI To' to *Reason*.**
7. **In the field directly below 'MIDI To', select the Reason instrument you want to feed with the MIDI data.**



Reason has been selected in the 'Audio From' field. The middle track is an audio track that is receiving the output from *Reason* channels three and four. The track on the right is a MIDI track, which is sending MIDI data to *Reason*.

MOTU *Digital Performer 4*

As usual, load the host first. However, *DP* already 'knows' that Rewire applications are present, even before you load *Reason*.

1. Go Project / Add Track / Aux Track, to create an Aux track.
2. In the track Input field, use the pop-up menu to choose the *Reason* output you want to feed into *DP*'s mixer (see the screen shot overleaf).
3. If you want to add several sets of *Reason* outputs, create more Aux tracks and continue as above.
4. Now open *Reason*, and press Play on either application. Assuming that you have a project in *Reason*, you should hear *DP* play along with it, and the meters will indicate that *Reason* is feeding audio into *DP*'s mixer.
5. *Reason* will also appear as MIDI output destinations in the MIDI output-assignment menus in *DP*'s MIDI tracks, so you can play *Reason*'s synths from MIDI tracks within *DP*, or record data in those tracks for driving *Reason*'s instruments.

Also note that Rewire MIDI ins and outs are published to Core MIDI-compatible software, so the client can receive MIDI data from such software as well as transmitting data to it.

Cakewalk *Sonar 5*

1. Go Insert / Rewire Device and you'll see a list of all registered Rewire-compatible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REWIRING TO AUDIO HOSTS

Assigning a *Reason* output to the Input field in an Aux track (circled in red) feeds the output into *Digital Performer*'s mixer.

devices. Select '*Reason*'.

2. A dialogue box will pop up with check boxes for 'First Synth Output' or 'All Synth Outputs'. The former chooses *Reason*'s mixed stereo outs, while the latter presents all available outs. Be careful if you choose 'All Synth Outputs', as *Reason* will add 64 channels to *Sonar*'s mixer. You can always delete the ones you don't use, but still, unless you need individual outs go for the main stereo outs.
3. The dialogue box also asks if you want a MIDI source track. This is necessary to send MIDI data to *Reason* (eg. for controlling it via a MIDI keyboard). Of course, you can insert a MIDI track later (and you probably will, if you want to drive several devices from different tracks), but if you do it now it will be named automatically and ready to go. You can also check 'Open Synth Property Page' if you want to bring *Reason*'s window to the fore, and/or 'Open Synth Rack', which shows Sonar's 'virtual rack' of whatever synth and Rewire devices you have open.
4. Now open *Reason*.
5. In the MIDI Source track, the devices present in *Reason* will be pre-assigned as options and the instrument names will show up as outputs. Choose the device to which you want to send MIDI data (see the screen shot on the right).

Arturia *Storm 3*

1. After opening *Storm*, click on the 'Rewire Software' tab. (If the tab doesn't appear, go Settings / General and tick 'Activate Rewire mixer on launch of *Storm*'.)
2. Drag the *Reason* box under 'Rewire Software' into an empty space in the *Storm* rack.
3. Use the drop-down menus on *Storm*'s Rewire 'rack unit' to assign particular outputs from *Reason* to the *Storm* mixer's right and left channels, which will already have dedicated a track to the Rewire device.
4. If you want to assign more *Reason* outputs to the *Storm* mixer, drag the *Reason* box below the Rewire Software tab over to another blank space in the *Storm* rack. *Storm* will open up another channel; assign the desired *Reason* outs to the new *Storm* channel as described previously, using the drop-down menu.

Adobe *Audition 1.5*

Audition added Rewire support in v1.5. However, it won't pass MIDI data through to the client, so, for example, if you want to play the soft synths in *Reason*, you'll need to record your data directly into *Reason*. Note that you won't need to open *Reason* after opening *Audition*, because *Audition* will do that for you.

1. In the Multitrack view, go Options / Device Properties / Rewire tab.
2. Click on the button 'Enable Audition as a Rewire Host'.
3. Enable *Reason* as the Rewire slave application.
4. Choose the desired track assignment. 'Insert summed stereo output into first available track' brings in the summed output, which needs one track, as *Audition* tracks are stereo. 'Insert all outputs to individual tracks' brings in all *Reason*'s tracks, which is a lot — so you might instead want to tick 'Insert outputs manually using track device input dialogues'. Then you can choose which client outputs show up at which *Audition* inputs.
5. After making your decision, click on Launch and *Reason* will launch automatically.

If you decide to assign inputs manually in step four, when you click on the track Input field you'll need to set the Device Type as 'Rewire'. Then you can specify whether the input will pick up the left channel, the right channel, or both. Once you've made assignments for all your tracks, you're ready to go. (Incidentally, *Audition* won't let you exit the program unless you return to the Device Properties page and disable *Audition* as a Rewire host.)

Reason's mixed stereo outs are inserted as tracks into *Sonar*. The MIDI track is being assigned to the *CCRMA E Piano* device. More MIDI tracks can be added to feed other instruments.

Sony *Acid 5*

1. After you have opened *Acid*, Go Insert / Soft Synth.
2. Now click on the 'Rewire Devices' tab. Doing this will cause a list of Rewire devices to appear, along with their available outputs.
3. Select an output. This creates a Soft Synth channel in the Mixer window. If *Reason* doesn't open automatically, open it manually.
4. Continue to insert *Reason* outputs, if you need more of them. Each of these will create a new channel in the *Acid* mixer window.
5. *Acid* will pass MIDI data to compatible applications. As an example, if you have chosen a MIDI input within *Acid* under Options / Preferences / MIDI, and you are Rewiring *Reason* into *Acid*, clicking on the MIDI icon for a *Reason* channel will allow you to play the associated instrument from whatever is feeding the input of your MIDI interface.
6. If you want to record that MIDI data inside *Acid*, go Insert / MIDI Track, name the track and click on Record. Then fill in the various Record fields, specify where you want to save the data, specify MIDI as the Record Type, and set MIDI Thru to the *Reason* device that you want to control. Click on Start and begin recording. SOS

The REX Connection

Craig Anderton

Using the *RXP* REX-file player in *Sonar 5 Producer Edition*

Now that the *RXP* REX-file player is bundled with the *Producer Edition* of *Sonar 5*, the creative possibilities of REX files have really opened up for *Sonar* users. We take a tour around *RXP* and discuss its features and uses.

Sonar users have enjoyed the ability to import, export, create and edit '*Acid*ised' loops since version 1.0 of the program. The *Acid*isation process was first introduced in Sonic Foundry's *Acid* program (subsequently acquired by Sony), and offered the unique ability to time-stretch and pitch-shift digital audio with reasonable — and sometimes outstanding — sound quality. In version 4.0, *Sonar* added the ability to change pan, pitch, and level for each 'slice' in an *Acid*ised file.

However, another time-stretching file format — the REX file, developed by Propellerhead software — remained out of reach to *Sonar* users until recently. (There has always been the workaround of 'Rewiring' a program that supports REX files, such as Propellerheads' *Reason*, with *Sonar*, but this isn't the same as native support.) Fortunately, *Sonar 5 Producer Edition* now bundles the *RXP* REX-file player, designed by René Ceballos of RGC Audio, thus opening up this alternative looping protocol to *Sonar* users. (Note that *RXP* is not included in *Sonar Studio Edition*.)

The Missing Help

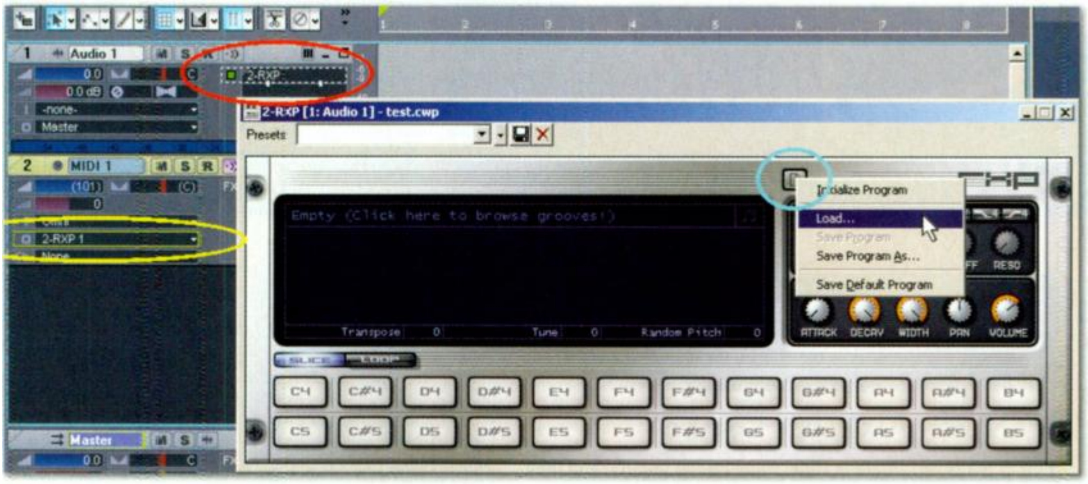
If you search on '*RXP*' in *Sonar's* Help file, you'll find... nothing. Instead, bring the *RXP* player window to the top, click anywhere within the instrument itself, then hit function key F1 to see the Help file.

REX File Basics

The most common tool for REX-file creation is Propellerheads' *Recycle* program (currently at version 2.1), but many sample libraries are available in REX format if you don't want to have to make your own. You bring in a WAV or AIFF file, then create 'slices' at each transient. This is a semi-automated process, but it generally requires some manual editing as well, to optimise the slice points. Ideally, each slice should have a discrete sound: a kick drum, kick + snare combination sound, hi-hat hit, and so on.

When you save the sliced REX file, a MIDI file that triggers each slice is bundled with it. When you bring a REX file into a host program, either a MIDI track will be created ▶

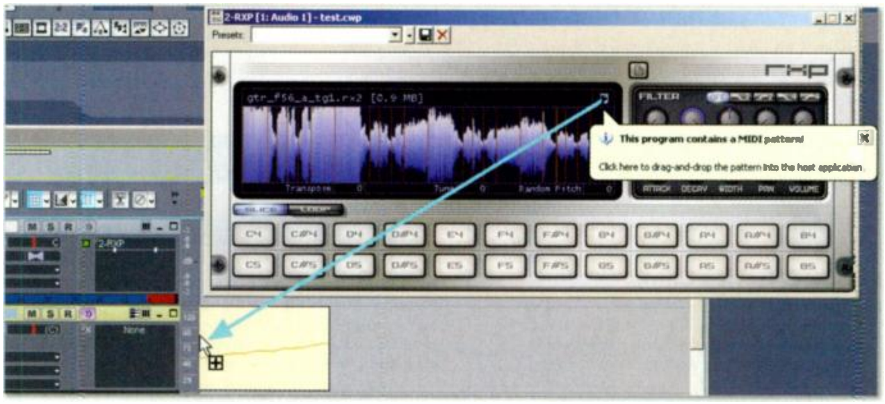
USING THE *RXP* REX-FILE PLAYER



The *RXP* loaded into *Sonar 5*. The insertion into the audio track's FX Bin is circled in red. The companion MIDI track's output has been assigned to the *RXP* (circled in yellow). The Load button (circled in blue) allows REX files to be loaded.

automatically, containing the MIDI portion of the REX file, or you'll need to somehow drag or import the MIDI portion into a MIDI track. *Sonar*'s *RXP* player uses the 'drag and drop into track' option.

When you time-stretch in a host that accepts REX files, slowing down the tempo causes the MIDI triggers to occur further apart, thus triggering the slices at a slower rate, while speeding up the tempo triggers the slices at a faster rate. Because the triggers use MIDI, editing the MIDI notes will alter how the slices are played back, and this offers many creative possibilities. You can copy MIDI notes to trigger several slices at the same time, remove MIDI notes to thin out a loop, and so on.



Upon loading, the REX file waveform appears in the *RXP*'s waveform view window. You can now drag the companion MIDI file into the MIDI track whose output feeds the *RXP*.

Inserting The *RXP*

Although you can use the Insert / Soft Synths command, I don't recommend this, as it makes available only a sub-set of the *RXP*'s parameter-automation options (we'll discuss this in depth later). Instead, do the following:

1. Create an audio track and a MIDI track.
2. Right-click in the audio track's FX Bin.
3. Go Soft Synths / *RXP*.
4. In the MIDI track's drop-down menu, select *RXP* 1 as the MIDI output.

Note that this insertion technique works with many soft synths. If you find you can't arm instrument parameters for automation using the standard Insert command, try it instead (see screen above).

Loading A REX file

When you load a REX file into the *RXP* player, remember that you're loading both digital audio slices and a MIDI file.

1. Click on the *RXP*'s Load button, and navigate to the desired file.
2. The first time you load a REX file during a project, a balloon will remind you that the file contains a MIDI pattern, and point to the note symbol to the lower left of the Load button.
3. Drag the symbol into the MIDI track where you want the MIDI file to begin (see screen, right).

An alternative way to load REX files is via the *RXP*'s virtual LCD. If nothing is currently loaded, you'll see a strip that says 'Empty (Click here to browse grooves)'. Otherwise, you'll see the name of the currently loaded file, and its size. In either case, if you click in it a Groove Browser appears. This accesses content located in C:\Program Files\Cakewalk\Shared DXi\RXP\Contents. There's also a User folder in the Contents folder, where you can store your own favourite REX files.

Sonar MIDI Groove Clips

One *Sonar* feature that's particularly useful with REX files is the ability to create MIDI groove clips. This means that you needn't laboriously copy and paste the MIDI file to cover the length of the section where you want the REX file to play back. Simply convert the MIDI file into a MIDI groove clip, then click and drag the clip's right edge to the right, thus 'rolling out' the clip and creating as many 'iterations' as you like. The clip's corners change from right angles to

Sonar News: v5 Demo Now Available

Still on the fence about whether you want *Sonar 5* or not? Find out for yourself with the web demo.

This version of *Sonar Producer Edition* 5.0.1 has several limitations, as detailed on the Cakewalk web site. You can't save files, put data on the Windows clipboard, export audio, *Acid* files, videos, or MIDI groove clips, or bounce to clips or tracks. There are several other limitations too, which are basically designed to allow you to work with the *Sonar 5* demo but not really preserve the results. Furthermore, several instruments and plug-ins aren't included.

However, you do get trial versions of 11 Sonitus Direct X plug-ins, the *Cyclone* and *Dreamstation* DXi2 synths, the *TTS1* DXi (although it stops functioning after 10 minutes and beeps every 10 seconds), the *Pentagon I* and *PSYN II* DXis, an SFZ Soundfont sample player, and eight Cakewalk MFX MIDI plug-ins. There's also the option of downloading a variety of content, including templates, to make it easier to evaluate the system. (Note that you need Direct X 9 installed on your computer for the demo to work.) For more info, as well as details about downloading, visit http://www.cakewalk.com/Support/kb/kb2005298.asp.

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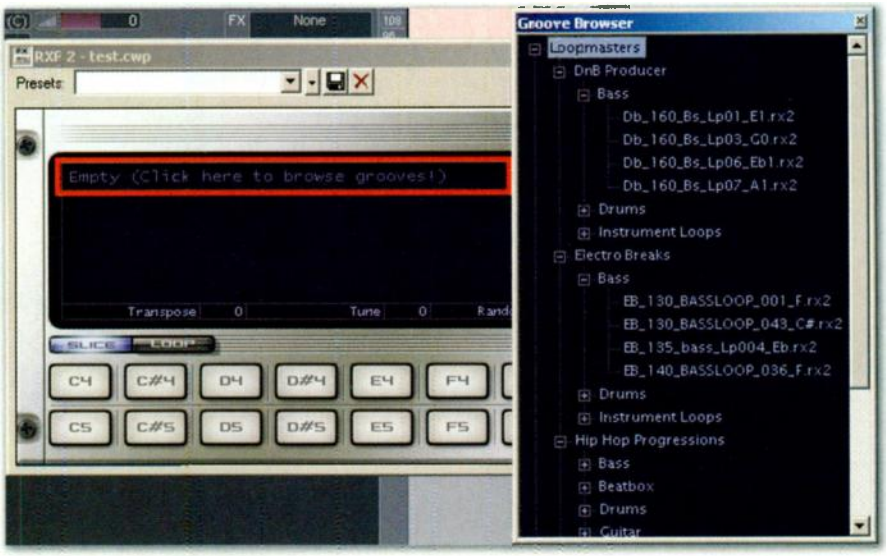

Click in the area highlighted in red to call up the Groove Browser. This shows the content that was included with *Sonar 5 Producer Edition*, as well as any REX files you've put in the User folder.

rounded corners, to indicate its groovy status.

One useful aspect of groove-clip looping is that if you want to edit the rolled-out loop, each iteration is editable individually. In other words, if you move some notes around in the first iteration, subsequent iterations won't reflect these edits. However, if you do want one edit to affect other iterations, no problem. 'Rolling into' the original-but-now-edited groove clip, then rolling back out again, causes the subsequent iterations to reflect any edits you made to the original clip. Rolling part-way toward the edited groove clip, then rolling back out again, makes the sections you rolled over reflect edits made to the original clip. However, those areas that weren't rolled over will remain as they were. Finally, if you turn off groove-clip looping for the clip (click on the clip and type Ctrl-L), then turn it back on again, subsequent iterations will reflect any edits you made to the original clip.

LCD Editing Options

There are several edits you can do in the *RXP*'s LCD window, without turning to its other controls. We'll cover the latter shortly.

- **Preview** a slice: Click on the slice you want to hear.
- **Rearrange** the slice order: Click on any slice and drag it elsewhere in the display.
- **Reverse** the slice order (the first slice becomes last and the last becomes first): Right-click on the waveform and select Reverse.
- **Randomise** slices: Right-click on the waveform and select Randomise.
- **Restore** all slices to their original positions after you've messed around with them: Right-click on the waveform and select Reset.
- **Transpose** the entire file in semitones: Click on the Transpose numerical and drag up or down to transpose up or down, respectively (up to ±48 semitones). Double-clicking on the numerical resets it to zero.
- **Tune** the entire file in cents: Click on the Tune numerical and drag up or down to transpose up or down, respectively (up to ±100 cents). As above, double-clicking on the numerical resets it to zero.
- **Randomise pitch** of each slice: Click on the Random Pitch numerical, and drag up to select a range of cents (up to 4800) over which slices will be transposed. Drag down to lower the range and double-click to remove random pitch effects.

Amplitude Editing Options

These controls, found to the right of the 'LCD', affect all slices globally; in other words, you can't edit individual slices, as you can with the *Dr. REX* module in *Reason*. Note that double-clicking any control restores it to its default setting.

- **Attack:** Changes the amplitude attack for each slice, from zero to 1000ms.
- **Decay:** Alters the decay for each slice, from zero to almost 1000ms. Going further selects infinite decay. With short decays, this control can create extremely percussive effects. For example, suppose you bring a white-noise file into *Recycle*. You can then add slices to create a certain rhythm. Save the file, bring it into *RXP*, then use the Decay control to create a percussive white-noise pattern.
- **Width:** Varies the stereo spread, from zero percent (mono) up to 100 percent (the maximum stereo width that is allowed by the file).
- **Pan:** Modifies the placement of a mono signal (or stereo signal converted to mono using the width control) from left to centre to right in the stereo image. With a stereo signal, this acts like a balance control. In other words, when it's set to 100 percent (full right), you'll hear only the right channel; when it's set to -100 percent (full left), you'll hear only the left channel. Other settings change the balance of the two channels.
- **Volume:** Sets the overall *RXP* output level.

Filter Editing Options

This section includes a multi-mode filter and associated envelope, the latter being re-triggered with every slice. As with the amplitude-editing options, any processing affects all slices globally.

- **Filter Slope:** Set by the four buttons in the upper right of the filter control cluster. Normally the filter is off unless you select one of these buttons or click the Off button to turn it off. From left to right, the buttons select two-pole low-pass, two-pole high-pass, four-pole low-pass and four-pole high-pass filter types.
- **Attack:** Changes the filter-envelope attack time for each slice, from zero to 10000ms.
- **Decay:** Changes the filter-envelope decay time for each slice, from zero to almost 1000ms, then to infinite decay.
- **Env:** Sets filter-envelope amplitude. When centred, the envelope has no effect. Turning clockwise adds positive modulation, while counter-clockwise adds negative modulation.
- **Cutoff:** Determines the initial filter-cutoff frequency, from 8.2Hz to around 22kHz. This is also the initial frequency to which the envelope is applied.
- **Reso:** Sets the filter resonance. The filter won't oscillate, but it can come close.

Again, as with the amplitude controls, double-clicking on a control resets it to the default position.

Automation

When you insert the *RXP*, by going Insert / Soft Synth, automation options (using MIDI track envelopes) are limited to reverse (Continuous Controller 1), pan (CC 10), expression (CC 11) and volume (CC 7). However, expression and volume seem to do the same thing and pan appears to have no real effect. Reverse, however, is pretty cool. It's switched so that values of 64 or below don't reverse slices but values above

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USING THE *RXP* REX-FILE PLAYER

64 do. Even better, you can do this in real time! In the Console view with *Sonar 5 Producer Edition*, you could set up Continuous Controller 1 on a fader.

Fortunately, when you insert the *RXP* manually into the FX Bin, as mentioned at the beginning of this article, you can access virtually all relevant parameters via either real-time control (tweak the *RXP* knobs) or MIDI envelopes. Here's how to set up for real-time control.

1. Right-click on the *RXP* label in the audio track's FX Bin.
2. Select Arm Parameter (see screen, right).
3. Tick the parameters for which you want to record automation.
4. Click on OK.
5. Click on the Transport's Record Automation button.
6. As the sequence plays, move the knobs that correspond to the parameters you selected.
7. On playback, the knobs will re-create your moves.

Right-clicking on the *RXP* label in the FX Bin, then clicking on 'Arm Parameter' readies parameters for automation.

To add automation using envelopes:

1. Right-click on the audio track where the *RXP* is inserted and go Envelopes / Create Track Envelope / *RXP*.
2. Tick the boxes that correspond to the envelopes you want to create.
3. Click on OK.
4. The envelopes you requested show up in the track.
5. Edit the envelopes as desired.

If you're automating using this method, bear in mind that Reverse is not available as an automatable parameter. Instead, you need to add CC1 messages in the MIDI track driving the *RXP*.

The Pad Section

We've just about finished our guided tour of the *RXP*, but there are a couple more important aspects of the device to discuss: Slice and Loop modes (as selected with two buttons just below the left side of the LCD window), and how they relate to the 24-strong pad section along the bottom of the *RXP*.

In Slice mode, triggering a pad plays the corresponding slice (the first pad triggers the first slice, the second pad triggers the second slice, and so on). You can also trigger the pads from a MIDI keyboard, where note C3 triggers pad number one, C#3 triggers pad number two, and so on until C5, which triggers pad number 24. If the REX file contains more than 24 slices, you can't trigger more than 24 with the pads, although they will still respond to MIDI notes above C5.

This mode is handy for a variety of purposes, one of which is vocals. You can throw together strings of words or phrases, bring them into *Recycle* and slice each word individually. After saving the file and bringing it into the *RXP* player, you can then 'play' the words with the pads. (This recalls *Sonar*'s *Cyclone* DXi instrument; however, *RXP* offers eight more pads, which may be important in the case of vocal phrasing.)

In Loop mode, clicking on a pad plays the loop through once. However, each pad transposes the loop by a certain amount, starting at -12 semitones and going up to +11 semitones. Note that the tempo doesn't change; we're dealing with pitch transposition only. As with Slice mode, the pads can still be triggered by MIDI notes, although triggering starts with MIDI note C1 for pad number one and ends with MIDI note B3 for pad number 24.

*Acid*isation Vs REX

Do we really need two time-stretch protocols? Yes, because they're quite different. *Acid*isation uses DSP-based stretching, sample removal and crossfading. REX files cut a piece of digital audio into slices, then vary when those are triggered. The REX process's main use of DSP is to synthesize a decay when gaps appear between slices, which happens when playing back a file at a slower tempo. With percussive sounds, REX files have the potential for higher fidelity, as the sound is essentially unaltered, except for the decay effect mentioned above. However, sustained sounds don't lend themselves to the REX process, as they can't be sliced without causing discontinuities in the sound. *Acid*isation's ability to crossfade covers up discontinuities in sustained sounds and delivers far smoother results.

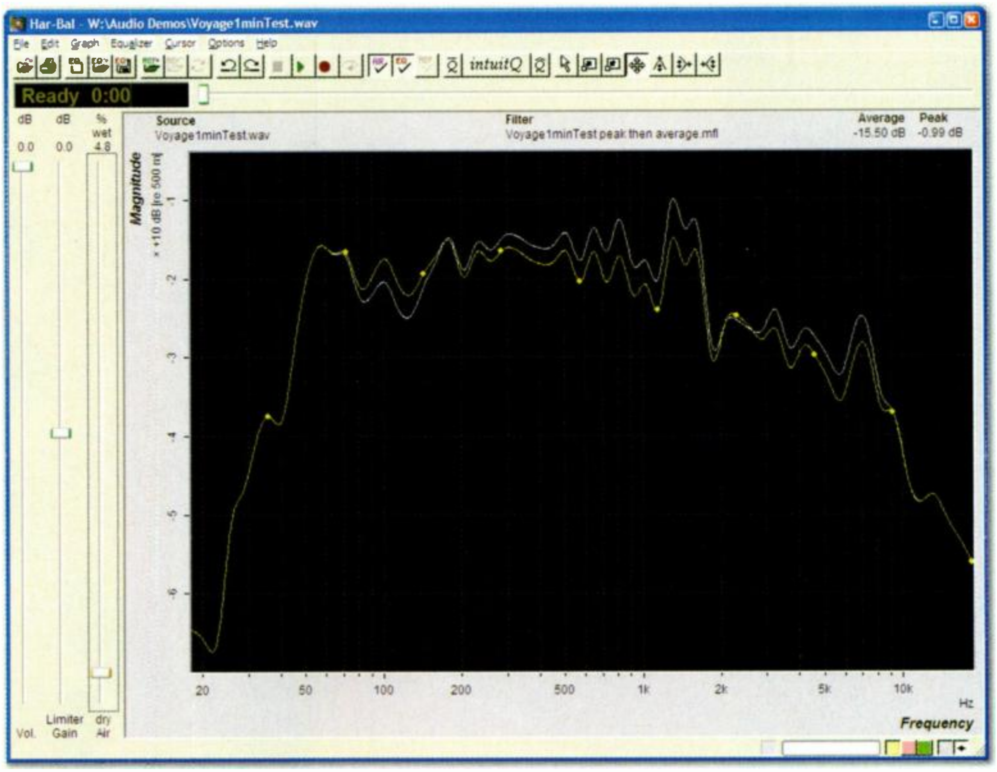
When altering pitch, both are somewhat compromised, and they work best for time-stretching. With either one, fidelity is better when speeding up compared to slowing down.

It's Not Just About REX

Although the *RXP* player is optimised for REX files, it also accommodates other file types. I loaded up a WAV file of a bass note, looped at the end for use with samplers and, surprisingly, I was able to play it like a sampler in Slice mode, with different pads playing back different pitches. The *RXP* even recognised and played back the loop, and of course I could warp the sound with the onboard filter, amplitude and tuning controls. Granted, anyone who has *Sonar 5 Producer Edition* probably already has a sampler, or could use the *Cyclone* DXi to do much the same thing. But as a quick playback engine for samples, *RXP* is undeniably useful. It will also load AIFF, OGG and WAV files without loop points (they're treated as a single sample, as there's no slice info) and imports SFZ-format files too. Don't have any SFZ samples to try out? You might if you own Cakewalk's *Dimension* or *Dimension Pro* synth. Look around in the *Dimension*'s Multisample files folder and you'll find some SFZ files suitable for experimentation. SOS

Can *Harbal*'s new automatic track EQ solve your equalisation problems with one click? We find out, as well as exploring the mysteries of the Windows Driver Model...

Martin Walker



For instant mastering EQ gratification, take a look at *Harbal 2.0*. Here you can see how the new one-click IntuitQ function has automatically removed a low-end hole between 80 and 150Hz and reduced a slightly prominent mid-range area (the white trace is the 'before' spectrum and the yellow one shows the results after IntuitQ has been applied).

Back in *SOS* March 2004, I reviewed the PC-only *Harbal* (*Harmonic Balancing*) application, which provides a sophisticated way to add final tweaks to the EQ of otherwise finished tracks at the pre-mastering stage, by referencing them against well-mixed tracks in various genres. Unlike the majority of 'EQ rippers' that attempt to match the frequency content of a destination track to that of a source (often resulting in huge boosts at some frequencies, to compensate for differences in instrumentation or key changes), *Harbal* encourages a more considered approach.

I was impressed with its utilitarian yet highly usable and informative interface, and by how it allows users to learn by experience, compare their own mixes with commercial releases and educate their ears. The dedicated *Harbal* forums also provide users with plenty of feedback, plus techniques from professional audio engineers.

Nevertheless, some musicians still prefer instant gratification, so *Harbal*'s developers have donned their thinking caps and have spent some time working on a way to provide an automated solution to EQ problems. The result is *Harbal* 2.0, which incorporates two new features: IntuitQ and HB Air.

One-click Mastering?

IntuitQ is essentially a one-click solution for EQ problems, yet it still avoids the trap of blindly forcing a track's frequency content to match that of some notional 'perfect response'. Instead, a single click on the 'Apply IntuitQ' button initiates a two-stage process that calculates the most appropriate modifications from the track itself, by reducing any prominent frequency bands and filling in any major holes that it finds.

The first stage smooths the average response to minimise masking effects (so that each instrument gets the best chance of being heard) and the second deals with any dominant peaks, to reduce harshness. You can also apply these stages individually, if you prefer, using the Average and Peak buttons situated on either side of the main IntuitQ button. The end result will be identical.

What IntuitQ won't attempt to do is change the basic sound of your mixes, nor try to completely iron out their characteristic peaks and dips, which is exactly as it should be — after all, if you've spent a long time trying to achieve a particular sound, you don't want it messed with unduly. So if your track is already well mixed and balanced, it's likely to receive only minor changes, while material that needs a bigger helping hand should get it: a boomy bass end might be reduced, a honky mid range or harsh top end improved, or frequency ranges boosted if they're too low in the mix.

This is exactly what a professional mastering engineer should do, and although no automated EQ algorithm is likely to produce perfect results every time on all types of material, I think the results are extremely good. *Harbal*'s developers recommend trying *IntuitQ* first, then using the result as a basis for further artistic tweaks. But, judging from the majority of tracks I tried it on, I suspect that many people will end up relying totally on IntuitQ. It gave me impressive results on a wide range of unreleased material, mostly providing clearer and better balanced mixes, even though the changes made were often of only a few dBs.

It's also great fun (and educational) to apply IntuitQ to commercial tracks, to see how much it changes them. On most of the commercial tracks I tried it with, tweaks of no more than a couple of dBs up or down, across two or three frequency bands, were made. A few tracks were altered by up to 5dB at one or more frequencies, mostly where the peaks created by a prominent snare sound dominated the mix.

I didn't always agree that the tweaked commercial tracks ▶

Tiny Tip

It's often very handy to be able to select a whole bunch of files in Microsoft's Explorer and compress them into one by right-clicking on them and selecting the 'Send To Compressed (zipped) folder' option. However, many people haven't twigged how the file name is chosen for the resulting zip file. Once you've got your selection of files highlighted, simply point to the one whose name you'd like Windows to use, then use the right-click command, and your zip will use the main part of this file name, but will replace its extender with '.zip'.

sounded better, but one of *Harbal*'s developers is a mastering engineer, so the current algorithm will, in part, be based on his personal artistic decisions. However, I think that *Harbal*'s goal "to automatically design near-optimum equalisation filters for any given track" has been met admirably. The algorithm only really works on complete tracks at the moment (it over-compensates somewhat on individual instruments or vocals), but it's an excellent tool to help during mixing and pre-mastering.

IntuitQ largely removes the need for the reference files in different genres used in previous versions of *Harbal*, although these are still useful for the Match Loudness function, so you can import another track (from your proposed album, or a commercial CD), create a reference file from it and then use it to establish a 'base camp' level for your other tracks. *Harbal* also has a built-in limiter to help you achieve louder tracks without 'squashing' dynamics.

The other main new feature in *Harbal 2.0* is HB Air. This doesn't, as its name suggests, add exotic HF equalisation or upper harmonics. Instead, it uses the well-known stereo widening technique of mixing the 'L minus R' difference signal back into the mix to enhance its stereo width and make it sound more spacious. You have to be careful not to overdo it and cause harshness. I found that five or 10 percent on the amount slider was quite sufficient for opening out the mix a little, and using the In/Out button helps you to judge the optimum setting.

Harbal 2.0 also incorporates several smaller improvements, including a 'loop all' playback function, and various new keyboard shortcuts, but for me IntuitQ is the star of the show. Overall, version 2.0 is a huge step forward, since it not only provides the means to improve the sound of your mixes, but also has a very good stab at doing this automatically. I can see IntuitQ being controversial in some mastering circles, but in my opinion the results speak for themselves. *Harbal 2.0* is £57.75 to new users, and the update is free from www.har-bal.com, for existing users.

Windows Driver Model

I've noticed quite a few queries recently on the *SOS* PC Music forum that ultimately boil down to 'which is the most suitable driver type' for a particular audio application. After all, you can often get two or three options, and this can result in a lot of confusion, especially since Microsoft's WDM (Windows Driver Model) is not, as many musicians think, a type of driver, but a driver 'architecture'. It actually uses APIs (Application Programming Interfaces) to provide MME (Wave) and Direct Sound drivers, both with fairly high audio latency, along with KS (Kernel Streaming) support, which has much lower latency.

If you're already using the MME-WDM drivers of your audio interface and you attempt to open *Sonar* with its WDM/KS driver mode activated, you'll get an error message like the one shown here.

Both MME and Direct Sound provide multi-client functionality (so you can run several Windows audio applications, such as Media Player and Sound Recorder, simultaneously), support mono, stereo and multi-channel audio formats, play back MIDI files using the supplied Microsoft GS Wavetable SW Synth, and play back sample rates not directly supported by the audio interface. The last trick is accomplished by behind-the-scenes sample-rate conversion, which is why it's often safer to disable system sounds on a music PC, to prevent conversion cutting in without your knowledge.

This is all clever stuff, but it comes at a price. Mixing all these audio streams together relies on the 'KMixer' (Kernel Mixer), that requires at least three 10ms audio buffers. Thus MME drivers (normally displayed inside audio applications as MME-WDM, MME, Multimedia or Wave) normally impose at least 30ms of extra latency.

Some interface manufacturers, such as Echo, provide a special driver option that bypasses this Kernel Mixer to connect directly to the hardware beneath. Echo call their option 'Pure Wave', but although Direct Sound support is disabled as a result, you get a Pure Wave buffer setting that goes right down to a much lower 64 samples. For instance, inside *Wavelab 5*, the lowest setting I can manage with the standard MME-WDM drivers for my Echo Mia soundcard is about 120ms, but after switching them to their Pure Wave setting of 64 samples I can achieve 6ms.

Such special options are great for anyone who wants lower-latency Wave drivers for use with applications that don't support ASIO, like some versions of *Acid*, *Cakewalk Pro Audio*, *Samplitude*, *SAW*, *Sound Forge* and *Vegas*. However, if you're running an application, such as *Sonar*, that provides WDM/KS support, which also bypasses the Kernel Mixer, it's better to stick with the standard WDM drivers provided by the interface manufacturer and rely on the application to ensure low latency.

Bypassing the Kernel Mixer does have other implications. In *Sonar*, if you choose the low-latency WDM/KS driver you lose the ability to use Direct Sound or MME drivers at the same time, which also precludes using Microsoft's GS wavetable software synth as a MIDI playback device. Conversely, if you're already running any audio application that requires MME or Direct Sound drivers, *Sonar* won't be able to run its WDM/KS driver option.

ASIO Confusion

Other driver formats, such as ASIO and GSIF, also use APIs and talk directly to the audio hardware, which is why they can also provide low latency. However, one of the biggest confusions for *Cubase* users is that, while this software only deals internally with its own ASIO protocol, Steinberg provide it with an 'ASIO Multimedia' driver and 'ASIO Direct X Full Duplex' driver that add MME and Direct Sound support respectively for interfaces that don't have true ASIO drivers. However, they remain Direct Sound and MME drivers underneath, which accounts for their relatively high latency. So, the best option for *Cubase* users is always a 'true' ASIO driver.

If your application offers both ASIO and WDM/KS driver options, the final choice is best made after practical tests, as relative performance may depend on the quality of each driver and may vary between manufacturers and models. I've found similar performance from both driver options with most review interfaces, although occasionally one driver format will manage lower latency than the other, or have a lower processor overhead. For instance, some Echo interface users running *Sonar* find that WDM/KS works better than ASIO.

The final caveat is that, since ASIO is now such a popular format, some manufacturers only release WDM drivers with limited I/O support. Echo's Audiofire WDM drivers, for instance, are limited to stereo in/out, while Emu's WDM drivers have recently gained multi-channel outputs but are still restricted to stereo inputs. In this situation, ASIO drivers are the only practical solution for *Sonar* users. SOS

If 2003 was Apple's 'year of the notebook', 2006 is set to be the 'year of Intel', with the prospect that every Mac computer in the company's range will use an Intel processor by the end of it.

Mark Wherry

At the January Macworld show in San Francisco, Apple CEO Steve Jobs' first keynote of the year contained the usual blend of somewhat expected and somewhat unexpected announcements for the Mac community. Among the expected product introductions were the now annual updates to the *iLife* and *iWork* bundles. But while there was much speculation before the show about the introduction of an Intel-based Mac Mini featuring Apple's new *Front Row* software for consumer media, this turned out to be just a rumour.

Get An *iLife*

iLife 06 features new versions of all the core applications, along with the addition of a new application called *iWeb*, which makes it easy to create web pages to share content created with *iLife*, such as movies, photos and music, as well as creating blogs. *iPhoto 6* now supports up to 250,000 photos (up from 25,000) and seems to be much faster at handling large numbers of photos; there's also a neat way of publishing photo albums online (a technique Apple has termed Photocasting), and a full-screen mode for viewing and manipulating photos.

iMovie HD 6 now allows you to have multiple projects open simultaneously, and introduces the concept of themes, allowing you to use ready-made, Apple-designed templates to create a slick-looking video by dragging your custom video clips into a pre-defined format. There are new real-time video effects, thanks to Tiger's Core Video technology, and Apple have also improved audio support, with new effects, including a noise-reduction processor, and the ability to access *Garage Band* songs. *iDVD 6* includes a feature called Magic iDVD, making it even easier to create a DVD-Video from scratch. It also supports widescreen format and offers a more detailed map view of a DVD's structure, similar to *DVD Studio Pro*.

Of most interest to music and audio enthusiasts will be *Garage Band 3*, which offers a new feature set aimed at making it easy to produce Podcasts. A Podcast track enables you to create a slide-show based on photos you have in *iPhoto*, and *Garage Band* makes it easy to automatically publish a Podcast online via *iWeb* integration. *Garage Band 3* also adds the possibility of a movie track, enabling users to easily produce a score for *iMovie* projects. We'll cover these features in more detail next month, but on first inspection it seems that Apple have focused on turning *Garage Band* into a general-purpose multimedia audio tool, rather than adding any new features for composing or producing music. This is perhaps a shame, compared to previous releases, but on the other hand will almost certainly make *Garage Band* appeal to a broader audience than before.

A Leap Year For Mac Users

Despite the lack of an Intel-based Mac Mini, this wasn't a Macworld devoid of interesting new Mac hardware. After an update about Apple's relationship with Intel, which featured Intel CEO Paul Otellini appearing on stage in a cloud of smoke, dressed in one of Intel's engineer 'bunny' suits, to present a silicon wafer to Jobs, along with plenty of corporate back-patting, the Apple CEO did indeed announce the 'first Mac with Intel processor': a new iMac.

The new iMac is basically the same as the old iMac, except that it now features an Intel Core Duo processor instead of a G5 chip. Intel's Core processor family (previously known to the computer world by its Yonah code-name) was officially launched at the CES (Consumer Electronics Show) in Las Vegas a few days before Macworld, and is basically an evolution of the Pentium-M processor, a high-performance, low-power processor that has become extremely popular in laptop computers.

There are two main types of Core processor: Core Solo and Core Duo. The Core name, as you might be able to guess, is a reference to processor cores, meaning that Core Solo is a single-core processor, while Core Duo is dual-core. It's therefore pretty significant that the new iMac uses a Core Duo processor. Among the Core processor features of interest to those using music and audio applications will be support for SSE3 instructions (the Pentium-M previously supported only SSE2). When utilised by programmers, these can offer a tremendous performance boost for DSP. Long-time Mac users can think of SSE as the equivalent to the Altivec instructions (or Velocity Engine) found in the G4 and G5 processors.

Other improvements to the Intel iMac include a fast 667MHz system buss and ATI Radeon X1600 graphics with 128MB video memory. Otherwise, the pricing and other features of the iMac (such as the built-in iSight camera and *Front Row*) remain the same, and the new iMac is also available in both 17- and 20-inch models, which feature a 1.83 and 2GHz ▶

With new Podcasting features, *Garage Band* is beginning to look more like a general-purpose multimedia tool than an app specifically for musicians.

Core Duo processor respectively. Just as I was finishing this month's column, a brand new 17-inch iMac arrived on my desk, so we'll be delving into that in more detail next month.

Before the keynote, there were also rumours that a new iBook might be discussed, and while they also proved to be incorrect, Jobs pleased the crowd by adding the now-customary 'one more thing' to the keynote, commenting that "there's been this pesky little problem of the Powerbooks." He acknowledged the lack of a G5 Powerbook and rectified the omission with the introduction of the Macbook Pro, a new 15-inch laptop for professional users. In some ways it's a shame to see a brand like Powerbook disappear; Jobs noted that "it's a new name because we're kind of done with Power".

The Macbook Pro features a new 15.4-inch screen, supporting a resolution of 1400 x 900, that's apparently as bright as Apple's Cinema Displays, and offers a built-in iSight camera and a remote control with infra-red support, for using the bundled *Front Row* media software. It shares many specifications with the iMac, such as the 667MHz system buss and the ATI X1600 graphics, and is also available in two models. The first model costs £1429 and features a 1.67GHz Core Duo processor, 512MB RAM and an 80GB 5400RPM drive, while the high-end model costs £1779 and offers a 1.83GHz Core Duo, 1GB RAM, a 100GB 5400RPM drive and 256MB memory. There are, however, a few curious omissions, such as the lack of a Firewire 800 port (only one 400 port is offered). In addition, Apple have yet to reveal battery life.

A Soft Option?

Aside from the physical computers, the most important aspect of such a major architectural transition is the availability of supporting applications and drivers. *iLife 06* is already shipping as a Universal Binary, along with *iWork 06*, and applications such as Mail and iChat that ship with OS X have also been ported successfully. At the Macworld keynote, Jobs mentioned that Apple's own 'Pro' applications (*Logic Pro*, *Aperture* and *Final Cut Studio*) will be available as Universal Binaries in March, and that a crossgrade to new versions will be available to existing users for $49 per title. This is perhaps a bit steep if you use all three, especially given that Emagic offered a freely downloadable OS X version of *Logic* to existing OS 9 users during the last major Apple transition.

At the recent NAMM show (just over a week after Macworld), Apple were showing the new *Logic Pro* 7.2 version — the first to be available as a Universal Binary — and announced that the upgrade would ship ahead of schedule in February. I was able to try this out briefly on a Macbook Pro at Apple's NAMM booth, and it seemed to be running pretty well: *Logic*'s mastermind, Dr. Gerhard Lengeling, mentioned that they had spent a great deal of time on the SSE optimisations.

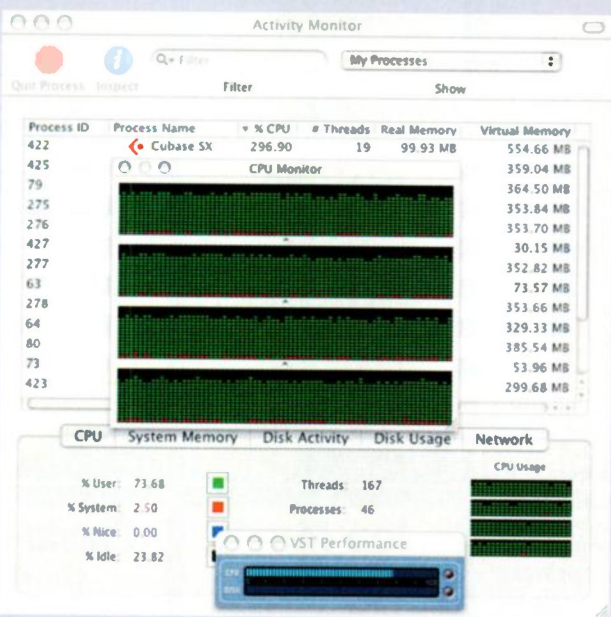
The Macbook Pro itself looks similar to the existing 15-inch Powerbook, but you really notice the differences when the two are side by side. The new Macbook is sleeker, and the brighter, slightly larger (and more widescreen) display is really nice. Given the lacklustre performance of the G4 Powerbook at the moment, the Macbook will be welcomed by many mobile musicians, and it will be really interesting to see what Apple have planned for the Macbook revision of the current 12- and 17-inch Powerbook models later in the year. SOS

Four Heads Aren't Necessarily Better Than Two

In last December's Apple Notes we discussed the introduction of the dual-processor, dual-core Power Mac Quad system: the fastest Power Mac on the planet — and possibly, with the Intel transition and the eschewing of the word 'Power' in newer products, the last Power Mac ever to be released. I was able to play with one this month, and the first thing I noticed is that the power cable is now much thicker than before and uses a modified connector meaning, that you can't just plug in any old kettle lead (IEC lead, for our American readers), as before.

The other rear-end observations of note were that there's now an extra USB port (making three in total) and an additional Gigabit Ethernet port, and the Airport and Bluetooth antennae ports have disappeared. Given that there is now a new vertical blank plastic panel on the back of the computer, I'm guessing that the antennae for these wireless technologies has been incorporated into the beast itself.

In terms of performance, our usual Mac testing involves loading up *Logic Pro* and seeing how many instances of various plug-ins can be used simultaneously. Having already used music and audio software on dual-processor, dual-core Opteron systems, I feel it's not unreasonable to expect somewhere close to double the audio performance of a two-core system, with properly written software. However, when I tried the usual 'more *Platinumverbs* than you can shake a stick at' test, I only managed 160 instances — the same number as the dual-2.7GHz Power Mac tested in Apple Notes of July last year. Further inspection in the Activity Monitor utility confirmed an obvious conclusion: while the dual-2.7GHz machine ran 160 instances with 98 percent user usage, the Quad ran with 47 percent usage, meaning that *Logic* wasn't taking advantage of the two additional processor cores.



Activity Monitor reporting on how *Cubase SX* 3.1 is utilising system performances when running 128 instances of *Reverb A*. Notice that four processor graphs show the utilisation of the four processor cores in the Power Mac Quad.

At the NAMM show, I asked an Apple representative about this, and he confirmed that *Logic Pro* hadn't yet been optimised for systems with more than two processor cores, since greater priority was given to the SSE optimisations to make the application run rather well on Intel chips. What this actually means is that while you should get a little extra power for the machine (you wouldn't have known the Quad was running 160 plug-ins, given the responsiveness of the user interface compared to the dual-2.7), *Logic*'s actual audio engine still only runs on two cores, so you won't get four cores worth of DSP power. Hopefully this will change in future versions of *Logic*, after the release of 7.2.

Although *Logic* doesn't yet take full advantage of the Power Mac Quad, Steinberg's *Cubase SX* 3.1 does; so I decided to see how many *Reverb A* instances I could run instead, since *SOS* has published many articles with *Reverb A*-related performance marks in the past. In this test I got to 128 instances, and this time, as you'll see from the screen shot, Activity Monitor reported approximately 75 percent CPU usage. This would suggest that *Cubase* was using three of the four cores, but you can see from the screenshot that the load is actually balanced across all cores.

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If you've ever considered taking out a subscription to *Sound On Sound*, now is definitely the time to do so, as anyone purchasing a subscription before the end of March could win a massive collection of microphones and recording channels courtesy of PMI Audio. Existing subscribers are also eligible — simply tick the 'I am already a subscriber' box on the entry form.

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If you would like to win this fine haul of studio kit, simply fill out the form opposite and send it back to us. Please ensure your entry and (if you're subscribing for the first time) your subscription details reach us by the closing date: 31st March, 2006. SOS

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VOCAL BOOTH, large, holds up to four people, well built, £750. Computer booth with extractor fan, £75. Floor standing rack unit, £29. Acoustic tiles, nine approx, £39. ☎ David 07950 645725 (Wembley, Middlesex).

wanted

DIGITECH STUDIO QUAD 4. Ibanez RG350 or similar (with Edge Pro 2), Ibanez GRG170 or Tanglewood 'Che', cosmetic condition unimportant. ☎ Martin 0161 427 9230 (Stockport, Cheshire).

EMU VIRTUOSO or Roland M-OC1 orchestral sound modlules wanted, must be in perfect working order and not over priced. ☎ 07908 030788 Selby, Yorkshire).

Brave new world?

John Savage

Looking back, my old studio was a mess. I was forever tripping over pieces of kit, and if I forgot to switch on the air-conditioning, it became damnably hot in there. There were racks of equipment, numerous keyboards, a large analogue mixer, and dedicated hardware boxes such as effects and routers, as well as the inevitable miles of trailing cables. My creativity was at a standstill, and the studio itself was an impediment to the frictionless path between inspiration and adoration. I had to do something about it. So I did. I sold the lot.

That might sound extreme, but it wasn't. I had just bought a new 3GHz dual-Xeon PC, and I had started to experiment with software synths and effects. I realised that to achieve fame, fortune, and unlimited offers of guilt-free sex, all I needed was a couple of processors, a bucketful of RAM, a controller keyboard, and software. Smart!

The problems began after I sold my analogue synths. I purchased all the latest software emulations, and although each worked in isolation, they didn't like co-existing with one another. I was so surprised by this that I did something that no male likes to admit to. I read the manuals. I discovered that *Softsynth #1* required all the power of my mega-computer, leaving nothing for *Softsynth #2*.

I called technical support. 'Why', I asked, 'does your synthesizer require enough computing power to run most of the world's financial institutions?' 'Ah...', they replied, 'the filter emulation is the thing. And the oscillator emulations. And the modulation busses. Especially when used polyphonically.'

Now, I'm not a programmer, but I remember when you could cram a decent flight simulator into 32KB of RAM and run it on an 8-bit processor. Nowadays, the same game requires the processing power of the Boeing 747 that it purports to emulate. With this in mind, I suggested to the disembodied voice that *Softsynth #1* was bloatware.

I had expected a vehement denial, but I received an excuse instead. I am, apparently, a consumer. What's more, I'm very demanding. I want everything now, and I want to pay next-to-nothing for it. To achieve this, the manufacturer has to produce products as quickly and cheaply as possible. This leaves no time for software optimisation. So if *Softsynth #1* needs a gigbyte of RAM in which to swim, and teraflops of processing power for sustenance... well, no matter. Moore's law ensures that, sometime in the next 18 months, the necessary PC will be available for under £1000 at the local computer emporium.

Unfortunately, I couldn't afford to wait a year and a half. In fact, I couldn't wait a week and a half. So I bought a second computer. The results were magic. Freed of the shackles of inadequate processing power and the conflicts of incompatible device drivers (whatever they might be), *Softsynth #2* leapt into life in a way that had previously seemed improbable.

A few weeks later, a similar thing happened. I was running a suite of the latest plug-in effects when the studio was filled with a burst of noise that would have woken the dead, had there been any lying around at the time. On this occasion, I was told, the problem was something to do with a lack of processing power and 'running out of real time'. Fortunately, I had a powerful laptop and a 24-bit USB/audio interface to hand, so I installed the stand-alone version of the effects software on this. Again, the problems disappeared. OK, so this wasn't as handy as having everything in a single, integrated environment, but using the laptop eliminated software glitches and, with the addition of a small digital desk, I found that this configuration allowed me to route everything very flexibly.

Things continued to progress in this fashion, with additional computers installed to run power-hungry soft synths and samplers, but a turning came when I was trying to install the latest *Deutscherübersynt* onto one of the PCs in the studio. Its dedicated controller needed a spare USB socket, of which there were now dozens. Unfortunately, the synth wouldn't talk to any of them if other devices were plugged into the same PC, so I nipped out to buy a dedicated USB hub for it.

Returning to the studio, I opened the door, walked in... and tripped over a piece of kit. Picking myself up, I realised that I had left without switching on the air-conditioning, and that it was damnably hot in there. I looked at the racks of PCs, numerous monitors and keyboards, a large digital mixer, dedicated hardware units such as keyboards and MIDI controllers, as well as the inevitable miles of trailing cables. 'Oh Brave New World that has such equipment in it', I proclaimed. Then I cried. SOS

About The Author

John runs BNW Studios, where he lives with his lovely wife, Lenina.

If you would like to air your views in this column, please send your submissions to soundingoff@soundonsound.com or to the postal address listed in the front of the magazine.

Sound On Sound's FREE live sound supplement

Sound On Sound LIVE

Line arrays

Could arena PA technology transform your band's sound?

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Sound On Sound Live supplement

Issue number 7: March 2006

SOUND ON SOUND

Sennheiser Freeport radio mic ▪ **Chris Trimby** award-winning monitor engineer ▪ **LD Systems PM 12-2** live sound mixer ▪ **Yamaha Stagepas 300** portable PA **Flightcasing** and equipment protection ▪ **Etymotic ER4S** in-ear headphones ▪ **Fohhn Xperience II** powered PA ▪ **AMT SP25B** acoustic bass miking system **Product news** the latest gear ▪ **Win** Carlsbro Liberty wireless speaker system

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H PRO Harman Pro UK

It may be hard to believe right now, especially with British weather being what it is, but we're quickly heading for that time of year when the live sound engineering types amongst you will start to get the chance to work at open-air gigs and summer festivals.

A festival can seem like a daunting prospect, especially if you don't have loads of experience; after all, at some open-air events it's routine to deal with up to a dozen bands in a day, and to have an impossibly short space of time in which to change over between acts. There's usually no time to soundcheck, so you have to be able to achieve a decent balance within a few bars of a band starting, and you always need to be on the alert for people mixing up the mics on stage. To try to make sense of this chaos, I usually reserve certain mics for the drums and backline and use coloured tags to identify these, so that I can see which is which from my mixing station. For vocal and instrument mics that might be moved around more, I use the simple expedient of coloured mic cables. Providing you have matching coloured stickers or marker-pen blobs on your mixer scribble-strip, you should be able to keep track of the various mics. If you have time, put another strip of tape below the first and write on it which voices or instruments relate to the various mics and DI feeds, so that you can react to changes more quickly. When the band has finished, stick on another strip, rearrange the mics as needed and note down their new functions.

With this kind of gig, I find that it is close to impossible to set up a good monitor mix for each band, unless you have a separate monitor console on stage, so a safer bet may be simply to feed the main FOH mix to the monitors but with some low end rolled off. That way, you won't pump excessive bass levels back onto the stage. While having your own custom monitor mix is certainly nice, this approach is workable, and it lets the band hear themselves as other people hear them.

If you're asked to tackle a multi-band event this summer, don't shy away from it, but at the same time realise that it is going to be a tough day, and seriously consider taking an assistant to help you redeploy the mics (and cover for you when you need the bathroom). The main requirement is an ability to concentrate exclusively on the performance in hand and to be confident about what each fader will adjust when you move it. Assuming that you have a good ear for music, you should be able to busk the rest!

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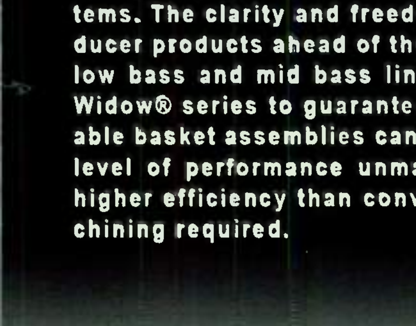
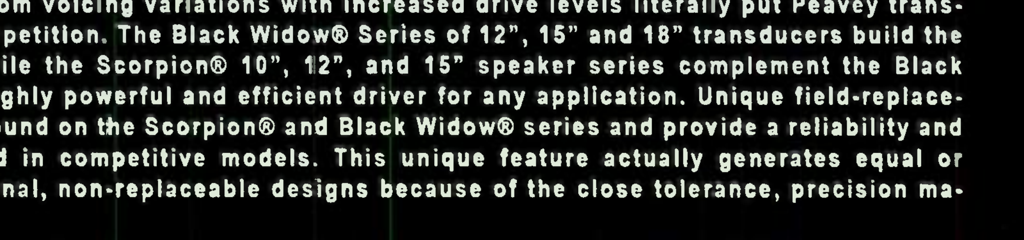
The primary function of any system or enclosure is its ability to respond to today's ever changing musical environment. Each Peavey UL™ enclosure, from the small satellite UL™10 to the low frequency UL™215S sub woofer enclosure, exhibits maximised reliability as the cornerstone of that foundation. Kevlar®, a light-weight, synthetic compound of phenomenal strength, is utilised in all Black Widow® series drivers as well as selected Scorpion® products. The resulting increased stiffness and superior mechanical damping ensures maximum energy transfer while maintaining ultra high power handling and low deformation of moving systems. The clarity and freedom from voicing variations with increased drive levels literally put Peavey transducer products ahead of the competition. The Black Widow® Series of 12", 15" and 18" transducers build the low bass and mid bass link, while the Scorpion® 10", 12", and 15" speaker series complement the Black Widow® series to guarantee a highly powerful and efficient driver for any application. Unique field-replaceable basket assemblies can be found on the Scorpion® and Black Widow® series and provide a reliability and level of performance unmatched in competitive models. This unique feature actually generates equal or higher efficiency than conventional, non-replaceable designs because of the close tolerance, precision machining required.





1) Remove three screws on back of magnet structure.

2) Lift magnet structure off basket frame.

3) Clean voice coil "gap".

4) Align screw holes, lower structure into place on new basket frame, insert screws and tighten.

Peavey Electronics Limited - Great Folds Road - Oakley Hay - Corby - Northants NN18 9ET- England - Phone +44 (0) 1536 461234 - Fax +44 (0) 1536 747222 www.peavey-eu.com

JTS debut UHF radio mic system

A new 1000-channel UHF radio mic system from JTS is about to hit the streets. This affordable true-diversity system is based around the US 1000D receiver and can be bought with a hand-held microphone transmitter, for vocals (£299), with a belt pack for instrumental use (also £299), or with a headset mic (£325).

The US 1000D receiver is a half-rack unit that operates on frequencies between 841 and 865MHz, and will therefore happily work in the UK licence-free band of 863-865MHz. If you're a licensed user of radio equipment, choose instead the frequencies between 841 and 863MHz. Other features of the receiver include a useful 'lock-on' mode, which avoids accidental modification of key parameters, PLL (Phase Locked Loop) technology for tuning stability, and true-diversity operation, ensuring that only the strongest signal is used at any given time. SAW filtering provides interference resistance, noise-mute and tone-squelch circuitry is built in, and the 1000D has also been designed to work around computerised equipment without suffering interference problems. Watch this space for an *SOS Live* review.

Proel (International)
+ 44(0)20 8761 9911
www.proelint.co.uk

New vocal mic from Beyerdynamic

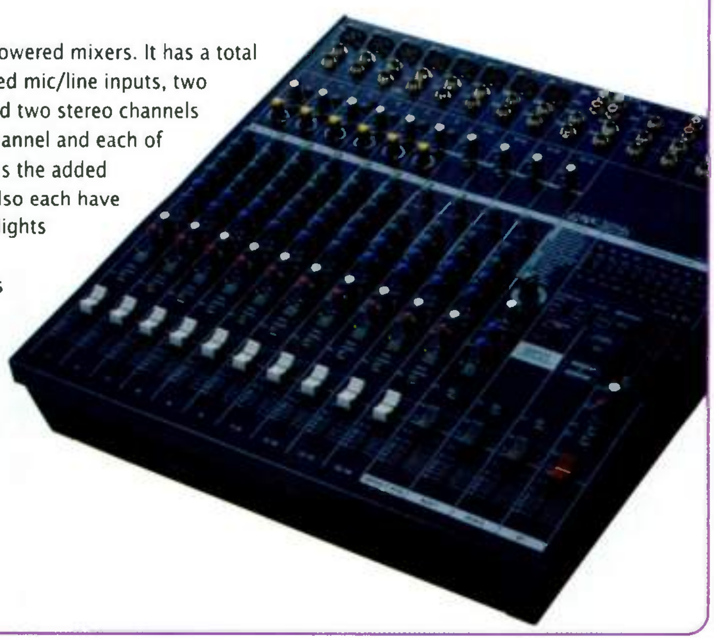
Beyerdynamic's Opus 89 is a new dynamic mic designed especially for vocalists. It has a hypercardioid pickup pattern and is designed to offer a wide frequency response and excellent isolation from handling noise, while being robust enough for touring. The Opus 89 costs £198 including VAT.

Beyerdynamic UK
+44 (01444)258 258
www.beyerdynamic.co.uk

Yamaha unveil new mixer

The EMX5014C is the latest addition to Yamaha's EMX range of powered mixers. It has a total of 14 input channels — six mono channels with phantom-powered mic/line inputs, two further mono/stereo channels (mono mic or stereo line input) and two stereo channels with unbalanced line inputs. There's a three-band EQ on every channel and each of the six mono channels has a sweepable mid-frequency band, plus the added bonus of an easy-to-use 'one knob' compressor. Channels 1-10 also each have a Feedback Channel Locating System (FCL) indicator LED, which lights when that channel goes into feedback — a handy feature for experienced engineers and novices alike! The mixer also features two aux channels, a global nine-band stereo graphic EQ, 16 SPX digital effects and two 500W amps. Weighing in at just 10.5kg, the 5014C is designed to be light enough to easily transport around (a useful carrying handle is built into the front of the mixer), but can also be installed using attachable rack ears. This looks like a very well-thought-out new product with all the right features to appeal to its target market of amateur and semi-pro engineers and bands, especially with a projected price of £599.

Yamaha-Kemble Music +44 (0)1908 366700
www.yamahaproaudio.com



Noise-free connectors from Neutrik

The XLR and quarter-inch jack standards have been with us for quite some time now and you might think that there was little scope for improvement, but somehow Neutrik keep finding meaningful ways of reinventing the wheel (or the jack plug anyway). Take, for example, the new Neutrik Silent Plug. This rugged quarter-inch jack connector features an integrated 'silent switch' which promises to put an end to the loud pops that occur when thoughtless guitarists plug or unplug their instruments on stage. Meanwhile, the new EMCXLR male and female XLR jacks feature integrated filters and a 360-degree shielded contact on the female connector, which protect the signal from RF interference, ground loops and low-frequency noise.

Neutrik UK +44 (0)1983 811441
www.neutrik.co.uk

Mackie introduce new live sound gear

Mackie used the 2006 Winter NAMM show in the USA to launch a whole range of live sound products. First of all, there are two new mid-format, four-buss live sound desks in the form of the 24-channel Onyx 24.4 and the 32-channel Onyx 32.4 consoles. The new models feature premium Onyx mic preamps, four-band Perkins EQs, a built-in compressor/limiter, and 100mm faders, direct outs and four-segment metering on every channel.

No less impressive are the Quad Series digital processors. The Quad EQ provides four channels of 30-band digital graphic EQ, while the Quad Comp/Gate offers four channels of 24-bit digital compression, limiting, gating and expansion. Both units feature balanced XLR and TRS connectors and 99 snapshot memories. In both units, channels A/B and C/D can be linked for dual-stereo operation.

The new high-output M-Series stereo power amps employ Class-H circuitry and a number of proprietary Mackie technologies, and are designed to accurately reproduce fast transients like kick and snare drums with minimum distortion. The M2000, M3000 and M4000 pump out an impressive 2000, 3000 and 4000W respectively.

Finally, joining Mackie's affordable Tapco range, the Thump TH15A is a 200W bi-amped PA speaker featuring a 15-inch woofer and one-inch compression driver, an active crossover and a three-band EQ with sweepable mid-range.

All the new products are set to go on sale in the Spring.

Mackie UK
+44 (0)1268 571212
www.mackie.com

Affordable wireless systems from AKG

AKG have launched three new wireless systems. Borrowing technology used in the WMS 400 and WMS 4000 systems, the WMS 40 PRO range consists of three packages. The WMS 40 PRO Flexx system (pictured) incorporates a true-diversity receiver with nine channels and three selectable frequencies per channel. The WMS 40 PRO Single and Dual systems operate on fixed frequencies, with the Dual providing two independent channels so that two performers can use one receiver. A range of transmitters is available to accommodate mics, guitars and other instruments, and AKG claim they can operate for up to 30 hours on a single 'AA' battery. Prices for transmitter and receiver sets start from £259 for the Flexx system, £199 for the Single system and £329 for the Dual system.

Harman Pro UK +44 (0)1707 668222
www.harmanprouk.com www.akg-acoustics.com

Personal monitors from TC-Helicon

Best known for their sophisticated vocal processing units, TC-Helicon have turned their hand to designing a range of on-stage near-field monitors for vocalists. The passive VSM200P and active VSM200 and VSM300 monitors feature a cast-aluminium enclosure and a driver specially designed for reproducing the human voice. The two active models are equipped with a 150W amp, while the top-of-the-range VSM300 (pictured right) features additional instrument and aux inputs, front-panel level and EQ controls and TC's Voice Shape circuitry, designed to add "studio microphone tone to a typical live microphone", apparently. The VSM range is due to ship in February.

TC Electronic UK +44 (0)800 917 8926
www.tc-helicon.com

Sennheiser Freeport

It's not difficult to find a cheap wireless mic system these days, but not all have a prestigious name tag — unlike this Sennheiser true-diversity setup, their most cost-effective to date.

A schoolteacher friend of mine recently asked me for advice about what sound equipment to buy for his school hall. As Gavin's requirements included a hand-held radio microphone and finances were limited, I suggested that he wait a week or so until I'd had the chance to look at the latest affordable UHF radio mic system from Sennheiser: the Freeport.

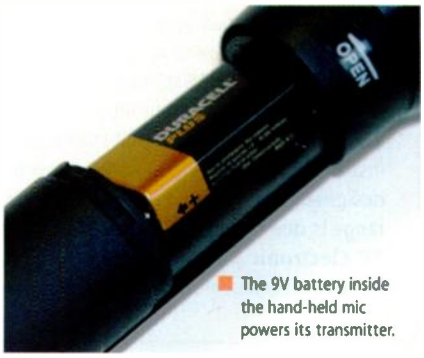
The Freeport is Sennheiser's budget UHF full-diversity radio microphone system, providing four user-selectable channels within the range 863-865MHz (licence-free in the UK) or 742.5-744.5MHz (for use only in the US and some European countries). All models supplied in the UK should operate in the licence-exempt band, which, I believe, is also unaffected by the introduction of the new digital TV channels and consequent spectrum re-allocation in the future (see our feature on wireless system licensing in the September 2005 issue of *SOS Live*). The four frequencies will operate simultaneously, free of intermodulation, so you can use up to four Freeport systems together (you will need a separate receiver for each microphone). A point to note is that the frequencies used by the Freeport are not exactly the same as those used in other radio systems from Sennheiser, such as the Evolution range, and you'll need to check frequency compatibility if combining different systems.

What's in the box?

The Freeport is available in hand-held, belt-pack and lapel versions, and the hand-held vocal set is the one reviewed here. It arrived well packed in a fitted cardboard carton, which is not intended to be a transit case for regular use, although it looks as if it would see you through a few gigs, or serve as a storage case if the mic were going to be kept and used at the same place. The set comprises the hand-held microphone, finished with a smooth grey non-slip surface, receiver unit, power supply and mic-stand clip.

The SK3 hand-held vocal mic is a dynamic cardioid type with a fixed capsule. Power for its 10mW integral transmitter is provided by a 9V battery (what us oldies call a PP3!), which gives around 10 hours of operation.

To gain access to the battery compartment and channel-selector switch, part of the microphone body which acts as a locking ring must be twisted about a quarter turn; the rest of the body then slides away from the capsule (but not completely off, so it shouldn't get damaged or mislaid) to allow access. One of four available channels can be selected by means of a small rotary switch, which really needs a thin screwdriver to turn, as it's recessed out of harm's way. The channel numbers one to four are clearly visible around the selector. On the bottom of the mic body is a slightly recessed on/off toggle switch, with a small red LED to indicate when the microphone transmitter is powered up. The



The 9V battery inside the hand-held mic powers its transmitter.

At a glance

Sennheiser Freeport

Pros
- Quality build.
- Neat design and appearance.
- Good performance — stable and quiet.
- Good range.

Cons
- None at this price.

Summary
The Freeport is an excellent little system which performs well, looks good and won't melt your credit card.

Information
E The three Freeport systems each cost £229 including VAT.
T Sennheiser +44 (0)1494 551551.
W www.sennheiser.co.uk

The Squelch control, seen on the rear panel of the EM1 receiver, is very much like the threshold control offered by an audio gate. It can be adjusted to allow through the signal from your own transmitter but to mute weaker signals that are lower than the level that has been set.

The simple front panel of the EM1 receiver features a power LED; a four-segment Radio Frequency LED meter to show signal strength; a recessed channel selector; a pair of Diversity LEDs to show whether channel A or channel B is in use; and an 'AF Peak' indicator to show audio peaking.

short antenna also protrudes from the base-plate, and, although flexible, is quite chunky and looks as though it would withstand the amount of punishment it would tend to get during normal use.

At the other end, the capsule is protected by a strong wire-mesh basket, which contains a removable and washable foam pop/moisture shield. Although the basket assembly unscrews for simple maintenance, the dynamic capsule is not removable and is therefore not interchangeable with another capsule. The capsule mounting is itself flexible to reduce physical shocks which would otherwise be transmitted through the body, and there's an additional foam screen on top of the head unit, providing extra protection. Overall, the microphone feels light and well-balanced, and seems very well built. Even the on/off switch has a quality action.

On the receiving end

The EM1 receiver is housed in a small metal case of about a half-rack size. However, as far as I know there's no rackmount kit available for this model. On the front panel are two swivel antennae (happily non-telescopic, therefore it's much easier not to poke your eye out with them), a rotary channel selector (of the same type as on the microphone) and LED indicators for power, channel activity, audio peaking and RF (radio frequency) level. On the rear panel are balanced XLR and unbalanced jack outputs, Gain and Squelch controls and the DC power connector.

The receiver is a full-diversity design, which means that it incorporates two independent receiver paths, one connected to each antenna; the receiver automatically switches to the one providing the strongest signal, thereby reducing the risk of system drop-outs and loss of signal continuity. It's a very neat little package, and again has that quality feel about it so typical of products from this manufacturer.

Also revealed when the mic body is slid apart is the recessed channel selector switch, used to choose one of the system's four available operating frequencies.

Power up

Having inserted a battery and checked that mic and receiver were both set to the same channel, I switched both parts on and did a few "one two"s. At first, I missed having an audio level meter on the receiver — it's very reassuring to see the signal going nicely up and down before you turn up the PA — but of course you can set this on your mixer anyway.

The sound quality offered by the Freeport is absolutely fine for a range of uses, from stage vocals to conference speech and PA announcements. The 3dB frequency response claimed in the handbook is 80Hz to 16kHz, and the Freeport's performance certainly sounds in line with this specification. Speech comes over very clearly, with a nice crisp top end, good solid punch in the middle (especially on louder signals) and enough smoothness lower down. There's no unwanted system noise either, and in a quick comparison with some of my favourite general-purpose vocal mics it more than held its own. Although I didn't get the chance to use the Freeport system in a full-blown live situation, I did manage to slip it into a full band rehearsal, where it made a very good showing.

The RF range of this system is impressive: reception is full-strength (four LEDss) up to about 30 metres 'line-of-sight', and I still had three green lights at over 100 metres of open country. Everyone who handled the mic liked the feel and balance of it. I also appreciate the Freeport's understated appearance, which is similar to the more upmarket Sennheiser models, and indeed it looks and sounds as though it should cost a lot more than it actually does.

Value judgement

Obviously, you can't have absolutely everything in this price range, and some features have been left out: there are only four preset channels to choose from, and there's no audio level meter or fancy pilot-tone technology. What you do get is a very nicely built, neat-looking and good-sounding system for relatively little money, which will do for me any day. When considering buying 'budget' products, I've always been keen on those from leading manufacturers in the particular field, because they have their reputation to maintain, and I know from personal experience that Sennheiser UK are able and willing to provide good technical support.

So, having had a look at the Freeport system, I like it a lot — and yes, I think it could be the system you want, Gavin. I'm buying this one, though. SOS

Test info

Reviewed by Mike Crofts
Photographs by Mark Ewing & Mike Crofts

Yamaha Stagepas 300

Portable powered PA system

An affordable complete vocal PA you can carry from the car to the stage in one trip!

Yamaha's Stagepas 300 is a surprisingly compact yet reasonably powerful stereo sound system comprising a pair of moulded, passive two-way speakers and a small powered mixer that stows into a recess in the rear of one of the speaker cabinets for transport or storage. The other speaker has a similar recess with a removable cover for stowing cables or mics. According to the specifications, the mixer's dual power amplifiers can produce 150W per channel into 6Ω with 10 percent distortion or 100W per channel with one percent distortion. Given that the system's mixer-amp is little larger than a decent box of chocolates, that is quite remarkable. Sadly, the accompanying documentation doesn't say what kind of amplifier technology is being used, but to make the amp this small and light I suspect that they're using some kind of Class-D circuit or, at the very least, a switch-mode power supply.

Speakers corner

The speaker cabinets, which measure approximately 18 x 11 x 10 inches, are moulded from a tough, resilient plastic with no obvious resonances but, unusually, they don't have a pole-mount fitting built in. You have to buy external stand adaptors as accessories if you wish to stand-mount the speakers, which seems rather odd to me, as how else would you use them? Both speakers are identical, so either one can hold the mixer amp while the other can be

Test info

Reviewed by Paul White
Photographs by Mark Ewing

used to store the supplied speaker and power cables or a couple of mics. To secure the mixer into the rear of the speaker, there are two twist locks that can be opened with any suitable coin, and the mixer is stowed 'controls out', so that it can be used while still fitted to the speaker if necessary (although the front panel would then be 'sideways on' to the user, if the speakers were being used upright). I'd prefer catches that didn't require tools to open, but even musicians tend to have the odd coin about their persons and the catches actually work very well. All the connections, including the IEC mains inlet, are on the top of the mixer.

The cabinets, which are quoted as having a 55Hz to 20kHz frequency response, each contain an eight-inch mid/woofer speaker and a one-inch HF compression driver feeding a moulded horn flare. Passive crossovers inside the speakers operate at 4kHz, and a strong, perforated-steel grille protects the cone driver. (This may be a budget system but it is built to the usual Yamaha standards of engineering.) The amplifier's response extends down to 20Hz and up to 20kHz (-3dB/+1dB), measured at an output of one Watt and at a distance of one metre. Maximum SPL is a respectable 112dB which, considering that the whole system weighs only 18kg, is pretty impressive. A preset limiter, with corresponding red LED on the mixer's front panel, is included in the amplifer section to protect against hard clipping.

Mixing it

The little mixer is fan cooled (the fan is quiet enough for most live sound applications) and offers eight inputs in all. Four of these are mono mic/line channels featuring both quarter-inch jack and balanced XLR mic inputs. There's no phantom power on the latter, which is a shame. The remaining four inputs are presented as two stereo line channels offering quarter-inch jack and/or phono inputs. Each channel has low (100Hz) and high (10kHz) EQ controls (with a +/-15dB range), as well as a rotary level control, while the four mono mic/line inputs also have illuminated reverb-select switches in place of the more familiar post-fade effects-send knob. A master reverb control adjusts the overall amount of reverb added, but as there is no variable send on the channel strips, the reverb amount is the same for all channels on which the reverb switch is engaged. Though the reverb doesn't come close to studio quality, it has a useful 'reverb meets echo' character that is suitable for general singing or — dare I say it — Karaoke!

In the output section, which is on the left of the mixer, rather than the right, where you'd normally expect it to be, are a five-LED level meter, a master level pot, and a further Monitor level control feeding a stereo pair of jack outputs that are ideal for connecting a pair of powered stage monitors. The two phono outputs in the same area are intended for use as recording outputs, when connected to an external recorder. There's also a Speech/Music switch that seems to create a slightly 'scooped' response in the music position, much as a conventional loudness button does, but arguably to more subtle effect.

Steel handles help to protect the front of the mixer when stowed, and although the main body of the mixer housing is made from resilient plastic, the front panel is metal and all the jacks are fixed with metal nuts, not plastic ones. As the mixer is stowed 'knobs out', you need to be reasonably careful how you pack it in your car, unless you have the optional gig bag, which I'd highly recommend.

Setting up

I tested the Stagepas with its speakers perched on tables, as the review system »

At a glance

Yamaha Stagepas 300

Pros
- Compact.
- Affordable.
- Tough construction.
- Good basic sound quality.

Cons
- Very limited feature set.
- No integral stand mounts.

Summary

The Stagepas 300 is good value for small-venue sound reinforcement where there is no requirement to play back low-frequency material at high sound levels. I feel it's best suited to solo acts and duos working the pub circuit, although it can double as a rehearsal PA and a stage monitor system for larger bands. It's also well suited to basic conference work.

Information

£ £399. Stand adaptors £32 per pair; optional gig bag £75. Prices include VAT.
T Yamaha-Kemble +44 (0)1908 366700.
W www.yamaha-music.co.uk

The mixer offers four mono mic/line channels which can each access the basic built-in reverb (but not in variable amounts), plus two stereo channels offering jack and/or phono inputs. High/low EQ is available to all channels. Other features include jack outputs for connecting powered monitoring and a pair of phono recording outputs.

Yamaha Stagepas 300

Portable PA

» didn't come with the stand adaptors (the omission of these as standard is perhaps the worst practical failing of the system). It's also worth mentioning that, unlike some other compact PA alternatives, you can't fasten the speakers face-to-face to protect them in transit (another good reason to invest in the optional gig bag). Plugging in the included mains and 5m speaker leads takes just moments, after which you're ready to go.

Although the speakers are quite small, they carry vocals very clearly, and even do reasonable justice to full-range material at lower levels. The coverage angle is a good compromise, and although the high end is understandably hotter 'on-axis', the overall sound is well balanced and resists feedback rather better than some small systems I've tried. There's enough vocal level to handle smaller pub gigs for bands or duos, and I rather liked the effect of the admittedly basic built-in reverb. Alternatively, the Stagepas 300 would make a great rehearsal PA for your front room or garage and could double as a compact stage-monitoring system. I used it for a fairly spirited band rehearsal and it delivered bags of vocal volume without undue feedback problems. It's not recommended to drive the system so hard that the limiter LED comes on, or sound quality will suffer somewhat, but the built-in Limiter is still a useful form of speaker protection.

The useful storage compartment is ideal for the mains lead and 5m speaker cables that are included with the system. The mixer slots into the storage space on whichever speaker isn't being used for cables.

Conclusion

The Stagepas 300 is strictly 'no frills', as its basic two-band EQ and switchable reverb attests, but it is solidly built and does its job very well. Many smaller acts will find four mic inputs perfectly adequate, while the mixer's stereo line inputs are ideal for connecting guitar preamps, keyboards or stereo backing tracks.

The mixer has no pre-fade sends, so you can't generate an independent monitor mix from it, and its mic inputs can't power capacitor mics or active DI boxes that need phantom power, but in the system's favour is its solid sound, its very compact format and its affordability. For stand mounting of the speakers you'd need to buy the optional adaptors, but other than that, this is perhaps the ideal system for a pub solo artist or duo, or for an acoustic act needing a bit of FOH reinforcement. SOS

Jargon explained

Class-D amplifier: One in which the output transistors are operated, effectively, as switches, being either fully off or fully on, with the audio signal modulating (controlling) the switching action. Class D is very efficient, allowing smaller power supplies to be used. The 'D' is sometimes mistakenly said to stand for 'digital' when in fact it was simply the next letter in the alphabetical series after Class A, Class B (and A/B) and Class C.

Compression driver: Specialised high-frequency loudspeaker with a small, domed diaphragm and usually attached to a directional horn. The narrow horn 'throat' constricts the volume of air into which the driver radiates, hence the 'compression' part of the name.

Crossover: A crossover is an electronic circuit designed to separate high- and low-frequency signals from each other so that each can be fed to speakers optimised for the role — ie. large and robust speakers for bass, small and light (and therefore fast) ones for high frequencies. If a crossover is placed between power amps and speakers (usually built into the speakers), it is said to be 'passive'. An 'active' crossover is used to divide signals before the power amps, so separate amps are then used for each band, further adding to efficiency. Crossovers can be two-way, just splitting highs and lows; three-way, adding a mid band; and occasionally four-way.

Phantom power: Standardised scheme of providing an invisible (hence 'phantom') power supply voltage to capacitor (condenser) microphones using the same cable as the balanced audio output.

SPL (Sound Pressure Level): Sound level calculated in decibels compared to a reference sound pressure, commonly 20 micropascal (20 uPa), defined as the threshold of human hearing (0dB SPL). The human ear is capable of hearing an enormous range of sound pressures — the ratio of the sound pressure from the minimum up to damaging levels from even short-term exposure is more than a million — necessitating a logarithmic scale, which also corresponds roughly to our psychological perception of loudness.

Switch-mode power supply: A high-efficiency power-supply design which allows the PSU to be smaller and lighter for a given capacity.

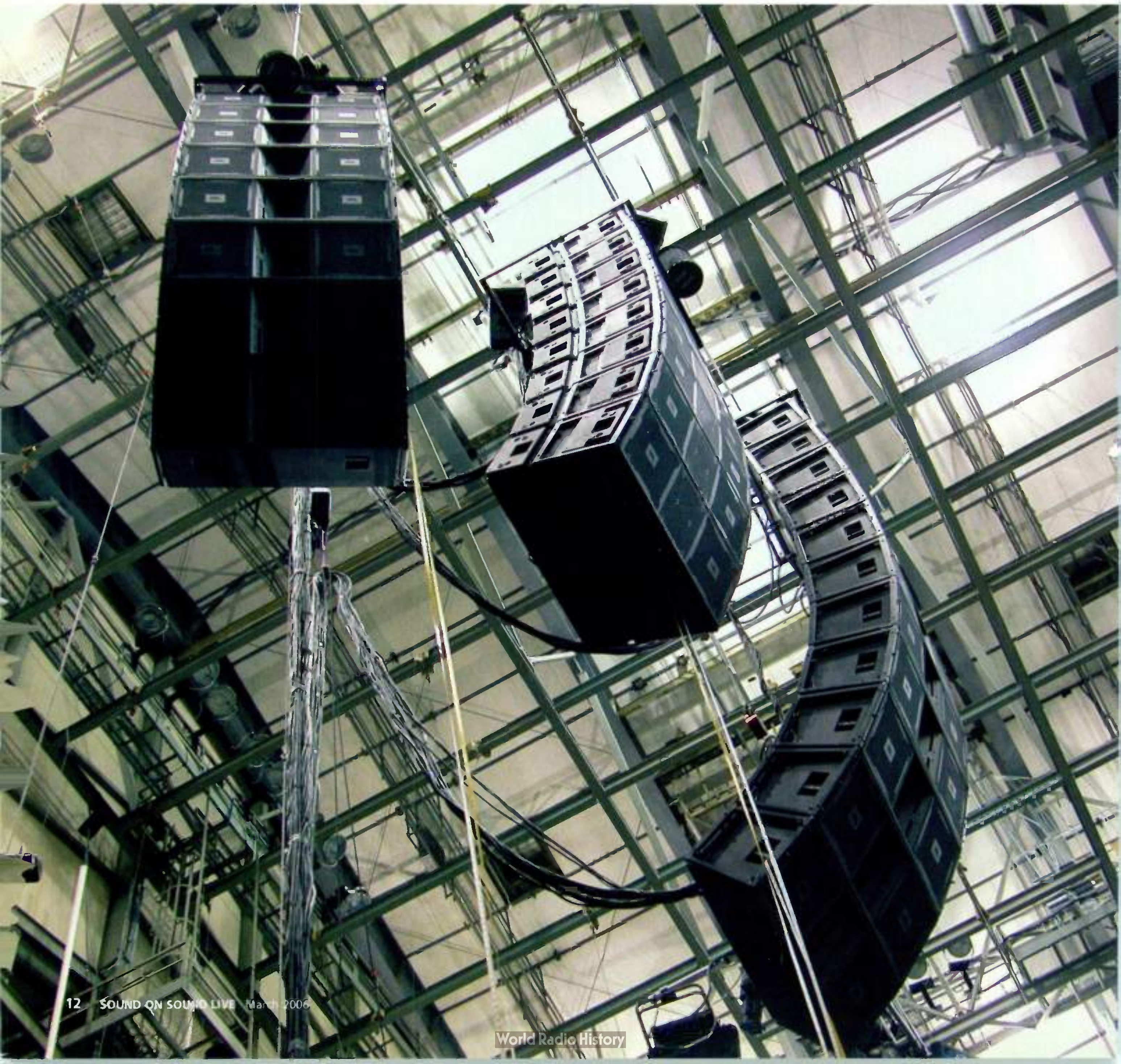
Line Arrays

The science and the magic

If you've been to a biggish gig or a festival in recent years, you've had the pleasure of hearing line arrays of loudspeakers in action. But why are line arrays the current 'best practice' in large-scale PA, how did they evolve, and will they ever filter down to more modest gig venues?

Here's a chance to show off what you know about live sound engineering. Simply complete the following sentence:
The function of a PA system is to...

That wasn't hard, was it? But in case you're struggling, the function of a PA system is to deliver your sound to the audience, and deliver it well. It's as easy as that. But hang on, it doesn't seem to be all *that* easy, does it? Whenever have you experienced perfect



sound as an audience member? And when have you ever felt that your band's sound has been delivered to the audience as well as it should have been? There must be additional criteria that need to be fulfilled to achieve satisfaction. And yes, there are. Three...

- Adequate level, in relation to purpose (clearly, heavy rock music needs to be louder than a classical guitarist).
- Low distortion, low noise and a flat frequency response.
- Adequate clarity, in relation to purpose (speech requires near-100 percent intelligibility; all the words in a theatre musical must be easily understood; other forms of music may not need to be absolutely crystal clear).

Achieving adequate level is never a problem. It hasn't been a problem since the 1970s, when PA systems as we know them today had fully matured. All you need is a recognition of how many watts you require for a particular venue, usually calculated by rule-of-thumb and reference to past experience, and the budget to hire enough amplifiers and loudspeakers. Achieving low distortion, low noise and a flat frequency response hasn't quite been fully solved, although if the noise level of your PA is audible to the audience there's a fault somewhere in the system: power amplifiers in general have a better signal-to-noise ratio than just about anything else you'll find in the whole of sound engineering. The frequency response of PA loudspeakers, however, leaves a lot to be desired, and it is definitely true to say that the only thing that produces more distortion than a loudspeaker is the lead guitarist's screaming Marshall on overdrive. But even though not all is yet perfect regarding the above points, most people find the sound quality of a decent PA system acceptable. And the typical sound of a PA has almost defined people's expectations of what a PA should sound like. A circular argument, perhaps, but there's a lot of truth in it.

There's still one point left unanswered: that of clarity. It is possible for a PA system to be capable of detailed, analytical clarity within itself. But when deployed in a real-life concert scenario it sounds anything but clear. You must have experienced it yourself many times as an audience member — that fuzzy mush of sound that clogs up your ears, but you can't really resolve it into music. Clarity, therefore, is the last unconquered frontier of PA. It is the last major problem that remains difficult to solve.

At this point I need to return to one of the requirements of PA that I previously said had been solved: that the PA system should be loud enough. There's no difficulty in making it loud enough, providing you have the budget — but it has to be loud enough for all members of the audience, and that's a problem that isn't necessarily solved just by spending a lot of money.

There are two scenarios here: one where the audience are seated, the other where they are standing and free to move. If the audience are free to move, it is acceptable to have different levels in different parts of the venue. Those who like it loud will gravitate towards the loudspeakers. Those who perhaps want to chat during the show will move further away. However, if the audience is entirely seated it suddenly becomes much more difficult. You don't want to deafen the front rows of the audience while leaving those at the back struggling to hear. If only certain members of the audience are delivered a level that is adequate, without being too quiet or too loud, the PA has not fully met its purpose. Let me therefore refine the requirements of PA into this simple statement: all of the audience should enjoy high-quality sound that is loud enough and clear enough.

Cover the audience, not the walls

A paramount rule of PA is to direct the sound towards the audience and not elsewhere. But how often do you see this rule flouted? The best and most classic example of this *not* being done was in several London Underground stations, some years ago. At the time, the tube network was decaying and falling into disrepair, so several stations were refurbished with bright, modern designs. Along with the visual aspects, these stations were given new sound systems too. Some bright spark designer decided that the loudspeakers should be mounted in cylinders (cylinder = tube, get it?) and several should be mounted at intervals along the platform, parallel to the platform and just above waiting passengers' heads. The result was that from any point on the platform, you could hear every loudspeaker, with delays increasing with distance. It was, indeed, possible to stand as close to a speaker as you could and still not understand what was being said! This state of affairs wasn't allowed to continue for long, and now the speakers point as they should — down at the passengers on the platform.

So the most important thing is to point the loudspeakers at the people in the most direct way possible. At the same time, consider how much sound is being 'sprayed' onto the walls and ceiling. The audience will absorb much of the sound energy that strikes them, meaning that it won't be reflected to bounce around the auditorium and cause confusion. But the walls and ceiling are very likely to be reflective, so the more sound that goes in these directions, the more mush-inducing reflections will be created.

In a situation where the information content of speech is of primary importance, the classic solution to intelligibility is to use many small loudspeakers and have them close to the people — obviously, pointing at them and not at reflective surfaces. This works extremely well and the information content gets through clearly. But this solution is not acceptable for a musical performance. The reason for this is that we expect a performance to take place on a stage. We

"If line arrays are good enough for top touring acts, surely they're good enough for the small gigging band too?"

watch the performers on the stage, and we expect the sound to come from the stage too. If the sound were coming from a small speaker mounted at just a couple of metres distance, up and to the side, that would cause a conflict between the visual and the auditory. Everything might be clear and intelligible, but we wouldn't enjoy the performance.

So the multiple small speaker solution doesn't work for performance. We need the sound to seem as much as possible as though it comes from the stage, and for this you can't do better than actually having loudspeakers at the sides of the stage, like a great big stereo system. However, there are still potential problems...

The first problem has been mentioned already and has to do with directivity. Loudspeakers naturally have a characteristic directional response — almost omnidirectional at low frequencies, tightening to a focused beam at high frequencies. Put another way, anyone sitting directly in front of a loudspeaker will experience a reasonably flat frequency response, but people sitting further and further to the side will hear less and less high frequencies, so the sound will be increasingly dull. So the 'big stereo system' style of PA suffers in that it sprays the walls and ceiling with low-frequency and low-mid energy that reflects into a confusion of reverberation, and only select members of the audience receive sound with a good balance of frequencies.

A second problem stems from the lack of directional control. Because much of the sound is spread widely, beyond the width of the audience, energy is lost. The more sound spreads out, the more thinly its energy is spread, and therefore the more level is lost with distance. This is an important point. The reason a sound source becomes apparently »

Info
Feature by David Mellor
Main photograph, JBL Vertec line array, courtesy of Harman Pro

» quieter as it becomes more distant is primarily because its energy is spread out. Yes, some level is lost through absorption in the air, but not much. It's distance that's the killer. An audience member sitting a long way from the loudspeakers will experience a distant and therefore quiet sound, while audience members close to the speakers are getting their heads blasted off!

Let's think in terms of light. Take a torch bulb. Intrinsically it emits light almost equally in all directions, so by itself it isn't much use for finding your way in the dark. But put a reflector behind it and a lens in front of it, so that its energy is concentrated into a beam, and you will notice immediately that it is now usefully bright. You'll also notice that the beam extends into the distance. So not only do you see the immediate area in front of your feet, but the area beyond where you direct the beam. The area of coverage is less, but you can now see where you're going. If the same could be done with loudspeakers, there would be two benefits: one, that the sound is focused on the audience and away from reflecting surfaces; and two, that the sound retains its level as it travels. So the audience members at the back are served as well as those at the front, and the difference in level between front and back is much less.

Directivity theory

If you understand the theory behind the directional characteristics of sound sources, you'll be in a good position to understand PA loudspeakers and get the best out of them. There are two extremes of directionality, between which there are other interesting cases. One extreme is the point source, which is a source of sound that has zero size. OK, there's no such thing as zero size, but in practice if a sound source is dimensionally smaller than the wavelength of sound it is emitting, it has the characteristics of a point source. The low-frequency output of a small loudspeaker would be a real-life example.

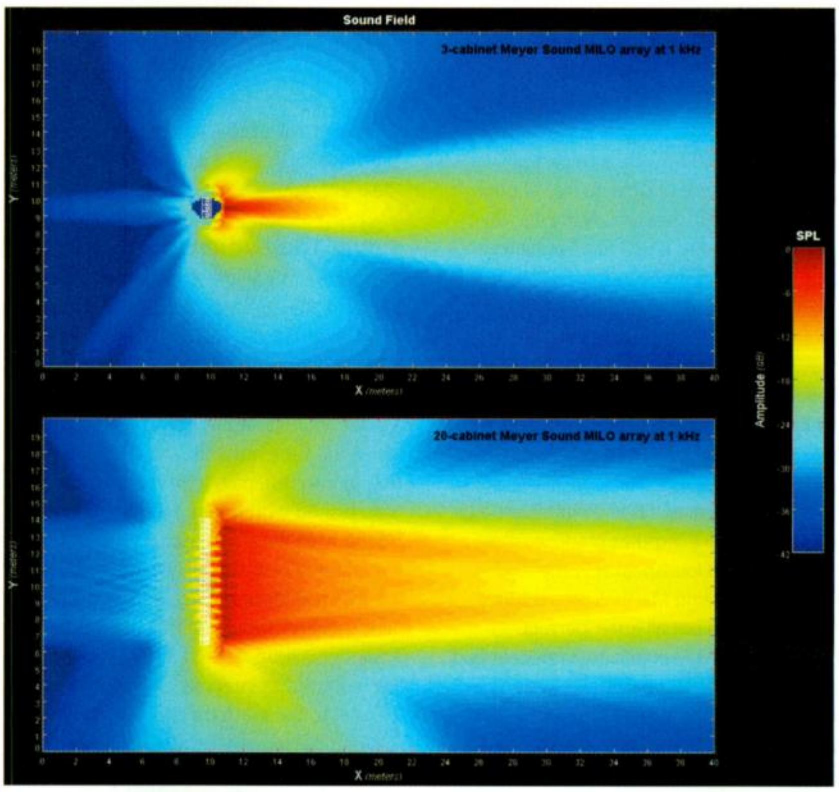
A point source emits sound equally in all directions. There you have it: all you need to know about the point source! Well, not quite all… but you'll need a little imagination. Imagine this very small point source pulsating outwards momentarily, just once. A sphere of high pressure leaves its surface and radiates outwards, becoming larger and larger. The point source has put a certain amount of energy into this pulse, and that same amount of energy over time has to cover a larger and larger area, the surface area of that continuously expanding sphere. I could at this point bore you to tears with detailed calculations concerning the surface area of a sphere, energy density and stuff like that, but instead I will cut directly to the chase and say this: for a point source, sound pressure decreases by 6dB for every doubling of distance. We call this the inverse square law.

One mistake or over-simplification is that it is commonly said that all sound obeys the inverse square law. This is not so. Only sound from a point source obeys the inverse square law. Any sound source that is not omnidirectional does not obey the inverse square law. (If you get so far away from it that visually it recedes to a point, from your point of observation it will appear to obey the inverse square law, but in practical terms this is not relevant to PA).

From this we can derive two interesting facts. The maximum rate at which sound level can decrease with distance is 6dB per doubling of distance. The only way sound can decay at a faster rate than that is if you actively do something to block it. Also, sound sources that are directional decay at a rate that is less than 6dB per doubling of distance.

It's interesting to consider the opposite extreme. Would it be possible to have a sound source, the level from which does not decay at all with increasing distance? Amazingly, the answer is yes. It is possible to have a sound source that is so focused that it will cover an amazing distance with hardly any reduction in level. You want an example? I'll give you two examples: an old-fashioned ship's speaking tube, and a tin-can telephone. We call this kind of sound source a plane source. In both cases, the sound energy isn't just focused, it is constrained to travel within an enclosed medium so that it cannot spread out at all. And since it cannot spread, no level »

Predicting loudspeaker behaviour with *MAPP*



MAPP Online is the *Multipurpose Acoustical Modeling Program* developed by loudspeaker manufacturer Meyer Sound to model the sound fields developed by its products in a variety of configurations. A sound designer is able to enter data into *MAPP Online*, including individual loudspeakers and arrays, then click the 'predict' button and get a graphical display of the expected coverage. Meyer Sound claim that *MAPP* will allow the user to:

- Plan an entire portable or fixed loudspeaker system and determine delay settings for fill loudspeakers.
- See interactions among loudspeakers and minimise destructive interference.
- Place microphones anywhere in the soundfield and predict the frequency response, impulse response and sound pressure at the microphone position.
- Refine system design to provide the best coverage of the intended audience area.
- Use a virtual equaliser to pre-determine the correct settings for best system response.
- Gain load information about the array, to determine rigging capacities.

Clearly, a system such as this, that is accurate in its predictions, is a tremendous tool for the sound designer. *MAPP Online* will run on Windows, Macintosh or Unix computers, and data that the user enters is sent to Meyer Sound's servers, where the analysis and prediction is made, then delivered back to your desktop. The example shown is a composite of two predictions made for arrays of Meyer Sound MILO cabinets, one with just three loudspeakers, the other with 20, both at 1kHz. It is clearly possible to see how much more directional, and how much louder, the larger array is. You can also clearly see the 'side lobes' that develop — an unfortunate by-product of all line arrays that the sound designer must take into account.

» is lost. (In practice, a little level is lost, but nothing's perfect.) You can see that this is not a practical way of delivering your sound to the audience, so we will leave it as a curiosity, but a curiosity that demonstrates a useful principle.

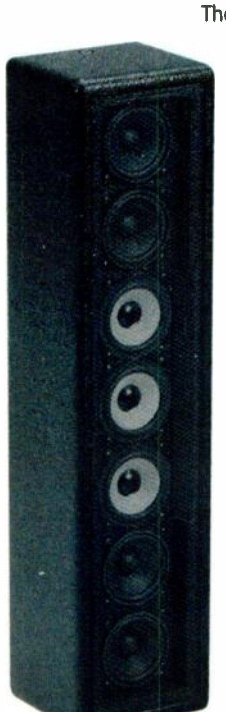
The next type of sound source is the whole purpose of this article, and is the salvation of PA as we know it. We call this type of source — fanfare of trumpets — the line source. To understand it, lets go back to the point source for a moment. I said that the point source (which is omnidirectional) needs to be small in comparison with the wavelength of sound that it is emitting. The converse is true too: when a sound source is larger than the wavelength it is emitting, it becomes more directional. And the larger it is, the more tightly directional it is. So a really large sound source would be tightly directional. This is what we want: a source that can be focused and directed to cover the audience, but not wasted on other areas of the auditorium.

But imagine you're a loudspeaker looking out from the stage to the audience. The audience in front of you are spread widely from left to right, but from top to bottom — in perspective, from the rear rows to the front — there is only a narrow spread. You can see the problem. If you made a large loudspeaker that focused the sound tightly enough to direct sound accurately in the vertical dimension, it wouldn't cover the full width of the audience. And vice versa: if it covered the full width, you'd end up covering the ceiling as well, and we know that's a bad thing.

The solution is to devise a loudspeaker that is tightly focused in the vertical dimension but spreads sound widely in the horizontal dimension. To do this, the speaker needs to be large vertically, but small horizontally. Like a column, in fact. And here we have it (bigger fanfare of trumpets): the column loudspeaker! Did I say 'column loudspeaker'? Sorry, I must use the more up-to-date and exciting terminology: line array. They are both examples of the line source.

The column loudspeaker

I often think that one of the best lessons of the past is not to go there again. However, the column loudspeaker has as important a place in the history of PA as the electric guitar does in rock music. Yes, really. One day there might be people who make a living as historians of PA, and they'll be able to tell us exactly how the column loudspeaker came to be developed. Until then, my guess is that it developed by chance and was found to work effectively. It seems like a natural development for a 1960s band to have speakers at either side of the stage for the vocals. Then they decide they want to be louder and need speakers with multiple drive units. But speakers that are wider take up more stage area, so they choose speakers that are taller. The typical pub band of the 1960s would therefore have a pair of column loudspeakers, for vocals, that typically would contain four 10-inch or 12-inch drive units, sometimes topped off with a small horn (for example, the WEM Vendetta). Although they might seem primitive now, in fact they worked surprisingly well. The small horizontal dimension meant that the full width of the audience was covered, while the large vertical dimension ensured that the sound was 'beamed' to the back of the room. However, the next generation of bands working at a higher level of the business moved on to 'bins and horns'. (A horn loudspeaker is the most efficient way of converting amplifier power to sound. A 'bin' is a bass loudspeaker, which is commonly in the design of a folded horn. 'Bin and horn' systems of adequate physical size can sound very good, but their directionality is not necessarily well controlled.) Small bands followed suit with similar but scaled-down systems, and the column loudspeaker was forgotten. Small column loudspeakers, however, continued very successfully in speech PA, such as for places of worship, where intelligibility is all-important (see the photo below). The 'bin and horn' system amounted to nothing more than the 'big stereo' commented on earlier, and directional control was lacking.



The EAW CLA37 column loudspeaker uses seven 3-inch drive units to achieve a coverage of 120 degrees horizontal x 30 degrees vertical, thus controlling the vertical dispersion tightly. It is suitable for speech reinforcement in large reverberant environments if several or many units are distributed amongst the listeners.

The next real development in PA technology was the centre cluster, much used in musical theatre. The centre cluster relies on another directional technology known as the constant directivity horn. The idea here is to combine multiple full-range loudspeakers, each of which is designed to have a consistent directional pattern over a wide range of frequencies. Horn loudspeakers can be designed to do this reasonably well. These full-range loudspeakers are arrayed together into a part of a sphere and mounted high up to cover the whole of the audience. Each member of the audience is delivered sound through only one full-range loudspeaker (apart, of course, from people sitting exactly on the dividing line between the coverage of two loudspeakers).

The centre cluster is outstanding for its intelligibility. It fulfils the criterion of directing sound only at the audience, and has the additional benefit that it forms a single sound source, therefore there is no possibility of hearing delayed sound from another loudspeaker somewhere else in the auditorium — at least, in a pure centre-cluster system. But there are two problems: the first is that ideally the centre cluster would be designed first, and then the auditorium designed around it! The second is that if each audience member is delivered sound (apart from the exception noted) by only one loudspeaker, plainly there is going to be a limit to how loud the sound can be. There will always be a role for centre clusters but, as we shall see, there are more flexible (literally) forms of loudspeaker distribution.

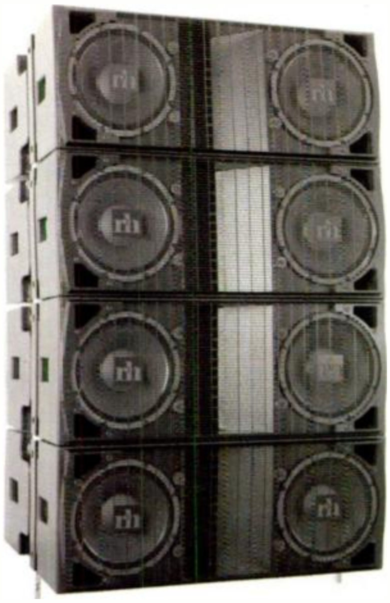
The line array

Although the column loudspeaker was effective in its context, it suffered from a lack of scale and a lack of science, each equally important. So to scale up a column loudspeaker to auditorium proportions took the best part of three decades. Still, we got there in the end. Here comes the science...

Going back to the point source, we find that level drops by 6dB for every doubling of distance. With the plane source, the level doesn't drop at all. So is there an in-between condition where the sound level drops by, say, 3dB? Yes there is, and it is the line source, which in theory can produce a cylindrical wave, as opposed to the spherical wave of the point source. A genuine cylindrical wave will have 360-degree dispersion in the horizontal dimension and zero dispersion in the vertical dimension. Any real-life source is going to be an approximation of this, but if someone offered you approximately £100, you would accept £75, wouldn't you?

Earlier, I said that to achieve directionality a sound source needs to be larger than the wavelength it is producing. To achieve focus, or near-zero dispersion, which is a more stringent requirement, it needs to be somewhere approaching four times the wavelength. The wavelengths of audible sound extend all the way to 17 metres (20Hz) and beyond. But taking a reasonable lowish frequency of 170Hz with a wavelength of two metres (taking 340 metres per second as a nice round figure for the speed of sound), a line source eight metres high will be necessary. Quite tall! But at least we have a notion with some science behind it.

The next question is: how exactly do you make a loudspeaker that is several metres high? Currently, the way to do it is to stack multiple loudspeakers on top of each other. But instead of stacking 10-inch or 12-inch loudspeakers featuring identical drive units with poor HF response, as they did in the



A four-module array of Renkus-Heinz STLA/9 cabinets.

1960s, each loudspeaker consists of LF and HF drive units and covers the full audio range (down to a reasonably low frequency). Also, rather than making one very tall cabinet, the modern line array consists of multiple small cabinets. The benefit of multiple cabinets is that you can assemble a line array that is as big or small as you like, or can fit in, or can budget for. You can also manipulate the shape of the array, which, as we shall see shortly, has significant benefits. Time for more science...

Since the line array is not actually one single tall-but-narrow drive unit, but is made up from discrete loudspeaker cabinets, one has to ask whether the individual units will couple together as though they were a genuine line source? The answer is yes, they will, but only where the drive units are separated by less than half a wavelength. This is easy for the lower frequencies, but more difficult to achieve as the wavelength shortens. As a benchmark, the wavelength at 400Hz is around 85 centimetres. So to couple at 400Hz the cabinets have to be less than 42.5 »

Deploying the line array

Clearly, you're not going to have a full-scale line-array system in the back of your band's Transit van. In fact, playing through a line array for the first time may mark your transition from wannabe band to successful band. But there will come a time, hopefully, when you are called upon to have an influence in the specification of your touring PA system. At first, the line array looks intimidating. All those cabinets, all that cable. Who's going to go up there and string the whole thing together? The answer is nobody, because the system is assembled at stage level and the whole thing hoisted up. The motorised hoists even have remote controls so that no-one has to shout instructions or converse through an intercom. A line array can actually be set up by as few as two or three people. Any kind of flying, however, involves considerable responsibility, and manufacturers are keen to use the words risk, damage, injury and death frequently in their operators' manuals. Apparently, the most dangerous part of the rigging process is when the equipment is at stage level. As it rises into the air, providing everything is done correctly and the equipment is in good condition, it flies out of the danger zone.

Setting up a conventional PA system on stage involves a certain amount of use of of rules-of-thumb. The line array is far too big a thing to set up in the same way, and once it's set up you don't really want to have to move it, so you need to be sure that the positioning is right, the height is right, the horizontal angling is right, and — most of all it — that the array takes up the optimum J-shaped curve to distribute sound evenly to the front and back of the audience, and everyone in between. To make this possible, manufacturers commonly provide software that can be used to calculate all the necessary parameters, examples of which are shown on the following two pages. Meyer Sound are good enough also to advise equipping yourself with binoculars, laser measuring tool, pedometer, laser inclinometer and a self-levelling, four-way laser. That should really be a last resort, as any decent venue should have a set of plans with accurate measurements!

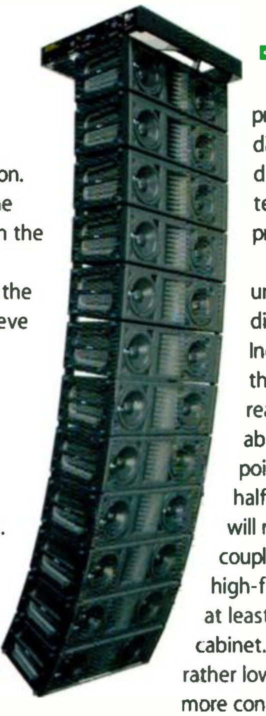
» centimetres high. OK, that's doable, but we are not even halfway up the audio band here.

Still, at least we know the criteria to aim for. The longer the array is, the more tightly directional it will be in the vertical dimension, and for individual cabinets to couple well into the array, they have to be small vertically. The better both of these criteria can be achieved, the more controllable the beam of sound from the array will be. A good point is made by Ralph Heinz of PA manufacturers Renkus-Heinz: "The answer to the question of whether a line array is a line source is 'almost never'." Heinz's comment demonstrates that a theoretically perfect line source is virtually impossible to achieve. Only the best line arrays will come close.

Waveguide

I wouldn't be surprised if some of the readers out there are microwave engineers concerned with the efficient transmission and reception of microwave signals, *SOS* readers tending towards the technical. To you guys and girls, I'd like to say thanks — you gave us all the technology we need to make great-sounding line arrays. Seriously, a lot of loudspeaker technology does borrow from microwave technology, as the wavelengths of microwaves and sound waves are comparable. I have said already that to couple together into a line source, or at least a close approximation of a line source, individual sound sources must be no further apart than half a wavelength. You can turn this around and say that the closer together the individual sound sources are, the higher up the frequency spectrum line-source behaviour will be maintained. So each individual cabinet must be as short as possible in the vertical dimension. For preference, the height of the cabinet should be no more than the diameter of the low-frequency drive unit plus the thickness of the cabinet walls. However, to achieve a high sound level, clearly the low-frequency drive units will have to be reasonably large. In the Meyer Sound M3D, for example, 15-inch (38cm) drive units are employed on either side of the high-frequency unit. Since 38cm is half a wavelength at around 450Hz, an array of M3D cabinets will approximate to a line source up to around this frequency. Above 450Hz, the directional characteristics will begin to depart from the ideal cylindrical wave, although not immediately.

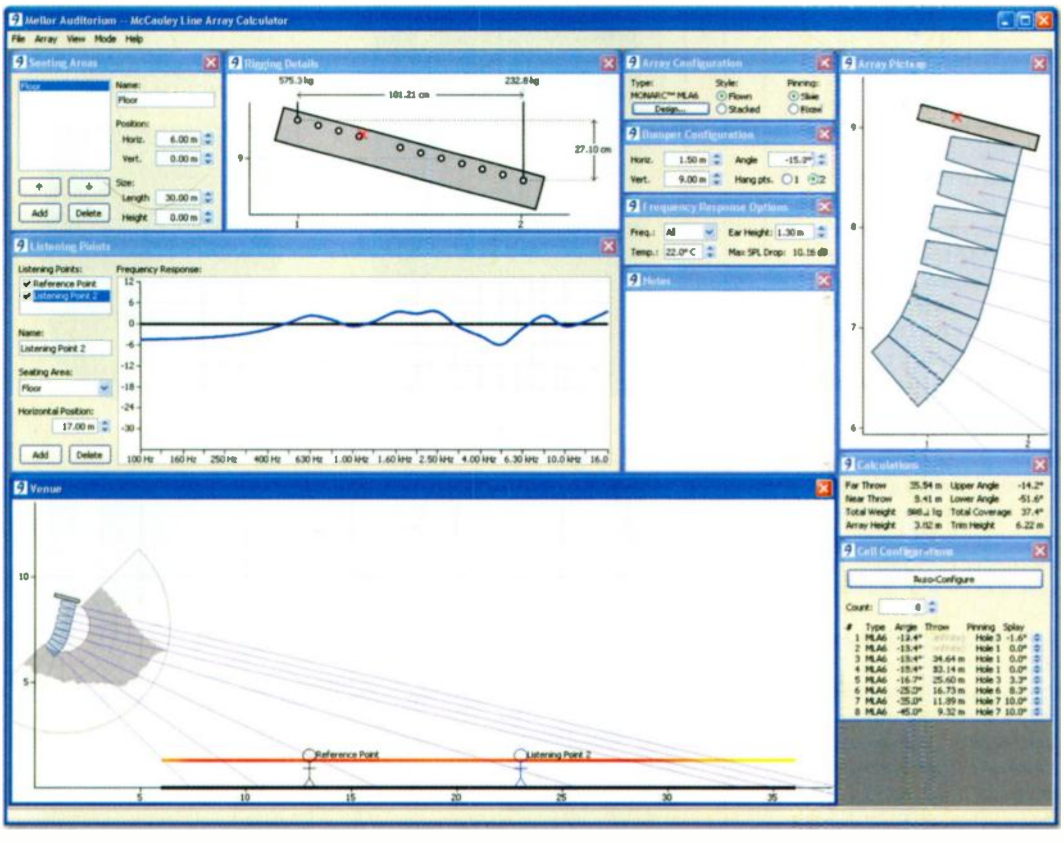
So what happens above 450Hz in the case of the M3D? At 580Hz the signal is crossed over from the low-frequency drive units to a specially designed high-frequency driver. What is special about the design? Well, to make the whole concept of the line array viable, each individual cabinet has to be a line source in its own right, or at least approximate a line source as closely as possible. For this, the high-frequency drive unit needs very sophisticated design to produce the required wavefront that diverges hardly at all in the vertical dimension. There are several possible techniques for doing this — some practical, some not.



A Meyer Sound 12-cabinet MICA array.

One possibility is the ribbon drive unit, which basically has a long, thin diaphragm up to around 15cm high. Incorporated into line-array cabinets, the ribbon driver will display at least reasonable line-source behaviour above around 4.5kHz, but below that point adjacent units will be more than half a wavelength apart and therefore will not couple correctly. For good coupling at higher frequencies, the high-frequency driver should radiate over at least 80 percent of the height of the cabinet. Ribbon drivers, in any case, are rather low on output when compared to more conventional compression drive units. A horn with compression driver would be another possible choice, but for a horn to have a suitable direction pattern and a mouth area covering 80 percent of the height of a typical enclosure it would need to be inconveniently long. A reflector can also be used to focus sound, as in the Nexo GEO system. However, it seems that the current favourite technique is the acoustic lens.

It's worth thinking for a moment about how an acoustic lens could be created. A lens for light works by slowing down light rays in a transparent medium of higher refractive index than air — i.e glass. This could be done for sound. Simply form a suitable medium into a lens shape and situate it in front of the drive unit. Sounds too simple to work? No, not at all, and this technique is indeed employed by Electro-Voice and McCauley. The lens is made out of foam, which acts as an 'obstacle array'



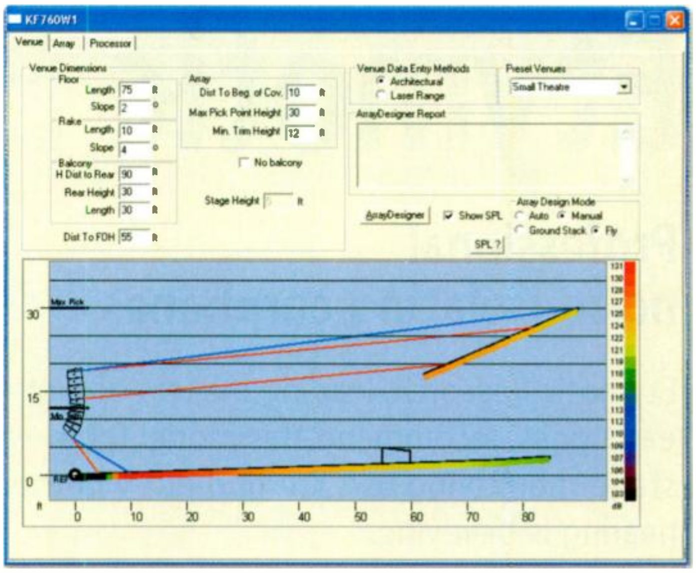
Many manufacturers of line array systems provide software for calculating optimum configuration and placement, which clearly will depend on each individual venue. Here we can see a calculator from McCauley that's particularly striking in its visualisation. You can enter the type of cabinet and quantity to be deployed, and the dimensions of the area to be covered. The software instantly shows the necessary angling of the cabinets (top right) with rigging information (top centre-left). In the lower-left panel, on the left we can see the line array surrounded by a graphic showing the vertical dispersion pattern. Listening points can be selected and the frequency response at those points will be displayed in the panel at centre left.

around which the sound wave has to pass, thus slowing it down. The foam doesn't have to have the conventional lens shape, as it can be of variable density, which provides the 'shaping'. Foam does have its limitations, as you would expect. At high frequencies it will absorb sound rather than slow it down, and at low frequencies it will have no effect. Nevertheless, the fact that it is used for some current line-array systems demonstrates that it is a viable solution to the problem.

The other way of producing an acoustic lens is the path-length refractor. This uses metal plates to direct sound through channels. The channels have varying lengths and therefore sound can be slowed down by varying amounts of time. With appropriate design, this can form a perfectly viable lens that works over a reasonably wide range of frequencies. Obviously, since there are four different techniques currently in popular use in this application, the ultimate solution hasn't quite been found yet.

Intensity shading and divergence shading

We've covered a lot of technical material so far, and it's worth going back for a moment to the purpose of the line array, which is to deliver sound to the entire audience, at pretty much the same level, all the way from the front to the back. It does that by focusing the sound vertically while allowing it to spread out horizontally. Even though a well-designed line array can achieve that reasonably successfully, it will remain the case that the front of the audience receives a higher sound-pressure level than the rear of the audience — which, of course conflicts with our requirement. The solution to this is intuitive: simply reduce the output of the lower section of the array. This is known as intensity shading. The front rows of the audience are much closer to the lower cabinets than they are to the upper cabinets of the array, therefore reducing the level from the lower cabinets will deliver a lower sound-pressure level to the front of the audience. However, there is a problem here: the front rows will still hear sound coming from the upper cabinets of the array, and they will hear it clearly because these cabinets are louder. But sound from the higher cabinets will be delayed with respect to the lower cabinets, and that will create an interference pattern and an uneven distribution. This problem could be tackled with equalisation and delay, but that would destroy the elegant concept that the line array is.



This EAW calculator superficially doesn't look as polished as the McCauley one, but it offers presets for different types of venue. This example shows a small theatre with a balcony, which is also covered by the line array. The software shows, by colour coding, the levels that can be achieved in different sections of the venue.

The alternative to intensity shading is simple and obvious, and you would probably do it by instinct anyway. When a line array is flown, it will take you precisely two seconds to observe that the front rows of the audience are almost underneath the array, whereas the rear rows are much more on the same level as the top of the array. So it seems appropriate to curve the lower section of the array so that it points down at the front of the audience. You have just created the familiar 'J' shape of the practical line array (see screen above). You have also implemented divergence shading. Simply by angling the cabinets apart more, you have required the sound they produce to cover a wider angle, therefore its intensity will be reduced at the listening position. Ideally this requires a more divergent cabinet for the curved part of the J, which manufacturers solve by designing specific long-throw and front-fill cabinets.

Line arrays for the gigging band

If line arrays are good for top touring acts, surely they're good enough for the small gigging band too? In my view, it can only be a matter of time before manufacturers of small PA systems (many of whom make large-scale systems too) bring the line array into the small pub and club venue. There is a vacuum at the moment that desperately needs to be filled. Oddly enough, since a large-scale line array is composed of multiple small cabinets, there is absolutely no reason why you couldn't stand one on stage and stack it all the way to the ceiling, taking safety precautions of course. The limitation on small-scale deployment of line-arrays is actually ceiling height. In a large auditorium, the line array is hoisted high over the audience, so that the lower section of the J-shape points down at the front rows, while the upper cabinets point roughly horizontally at the rear of the audience. Raising the array like this reduces the difference in distance between front and rear, thus reducing the level difference due to distance. In a small venue, the line array would fire into the audience as much as it fires over their heads. Although the approximation to a cylindrical wave it produces would be advantageous, the loss of the downward perspective severely limits this advantage. And, of course, in a small venue the audience bunch up around the stage, so the front rows are very much closer to the loudspeakers than the people at the rear.

By now, you should be realising that there's a awful lot to know about line-array technology. There simply isn't room to cover it all here, plus the top manufacturers are constantly researching new developments, particularly with regard to focusing and steering of arrays. Although we wait for line-array technology to re-emerge as a major force at the gigging band level, my expectation is that it will. Although line arrays need to be large to work at their best, in respect of directional characteristics, there is no reason why smaller bands should not take advantage of the technology. Indeed JBL have scaled down that contained in their large-scale Vertec series (shown at the start of this article) into the new VRX932LA, designed for smaller venues. Each cabinet contains a 12-inch LF drive unit and an HF horn, designed for arrayability. Practical array sizes start at just two or three cabinets, and JBL advise up to six for optimum control over dispersion. A six-cabinet array would have to be flown, just like a full-scale line array, but JBL have cleverly provided the option of mounting two cabinets on a tripod stand, or on a pole on top of the SRX718S subwoofer.

An understanding of the directional properties and coverage of loudspeakers and arrays can only benefit the successful delivery of sound to the audience. Great sound should not only be the province of the large-scale auditorium PA, but should be available to all, at a reasonable price. SOS

Some contacts

- www.eaw.com
- www.electrovoice.com
- www.jblpro.com
- www.meyersound.com
- www.mccauley.com
- www.nexo.fr
- www.renkus-heinz.co

Etymotic ER4S

Professional noise-isolating earphones

Etymotic Research, leading makers of high-quality earphones, recommend this model for in-ear stage-monitoring use. We find out whether hearing is believing.

In-ear monitoring has been with us for some years now, and it divides people into believers and cynics more than any other technology I've come across. I think that this is to do with the fact that the more isolation the manufacturer manages to achieve, the more it becomes clear that we subconsciously use our ears to adjust to our surroundings. Not only are our ears crucial to our balance (which is why we feel disorientated and off-balance on aircraft when there are pressure problems), but our brain also interprets incoming sound for sonic clues about the space we are in. I have some great Bose noise-cancelling headphones which some friends absolutely hate, because although the headphones sound great, they make them feel 'weird.' If this happens even before you go to in-ear monitoring, the effect is likely to be exaggerated with the increased isolation that results from the ability to exclude external sound via a seal between the ear canal and the phones themselves.

A word in your ear

This kind of isolation is exactly what Etymotic Research have managed to produce, using a dual-option monitor earphone that offers a choice of three-flange plastic eartips (giving 35dB of isolation from external sources) or foam tips (giving 42dB of isolation). The earphones are supplied in a neat protective plastic storage case with foam insert, and come with two spare pairs of three-flange tips and various other accessories.

Crushing the eartips before you put them in your ear means that the tips then expand back out to form a good seal with your ear canal, which can be improved, in the case of the three-flange plastic tips, by applying moisture before you insert them. The whole process of inserting them is relatively easy and gets better with practice, but it can be a bit tricky to get them in exactly the right position, and if you don't achieve this the bass response can be a bit weak. However, while you move them around it becomes instantly clear when they're pointing in exactly the right direction, as the bass response improves significantly. It can take up to 30 seconds for the pressure to equalise, which means that the initial sound that you hear is not very accurate. After a minute or so, once everything has settled down, the situation improves remarkably.

The effect is quite astounding, as the environment that the sound was recorded in (or the artificial reverb that has been added to studio sound) is now more audible than ever. (Of course, this is precisely what can lead to the alienated feeling that some people experience.) Certainly, the ER4s are far more effective at noise isolation than even the best closed headphones.

Rehearsal test

Using music sourced from a CD player, the results are sensational, but the real test is

At a glance

Etymotic Research ER4S

Pros
- Excellent fidelity and detail.
- Extremely good isolation.
- Unobtrusive appearance.

Cons
- Lengthy, fiddly insertion process.
- Slightly disconcerting physical intrusion.

Summary
Once the insertion process has been mastered, this is an extremely good way to hear everything that's happening on stage, or to enjoy the most from digital music sources.

Information
£ £199.98 per pair including VAT.
T Widget 0845 055 0005.
W www.widget.co.uk

when you use the earphones elsewhere, so I took them along to rehearsals with my band. We suffer from the usual problem of a guitarist who won't turn down (because the amp doesn't sound right). Using the earphones with a feed from the PA in the rehearsal room, I found myself not only able to hear for the first time what I was playing on the keyboards (I always go direct to the PA), but also, better than ever before, what all three singers were doing. The ER4s are particularly responsive to the frequencies at which the human voice exists. Not being a singer myself, I am unsure how successfully a singer could use them. Chewing causes the ear channel to change shape, changing the position of the earphones. I suspect that a heartfelt emotional performance might have a similar effect.

I find the rehearsal room a very problematic environment, because the guitar amps are in such a confined space, so rehearsing with these earphones has been a revelation. Not only is the clarity there, but you also hear a much more faithful reproduction of the sound coming from your instrument, as the colouration caused by the imperfections of the PA system and the effect of the room on the sound are completely removed.

The ER4P

We asked record producer Martyn Phillips (Jesus Jones, Londonbeat, Roachford, Erasure) for his thoughts on the ER4P version of Etymotic's earphones (also £199.98), which have 10db increased sensitivity in the mids and highs and 13dB extra at the low end, and are designed for use in conjunction with MP3 or portable CD players. Martyn comments:

"I did find it a bit tricky to get the right position for good bass response, but once you move them around you soon hit the sweet spot. There's good isolation, especially when you get the rubber seal right (then there's the benefit that you don't need to have the sound so loud to hear detail), good definition — quite a perceived lift from 1K up to 3K, peaking around 2K — and added sparkle. This does make the sound a little hard, as compared, say, to Apple's iPod in-ear phones. But as long as you don't mind the fact that the insertion process is a bit fiddly and that it's a minute or so before you hear the best sound, it's quite a rewarding experience."

Conclusion

Etymotic Research have won awards for their earphones, which are produced in various versions, and have apparently found the ER4S model popular with stage musicians. I can see why: the disorientation caused by your separation from your sonic environment soon passes, and once you relax into hearing what you (and others around you) are actually producing, it is a very liberating experience. While you may not want to wear them in the street, for fear of being run over, in the studio, rehearsal room or on stage you really want to hear what you are actually doing. The Etymotic ER4S earphones gave me more confidence than ever before that this is what I was hearing. They may seem a little insubstantial for £200, but they actually deliver the same sort of quality as an expensive pair of studio headphones. SOS

Test info

Reviewed by Paul Wiffen

This traditionally-designed mixer/amp packs quite a punch, with 1400W of power under the hood, and also boasts very sturdy construction and built-in digital effects.

12-channel powered mixer

LD Systems PM 12-2

The idea of putting power amps and a mixer into the same box together isn't a new one, but today's powered desks are offering a lot more power and a lot more features than ever before. Requiring only the addition of loudspeakers and signal sources, the LD Systems PM 12-2 mixer we're about to look at is designed as a complete all-in-one small PA system, and contains a built-in graphic equaliser section and onboard digital effects unit, partnered with a high-power Class-H 2 x 700W amplifier section and a few flexible configuration options.

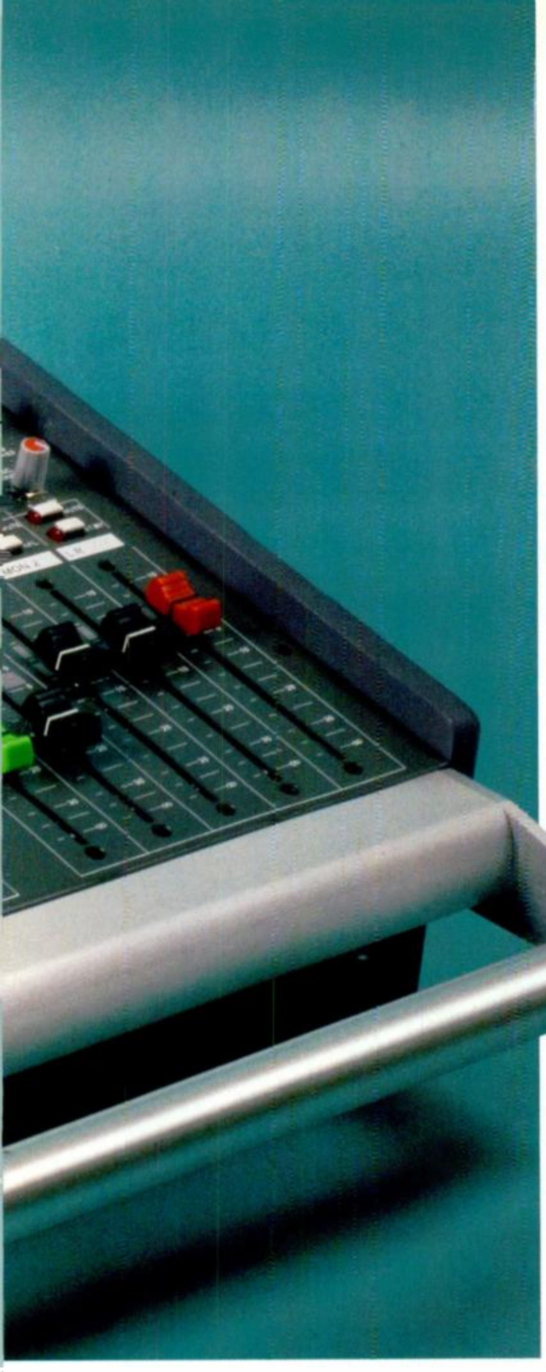
The first striking physical feature of this mixer is the substantial carrying handle, which is in the form of an aluminium bar extending right across the front panel. It's a very sturdy assembly, which is just as well, because this mixer is a fairly heavy item, weighing in at 23kg. (In an official 'place of employment', that weight would attract official guidance on how best to lift it!)

Channel hopping

The tour of facilities starts with eight mono input channels, which are equipped with everything you'd expect in a live sound mixer. At the top of each mono channel strip are the input connectors — XLRs for mics and jacks for line inputs. The line inputs are unbalanced, although the jacks are stereo and double as insert send/return points. Also present on each mono channel is a 48V phantom-power switch and accompanying LED indicator. While it's not unusual to have phantom power on a mixer at this market level, one usually finds it's

Test info

Reviewed by Mike Crofts
Photographs by Mark Ewing



Jargon explained

Bridging: A method of achieving more output from a power amp. If the same signal is applied to two channels of an amplifier, one of them with opposite polarity, a speaker ouput can be derived between the positive output of one channel and the negative of the other, allowing the amp to apply twice the voltage across the load.

Class-H amplifier: High-efficiency power amp design in which the input signal amplitude determines the power supply voltage, delivering the optimum supply to the output devices at all times.

Graphic equaliser: Tone controls where physically adjacent sliders are used to set the level in each frequency band, with anywhere between three and 31 bands used to cover the audible range. Called 'graphic' because the position of the sliders is supposed to 'graphically' represent the resulting frequency response. In practice, interaction between adjacent bands and the width of the bands themselves means that the position of the sliders is only ever something of an approximation of the response.

PFL/AFL: Pre-Fade Listen/After-Fade Listen. On a live sound desk, pressing the PFL switch on a channel will allow its signal to be heard in the engineer's headphone mix in isolation. The channel fader position has no effect on this, as the signal is picked up from before the fader in the circuit, allowing a source to be identified and checked for quality before introducing it into the mix. AFL does the same thing, but derives its signal after the fader.

Phantom power: Standardised scheme of providing an invisible (hence 'phantom') power supply voltage to capacitor (condenser) microphones using the same cable as the balanced audio output.

globally switchable, and I like the provision of per-channel phantom on the PM 12-2.

On to the control section, and the first knob in the signal path, which is the channel gain control. This is used to match the input signal to the mixer's nominal 0dBu operating input level. Just below this is a low-cut filter switch, which slopes off signals below 100Hz and is useful for reducing unwanted LF signals such as stage-floor noise, spill from bass instruments into vocal mics, and so on. The switch is a push-button type that has a nice positive action, and quite a long 'throw', which makes it easier to tell which position it's in.

Next up is the equaliser section: the HF and LF controls are fixed shelving circuits focused on 12kHz and 60Hz respectively, providing up to 15dB of cut or boost. The mid-frequency circuit is a semi-parametric design and allows the same +/-15dB of control, but with a variable centre frequency of between 250Hz and 7kHz.

The next group of four controls, labelled Mon 1, Mon 2, Aux and FX, is for the auxiliary sends. The Mon 1 and 2 sends are always pre-fader and are primarily designed for providing two independent monitor mixes to the stage; the 'Aux' send is switchable between pre- and post-fader, so it can be used as a third pre-fade monitor feed, or as a post-fade effects/recording send, or to feed an auxiliary PA, such as a theatre foyer system or outside relay. The FX send can be used to send an external signal (for example, in case four foldback mixes were required), but this control taps off the channel signal post-fader and cannot be switched. Its normal function is to feed the input of the internal effects processor, and therefore requires no external patching.

The final section of the mono channel strip contains the pan control, for placing the channel signal within the main left and right stereo image; a channel all-mute switch; a channel pre-fade listen switch; and, finally, the 100mm channel fader. The mute and PFL switches have red warning LEDs built in, which is useful, although if they had been different in colour it would be a bit easier to see at a glance which mutes and which PFLs are selected. The mute buttons affect all outputs, both pre- and post-fade, and effectively kill the channel in question.

The PFL function is essential for achieving optimum channel-gain settings, and for listening to individual channel signals in the headphones: the gain of a channel can be adjusted while you use the PFL switch, even if the channel is muted. One further feature of the PFL button is that its associated red LED functions as a channel peak-signal indicator, which will light up when the (pre-fader) channel signal reaches about 5dB before the onset of clipping. The channel faders move smoothly over their full travel and are 'calibrated' to a nominal operating position (labelled '0'), but a further 10dB of gain is available above this position.

Channels 9/10 and 11/12 are arranged as two stereo channel strips, with independent left and right line-level jack inputs on each. They don't feature microphone preamps. If a source is connected using only the right (lower) jack socket, it will be treated as a mono signal and fed equally to both left and right sides of the strip. Stereo channels have a simpler, two-band equaliser section; all other features for the stereo channels are the same as for the mono channels, other than the Pan knob, which becomes the 'balance' control, and the use of grey fader caps instead of white ones.

Master section

The clearly laid-out master section incorporates all controls for the main and auxiliary outputs, returns and effects. At the top of the mixer is the main patchbay section, which includes all inputs and outputs except for loudspeaker connections and mains power.

»

At a glance

LD Systems PM 12-2

Pros

- Solidly engineered.
- Powerful and efficient amplifiers.
- Flexible and convenient.
- Per-channel switchable phantom power.
- Easy to set up and use.
- Helpful UK distributors.
- Difficult for someone to run off with it.

Cons

- Heavy.
- Pricey for only eight mic channels.

Summary

A robust integrated mixer/amplifier, benefiting from a powerful amp section and on-board effects. However, additional input channels would increase its appeal and potential.

Information

£ £822 including VAT.
T Adam Hall +44 (0)1702 613922.
W www.adamhall.co.uk

LD Systems PM 12-2

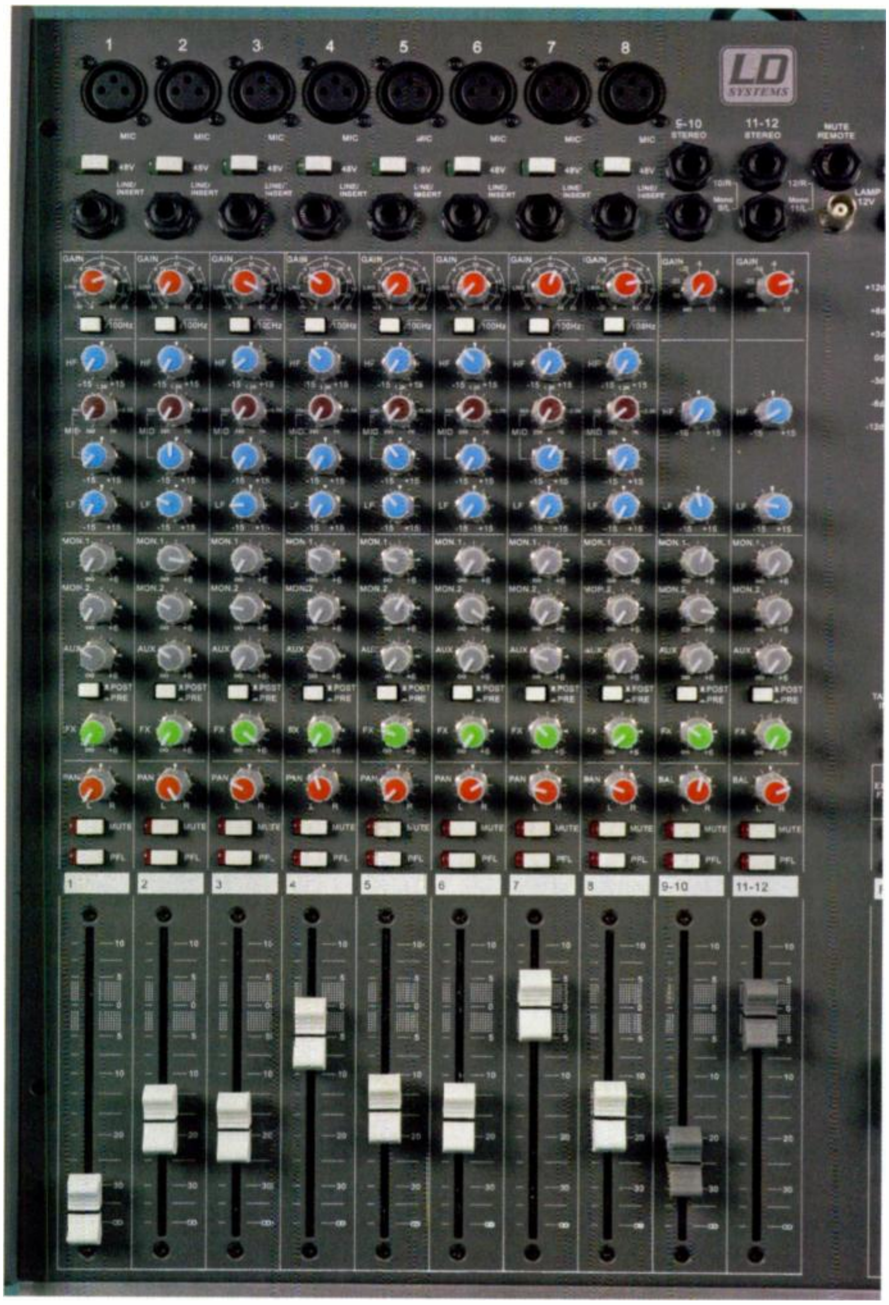
Powered mixer

» Hundred-millimetre faders are used to control the output levels of the mixer section, and it's a nice touch that faders, rather than rotaries, have been used for the auxiliary outputs (FX, Aux, Mon 1 and Mon 2) as well as the main L/R signal. The red fader on the extreme right is the stereo master output fader, controlling the level sent to the power amps or to the main and mono line outputs. Although a single fader controls both channels, there are individual rotary controls at the power-amp inputs, so that the left/right balance can be adjusted if necessary. Directly above this fader are two buttons: one that mutes the main output (called 'Mute', yes); and one, labelled Standby, that mutes the entire mixer apart from the tape-return path. This is a very useful feature, making it possible for the front-of-house and monitor sound to be silenced while background music is played, without the need to change a single fader position. The Standby switch is also a good last resort if something goes seriously wrong with the system at a very bad time!

The FX, Aux and Mon 1 and 2 busses, as mentioned earlier, are similarly equipped with faders, each of which has a mute button for its output, plus an after-fade listen switch (pre-fade listen in the case of the FX send) so that the individual aux mixes can be monitored. A rotary level control is associated with each auxiliary master fader, and these controls are used to return the effects signal into monitor mixes (in case you want a different amount of some effect in the monitor mix than you have in the main mix), or — in the case of the FX buss — to allow *external* effects to be mixed in.

Additional rotary controls in the Master section include the 'Aux in' control, which feeds the external input signal through to the main mix and has a pre-fade listen switch for monitoring the signal, and to allow you to set the correct level. The 'Mono' control governs the level going from the mono output jack. This output can be configured as a mono subwoofer output if you engage the switch marked 'sub bass', which introduces a filter into the signal path, allowing only frequencies below 100Hz to pass through.

There are individual controls for the left and right power-amp inputs (called channels A and B), and each input channel has two LEDs to indicate signal presence and warn of clipping. This is a nice detail that gives a useful guide as to the incoming level of signal and could also be very useful for fault tracing, should the need arise. A further example of flexibility is the option — via a switch next to the channel-A control — of feeding the power amps either with the main L/R signal or the Monitor 1 and 2 signals. This is an excellent idea. If, for example, you needed more power in a large venue, you could use the PM 12-2's internal power amplifiers as monitor amps, sending the main front-of-house mix to a bigger or installed system, or to powered speakers, without having to use a different desk or external patch leads.



The mono and stereo channels of the PM 12-2.

Patch me in, Scotty

There are numerous access points to the signal paths, which allows a fair degree of flexibility. Balanced jack sockets are provided for main left and right outputs, Monitors 1 and 2, the Aux send and the FX send, while the inputs are unbalanced high-impedance TS jacks, and main stereo line outputs are all equipped with insert jacks, for connection to additional outboard processors. Two mono inputs are provided for external effects, and these can, of course, be used together as a stereo return.

Direct power-amp inputs are also available, which break the normal signal path and therefore completely bypass the mixer sections.

Wrapping up the connections are a single stereo headphone jack, a remote effects-mute switch socket, and unbalanced RCA-type connections for recording and tape replay, plus a BNC lamp socket. This will supply any 12V gooseneck light and is invaluable for working in the dark!

Graphic description

The nine-band stereo graphic equaliser sits neatly below the patchbay and employs a single set of slider controls that affect either the main left/right mix or the Mon 1 and Mon 2 mixes. The entire EQ section can be hard bypassed by means of a switch »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6. Mini Boom Stand - ideal for amp/kick-drum miking
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LD Systems PM 12-2

Powered mixer

The Master (bottom), effects (centre) and graphic EQ sections of the PM 12-2.

» located alongside — which is handy for an instant 'with and without' comparison of the overall mix.

The EQ's bands are spaced at one-octave intervals, with the lowest being centred on 63Hz and the highest on 16kHz, while the slider controls give generous amounts of cut and boost, in the region of 12dB each way. This is a facility to be used with care: although the ability to cut and boost agressively is provided, it is best used sparingly — especially at the bottom end!

Built-in effects

As the PM 12-2 is designed to provide a simple, all-in-one solution to live sound, you don't want to have to connect loads of other equipment to achieve a good basic working system. This is why it's great to have built-in effects, Although the integral digital effects processor offers fairly basic facilities, it's convenient and saves loads of time when setting up and packing away.

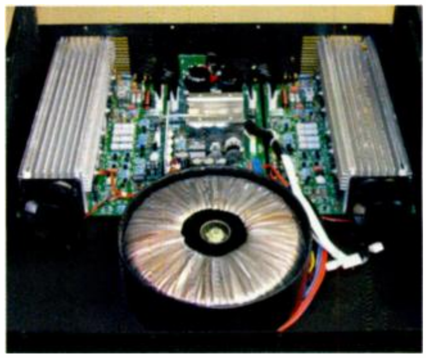
The internal effects provided are reverb, delay and echo, presented in the form of a number of preset programs that can be adjusted to provide what's needed. The display and controls are simplicity itself, with only three push-buttons and a two-line LCD panel to worry about.

Of the 14 presets provided, the first five are Room reverb settings ranging from short (0.2 seconds) to long (five seconds). The next three are Hall settings and have longer reverb times, of up to 10 seconds, while the final preset is a Church reverb treatment. There are two dedicated echo presets: 'Echo A', a mono echo adjustable between 100 and 700ms; and 'Echo B', a left/right ping-pong effect. Three further presets provide echo combined with, respectively, Room, Hall and Church reverbs. The basic reverb programs are quite usable as they are, and the small amount of user control offered — reverberation time for Room, Hall and Church presets, and echo interval (100-700ms) for echo effects, including the combined effects — should provide enough variation to cover a range of situations, although it is important to optimise the signal level sent to the effects section, to achieve the best signal-to-noise ratio.

The effects can be switched in and out (actually, the effects path is muted) by means of an external footswitch. I also discovered that access to additional effects parameters can be gained by pressing and holding the Enter button. This brings up three additional screens, where pre-delay, noise gate and mix percentage (wet/dry ratio) settings can be adjusted. Note that the manual doesn't mention these. I also established that if you adjust them, they stay adjusted after re-powering the mixer, so perhaps it's best I don't mention this at all...

Heavy metal

At last, we come to the very things that make this a powered mixer: the power amps. When the PM 12-2 first arrived, I was



▲ The power-amp section of the mixer
▼ A view of the individual circuit boards used for each channel.

surprised by the weight of the thing — it's a heavy beast, but then it does contain all of the above goodies, plus some 1400 watts of audio power.

Taking a quick look inside, it's easy to see the reason for the weight. This thing is pretty solidly built and has an old-fashioned quality about it that's quite reassuring. The two amplifiers are laid out as completely separate blocks, each with a cooling fan mounted well inside the case, at the end of an aluminium-finned heatsink. A substantial toroidal transformer sits between the amp boards (held in place by a massive bolt »

They've got it covered

The PM 12-2 comes supplied with a black padded fabric cover that features a zip-around top and a velcro front flap. The cover should protect the desk from casual bumps and scrapes during transit and storage, and is thin enough to be used inside a rigid flightcase without taking up extra space. However, it's not designed to be left in place, even unzipped, while the mixer is in use, as it would block the cooling intakes. It's worth mentioning that the mixer's plastic side panels can be removed and replaced with standard 19-inch rackmount strips, if required.

▶ Unusually, the 12-2 comes with a padded fabric cover, which should protect the finish even inside a flightcase.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» through the chassis), and the whole power section is shielded by a metal plate that's screwed directly into the heatsinks. The mixer section appears well constructed, too, and has individual circuit boards for every channel, which should make life much easier should a fault ever occur.

The power amplifier is a Class-H design, which means that it employs a variable rail voltage depending on the demand placed on the circuit at any given time. This approach offers increased efficiency over traditional class A/B designs, and results in less heat output and lighter component weight overall. The manual states that 800W is possible into a 4Ω load with one channel driven, which reduces to 700W per channel when operating in stereo. Bridging, apparently, is not allowed. The amps are protected against short-circuiting or overloading the output, DC components, clipping and over-temperature, and the two cooling fans are speed-controlled and designed to stay off — and therefore silent — when not needed (ie. in low-temperature situations).

■ Rear panel, showing Speakon speaker connectors, mains power inlet and power switch, and coolong fan outlets.

Through its paces

To fire this beast up, all you do is plug it in, connect speakers to the rear-panel Speakon connectors, and away you go. I experimented with vocal mics, high-impedance instruments (direct into the unbalanced stereo channels) and lots of recorded music that I normally use for testing and setting up.

The PM 12-2 certainly has loads of power for use in a small venue, and it delivers plenty of low-end punch even when driven hard. I hooked it up to a full set of Turbosound Impact speakers (18-inch subs with '12-inch + HF' tops) that have built-in passive crossovers, and the results were very acceptable, much as you'd expect from the maker's figures. Driving a small set of DAS eight-inchers in a conference room at a much lower level produced good, controllable performance too, and the variable fan speed meant that mechanical noise was hardly noticeable — an important point if the mixer is used for speech and audio-visual presentations. The graphic equaliser does what it's supposed to, and the graphic in/out switch is useful when you want to check and compare your settings.

The digital effects processor is simple, but it does the job and is very straightforward to operate. Although there's a limited number of presets, the user-defined settings remain in memory after the mixer is switched off, so it's worth experimenting and saving the settings that work well for particular applications. The user manual doesn't tell you how to restore the factory presets, but apparently more detailed technical information is available from the ever-helpful people at the UK distributors, Adam Hall, should you need it.

When using the internal effects, I did notice that there are guide marks on all the channel 'fx' send knobs which suggest a notional setting at the three o'clock position (about 75 percent up). These settings do, in fact, represent the kind of level you need to drive the effects processor hard enough to optimise signal/noise performance and do provide a good starting point when setting up.

I like the 100mm faders; they are consistent and smooth in operation and there's plenty of panel space between most of the controls for the average finger sizes. The main panel is easy to read, understand and operate, and it's organised logically enough — although I did wonder why the auxiliary masters are laid out in a different order to the channel Aux sends (the channel sends are arranged, top to bottom, as Mon 1, Mon 2, Aux and FX whereas the masters are arranged, left to right, as FX, Aux, Mon 1 and Mon 2).

The 12-2's 23kg weight means that careful handling is required, although the carry-bar is comfortable enough to allow you to transport it single-handed, provided you don't park the van miles away! General build quality is good, but you may want to keep an eye on the screws that secure the carrying handle, and also on the big bolt holding the transformer in place (easily visible underneath the mixer) just to check that everything stays nice and tight. A transformer of this size could do a lot of mischief if it came loose. On the review model, there was one carry-bar screw and a couple of side-panel screws missing, and the plastic window over the digital effects display was untidily glued into place.

Conclusion

Speed of setting up, flexibility and ease of use are my favourite things about the PM 12-2. It's a substantially built, one-box solution and packs a very powerful punch, much more than enough for most pub and club gigs.

On the down side, I'm not entirely convinced by the power-to-channels ratio of this product: it's got a monster power amplifier and it's a solidly built affair, but it only has eight mono channels, and I can't think of many live sound jobs where this combination would be enough for me. On the other hand, for a smallish band who want to own a rig capable of doing larger venues without having to add more equipment, or for smaller installation applications, it's definitely worth serious consideration. SOS

Mr Monitor

Chris Trimby: Monitor engineering

The award-winning Chris Trimby not only tours with top artists and spends summers working the major festivals but also mixes live sound for TV music shows including *Top Of The Pops*, proving how varied and interesting a career as a monitor engineer can be. We find out how it all happened.

Chris Trimby chuckles to himself as I enquire whether he has a musical background. "Lots of people have said I have a musical ear, but when it comes to playing the piano or guitar you might call my playing *avant garde* at best..." Being unable to play an instrument has clearly not proved too much of a hindrance for Chris, however. In 2004 he was voted Monitor Engineer Of The Year in the *TPi* [*Total Production International* magazine] Awards, and he's one of the few sound engineers on the planet to have received an Emmy Award for mixing live sound. With touring credits including David Gray, Sting, Wet Wet Wet, The Beautiful South, Tina Turner, Madness, and most recently the Sugababes, he must be doing something right.

The last time I met with Chris was a little over a year ago when we worked together on BBC's *Top Of The Pops*, a show for which he has provided the studio sound reinforcement and monitoring for the past four years or more. In fact, Chris is a veteran of providing sound reinforcement for television, a very niche market in the live sound world. As if this doesn't keep him busy enough, he also spends up to three months of the year working on the biggest music festivals in the country, looking after the 'monitor world'. If you ever find yourself standing in the stage-left wing of the main stage at Reading, V, or Glastonbury it's more than likely you'll bump into Trimby, and I'd be surprised to find any signed band left that he hasn't worked with at some point.

I'm feeling a little uncomfortable at the thought of interviewing one of the engineers who played such a huge part in starting me out in the industry. It was seven years ago when I first met him, while struggling to find my way in the audio world, doing work experience on a music festival where he happened to be 'babysitting' the monitor rig. He was kind enough to show me the ropes and organised some other opportunities for me to learn on the job, so it's almost poetic that I now find myself talking to him about how he started out.

In the beginning

"I was a printer by trade, and was doing my apprenticeship between the ages of 17 and 21, but I also drove a van for a local band in Reading. They had a small PA system that I started fiddling around with, and somehow I got a good result, so they decided to keep me on. When the band split up, I bought the PA and continued to do world tours of Reading, Oxford and Swindon! That's where I first learnt the trade.

"When I finished my apprenticeship, I went to work for Entec (a London-based PA-hire company) full-time in around 1983. I worked there for three or four years, starting in the warehouse, which was the best thing that could have happened to me. I was fixing amplifiers, fixing bins, learning about phasing, how bins and horns work, how to repair them... it was a great grounding that lots of young engineers coming up these days don't get the opportunity to gain, because the industry has shrunk. I went freelance in 1986 and have been freelance ever since."

The transition from warehouse and crew

Info
Interview by John Gale
Main photography by Richard Ecclestone
David Gray stage photograph courtesy of Innovason

to engineering bands wasn't something Chris had to wait too long for: "For me, thankfully, it happened very quickly. My first gig was Ted Nugent at the Hammersmith Apollo. I was the fourth man on the PA, plugging it up, and so on, and I did a few months of that. Then I went on to a festival tour, where I was one of the 'miking-uppers' and was in charge of the leads. I labelled every trunk, so that when anyone needed anything I always knew where it was. That impressed the people I was working for, enough that they gave me a job that was kind of half production, half engineering on a television show called *The Tube*. A guy called Gary Bradshaw, who is a fantastic FOH engineer, looked after me for six weeks while I bedded in, and then I was let loose on my own. I did that for four and half years, for six months of each year".

It's at this point that I show my age. I have never heard of *The Tube*, and this amuses Chris. After a quick Google, I discover that it was a Channel Four show with Jools Holland and Paula Yates that is now recognised to be the blueprint for many of the music shows we see on our TVs today. It ran from 1982 to 1987 and is considered a landmark in music television.

Chris: "I think the job was offered to other people, but at that age most people are more interested in touring round the world with The Stones or whoever. I welcomed it with open arms and halfway through the first series I realised that it would be a benefit for me in years to come. In January and February, when other freelancers aren't working, I am."

Chris is referring to his distinguished career as a live sound engineer in the television studio. Subsequent television shows he has mixed live sound for include *The White Room*, *Big World Café*, *TFI Friday* and *Top Of The Pops*, for which he works two thirds of the year on behalf of SSE Hire. On a show like *TOTP*, his role is predominantly as a monitor engineer for the artists that perform in the studio. The show varies in layout from week to week, sometimes with two or three stages in the television studio and with acts varying from a simple playback with a live vocal up to a full band setup. In addition, he also provides a front-of-house mix of the band for the studio audience, all mixed from a little 'cage' in the next room.

Chris: "There is no 'line of sight' to the stages. During soundcheck periods, I have a stage person with in-ear monitoring (IEM) and a radio mic communicating with me. The main issue with mixing monitors for TV is spill. My friend Dave Lee, the sound supervisor on *TOTP*, is always talking about spill and how it trashes the broadcast mixes. It's the same for gig front-of-house engineers, but it's a little more difficult with television."

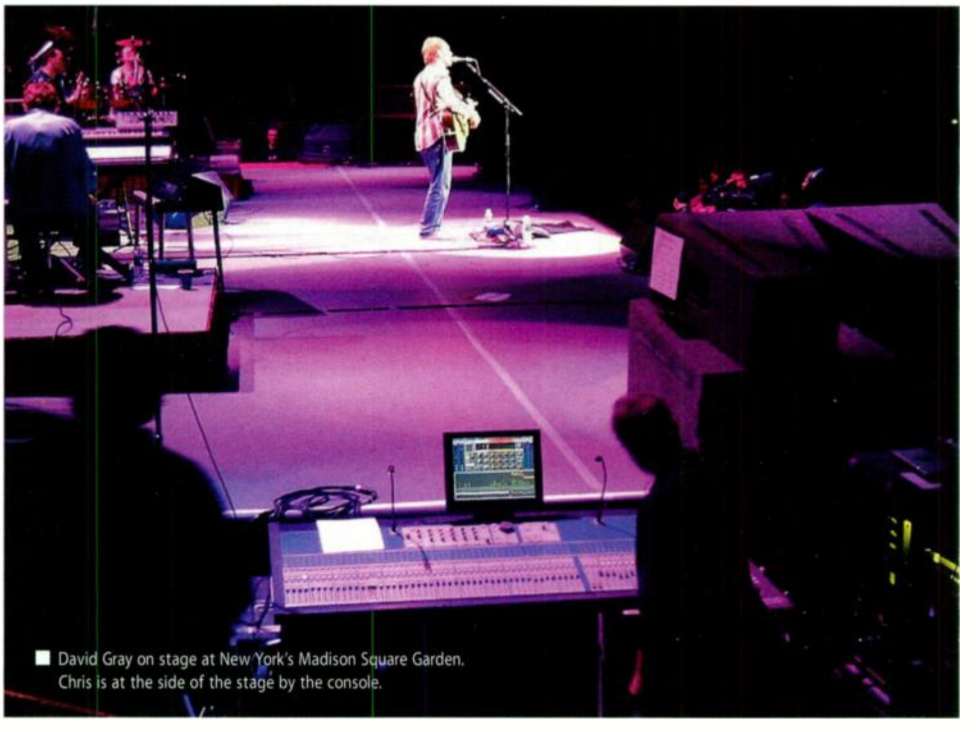
Stacks and racks

I've noticed how dramatically things have changed in the time that I've been working as a live engineer. As little as five years ago, a festival PA was a huge affair, with many speakers hung in massive cluster arrays and delay towers all over the field. In recent years, line-array PA systems [*see our feature on line arrays elsewhere in this issue*] have become the norm, and with many fewer boxes required to do the same job as before, so I am intrigued to hear from Chris how dramatically things must have changed over the past 20 years.

"When I started out, the desks were all Midas Pro 2s or Pro 4s. On the Pro 2, the input module had presence, treble and bass. That was your EQ. There were graphic EQs and reverbs, and I remember graphics with 31 rotary pots rather than 31 faders. The PAs in those days were Martin systems: bins, horns, fillers. Theatres were the venues of those days. When arenas were used and when you had to fly PA in an arena, essentially what you had was a flat bit of wood that you stacked the bins and the horns on, you ratchet-strapped them, put a curtain round it, and flew the whole thing in the air."

I think I might have been wearing my alarmed expression at this point, and Chris quickly continues...

"It was all obviously very, very safe, but that was essentially all there was. In those days, when we used to do the festivals it was pre-flying, really. We used to have two enormous wings either side of the stage that were just scaffold with double decking. We would stack Martin double bins three wide and five high, and then do exactly the same on the tier above, so it was three wide and 10 high. The bins would couple together and act like one huge bin. In the gap between one decking and the other, where the scaffolding was, we used to put bits of cardboard and then gaffa it all up. This contraption used to reach to the top of the field! It gave you a good grounding in how things work. These days the pre-packaged setups that are delivered to you are fantastic, but they don't actually teach you anything."



David Gray on stage at New York's Madison Square Garden. Chris is at the side of the stage by the console.

With all this talk of ground-stacking front-of-house PA systems, I wondered if Chris ever considered going down the front-of-house engineering route?

"When I owned my own little PA system, I used to engineer front-of-house, and then I went to work for Entec. In those days, the band would always have a front-of-house engineer but the monitor engineer would be provided by the service company. These days there is much more parity between the two roles, but once upon a time monitor engineering was considered more a sideline. Quoting a good friend of mine, Vish Wadi, he was working for this guy who had some problems with his monitor engineer, so Vish went to do monitors for him instead. When Vish got onto the board he just went 'Oh my God, this is what I want to do!' I said 'Yeah, dead right. That's exactly how I felt!' Then I asked Vish who the bloke was and he replied, 'Oh, Bob Marley'. »

Chris uses an Innovason SY80 digital console — "an incredibly powerful machine" — to mix the live acts on *Top Of The Pops*.

"I have no interest whatsoever in doing front-of-house. I just like being on the side of the stage. We were talking earlier about hand signals [*Chris is referring to an earlier conversation about how he has been known to communicate only in hand signals, even on the tour bus after a gig*]. Well, it was the other day with the Sugababes in Vienna. The bass player plays the keyboard bass at one point. He was facing away from me, and signalled for me to turn it up. I turned it up and he just gave me the thumbs-up. Didn't even look at me — he just knew that I was aware. There are nine people on the stage other than him. How did he know I'd be looking at him?"

It's completely obvious from how Chris talks that he is in love with his job, and that he enjoys working so closely with the musicians on stage.

"It is a service industry. Marty Pellow (the lead singer of Wet Wet Wet) and I used to have some major arguments in the early days about monitoring. He would say 'I don't know all that 2k, 4k stuff!' and I'd reply 'that's why you're employing me — is it thin, fat, dull? You tell me what you think of it and it's my job to interpret that.' I'm glad to say we now consider each other friends. I have this policy about being good at your job. You can pull the wool really easily over your own eyes and say to yourself that you were fantastic, but when someone else says to you that it was good that night, unprompted, that's when you know you're doing your job properly."

Touring tactics

Quite how Chris can find time to fit touring in with all his other commitments is a mystery to me. Chris: "I found it hard to get time to tour, with the TV work. Because I embraced the TV thing, the longevity of that is six months of the year for five years, with festivals in the summer. For any band that's worldwide, you have to be available to be away for long periods of time, and I can't always fit that in with the television jobs, but I enjoyed TV and thought it was better for me. Twenty-two years later I'm still doing the same job — television, festivals in the summer, and some touring. There have been times when there hasn't been an awful lot of TV and I've had to go out on the road, but I really like the balance of the two. When I got bored with touring I used TV as an avenue to get out of it, but like

Radio mic systems are used extensively on the show. These racks house Sennheiser E500 and Shure S2 radio mic receivers.

Pop goes the gear

For *TOTP*, Chris mixes on an Innovason SY80 digital mixing console, and uses SSE Hire's MB4 wedges with Camco Vortex amplifiers. He has been a great believer in digital mixing consoles since their early days, often choosing digital boards over analogue while touring, and he's very happy using the built-in processing of the console when running the show. In fact, the only outboard that Chris uses is a couple of external reverbs, as the Innovason does not have reverb built in. All EQ, including graphic EQ for the wedges, is built into the board.

Chris: "Now that it's live to air, the show has changed completely. Previously, we would record one band then stop, set up and record the next. Now, of course, we have to go directly from one to the next. This meant that we needed more inputs and outputs, because we don't have time to patch."

any job you get bored and veer back the other way. I really enjoy touring now, because I can pick and choose."

Toys vs tools

As our conversation progresses, I try to steer Chris towards discussing some of his equipment preferences. I assume that he is probably using only high-end professional equipment and I'm definitely intrigued as to whether he has any special toys...

"Although I have preferences regarding gear, I do find it difficult to talk about it. It depends on who you're working for and what budget they have. I have worked for some of the top artists, but then you invest yourself in the future by working for up-and-coming bands, in smaller venues with lesser budgets, and you still have to get the same result, although you find yourself working with your second, third or fourth choice of equipment."

I get the impression that Chris would prefer to keep things simple and affordable, and that as long as the gear sits somewhere in the professional area of the market he's happy to make it work. Eventually, however, I do get him to divulge some of his favourites. First, wedge monitors...

"I was brought up on the Martin LE200s, which became the LE400s and then the LE700s, which the SSE MB4 [*the wedge that Chris uses on TOTP*] is based on. It's essentially a single fifteen-inch low/mid driver and a two-inch LF driver with a port. It is a very natural-sounding wedge for me, and I've travelled round and been given all sorts of homegrown wedges. Those with the fifteen-inch and two-inch drivers, with a proper-sized port, have always sounded the most natural in the low end to me.

"For monitor engineers, outboard is never as lavish as for FOH engineers. I like the Yamaha SPX series and Lexicon reverbs, sometimes a Harmoniser, BSS compressors, Drawmer Gates. EQ is generally on board on the Innovason console. If anybody asks me about any sort of miking preference, I have no preference about anything except a Shure Beta 91 on the kick drum, although I'd use an AKG C414 on a vocal if it comes to it. Generally it's the FOH engineer who tends to have the choice, but I'll always take a 91. It's a boundary layer microphone, essentially a Beta 98 if you take it apart. You put it inside a drum and it picks up what the drum is doing naturally, rather than putting a mic in front of a skin.

"I loved the Sabine Workstation 4000 during the *TFI Friday* days. It's a third-octave graphic equaliser and automatic notch filter, a fantastic piece of machinery. The user manual was brilliant: it told you what it did, and then how they thought you should use it, although I don't use it anymore because the digital boards do all the EQ. It took me a while to get to know it, as you don't always get to practice with pieces of new machinery. I'm now in a position within the industry where people come to me with new pieces of equipment because of my experience, and also because of the amount of acts that come through the jobs that I do. When I'm doing festivals and television I might see 10 bands in a day."

A mixture of in-ears and wedges are used on *TOTP* for monitoring. This rack contains Shure PSM700 in-ear receivers.

Chris' way of working is to try and keep things as straightforward as possible. If he's experiencing a problem with equipment, he's a big believer in going back to basics.

"I did a gig in Bristol where they had been struggling with their monitors. The graphic looked as though someone had fallen over and pulled the faders with them. I had a listen to the monitors and they didn't sound very good, so I flattened the graphic and listened to the boxes flat. They were bi-amped, so I asked the house engineer to turn the high end down a couple of clicks on the amp. He asked if he should do it on the crossover, but I said 'no, on the amp'. We turned the amp down a little bit and the monitors sounded a lot better; a little bit of EQ and they sounded great! When you turn a crossover down, it turns down, say, 2dB of the crossover, but then if it then goes into the amplifier and the amplifier is too big, turning the crossover down 2dB in proportion to what it's going to amplify is a meaningless action. Turn the amplifier down. Quite often especially on monitors, the amplifiers are far too loud for the high end. In places where people don't have enough experience, all that is needed is someone to take things back to basics. If you find yourself hacking away with a graphic EQ, the problem is rooted somewhere else in the chain."


This Crest 12-channel mixer is used for the presenter mics and VT (videotape) playbacks on *TOTP*.

In-ear fears?

In recent years, it has become far more common for musicians to use in-ear »

» monitoring (IEM) and I wondered if Chris had any views or tips on how to get the best out of them. I remember reading an interview with Chris when he was touring with David Gray, where he mentioned that all members of the band were using IEM except the bassist. I was intrigued as to whether this was due to input from Chris, and whether he prefers the in-ear or wedge-monitoring approach.

"That's what they were doing when I got there. I like in-ears as much as I like wedges. They are undoubtedly better for FOH engineers, as there's a lot less spill, but I can achieve that with wedges too. There's a whole in-ear/wedge confrontation with people that don't really know how to use them… I've seen people with in-ears, thumpers, subs, wedges, the lot! But with a band that use a lot of in-ears on stage, what I tend to do is just have two side-fills providing a nice wash across the stage. It adds a proper low end and richness on stage, and also acts as a backup. If someone's in-ears were to suddenly go down, they could pop them out and have something to work to immediately.

"For the Sugababes, the whole band is on in-ears and the girls are on wedges, which is the other way round from what you would normally expect. There are 50-odd channels in their setup — it's huge! Drums, bass, guitars, two keyboard players, percussionists, eight channels of electrics, and the three girls."

With such large ensembles on stage, I ask whether Chris ever worries about hearing damage, considering that he's subjected to amplified sound day in and day out.

"I think you can have it as loud as you want and it won't damage your hearing. The wrong frequencies at the wrong level is

■ Chris checks that everything is working backstage. As the *TOTP* stages are moved according to the requirements of the show, the PA system is frequently dismantled and set up again.

■ Glastonbury is one of several summer festivals that Chris works.

what will damage your hearing. Loud bands are great, but vicious amounts of mid-range and high-end are not good. Obviously, sound from loud bands can damage your hearing, but I believe it's more to do with frequency than volume, or too much volume of the wrong frequency."

Doing shows like *Top Of The Pops* and the festivals, Chris is in the fortunate position of being able to engineer a lot of bands. I asked him about the approach he takes when working with a new artist for the first time.

"I have a very specific way of doing things. If I walk into a job and ask the band how they want to do it, and they say 'you tell us', then I run riot! If they don't like it, we'll try something else. Most of the time they seem to like it. If I get a band with a very specific way of doing things, then I start off doing it exactly as they like it. If I feel I can better the situation, as time goes on I might try a few things. I might try a rear

Digital debates

There's always some debate over digital mixers in the live sound world. Most live engineers either love or hate them, with very few sitting on the fence. In the early days, reliability was the issue, while some purists even now claim that digital boards sound inferior to their analogue counterparts. Chris's position in the great debate is very clear:

"I use digital boards virtually all the time now. The only thing the analogue board has over a digital board is that most people can walk up to an analogue board and use it whether they have used it before or not. With a digital board, it's a bit like me asking you to put a document together in Microsoft *Word* without you having ever used the programme before. You have to be taught.

"I was chatting with Ian Barten, who was doing the Charlatans monitors at Reading Festival a year or so back, and he patted SSE's old analogue Midas XL3 and said 'you're not going to get rid of these, are you?' It made me think. I assume that SSE will get rid of them at some point, but the fact is that a board like that is a workhorse everybody knows, and it's really easy to get a sound together quickly with it. But I prefer digital because of the sheer power and the information transfer. I can sit on the train and do my patch for *TOTP*, go in, stick a disk in and it just fires up!"

I wondered whether Chris ever worries about the fact that a digital mixer is just a computer, and computers can always crash in the middle of a concert.

"It's an interesting question. But when someone says about a digital mixer 'it's got a mind of its own'…. No. You did something you were unaware of. You clicked something twice, the cursor was in the wrong place and you pressed the wrong button because your attention was elsewhere. When these things happen, they're normally mistakes by the operator.

"Having said that, the few times I've come across problems with digital boards haven't been during a show but during the boot-up stage, with both the Innovason and Yamaha. The Yamaha and Innovason problems were literally during the boot-up sequence. In Glasgow with the Sugababes, I booted the desk and it wasn't doing what it was supposed to do, so I just re-booted and it was absolutely fine. It's like any other computer. A show stopper? Absolutely not. When I was touring with David Gray and we took our boards around the world with us, we had this problem where a desk wouldn't boot at all because the CPU had fallen out in transit. When I arrived, the tech was looking a bit sheepish, with this board literally in bits. I told him not to worry. I'm more confident about digital boards than analogue, because wherever you are in the world you can phone someone who can come out and fix a computer. They don't have to know about sound, they just have to get a computer to work!'

"I find there's actually more safety with digital. With most of the digital boards, you can hook your laptop to them and run the show off your laptop, if it comes to it — it's like having a spare board with you. In the analogue days, all you would ever take out was a spare power supply. If someone pours water into your digital desk, you can possibly mix the show on your laptop. In the analogue days, the show would be over."

wedge for a soundcheck one day, so that I can turn a front wedge down and pull it back from the edge of feedback. They might have heard from an engineer years ago that that doesn't work, and it has put them off doing it.

"David Gray soundchecked every day for at least two hours. Madness soundchecks, on the other hand, are very short. The girls in the Sugababes never soundcheck — for their age, they are extremely experienced and very professional. I like bands that jam for soundchecks too. When they come in and rehearse a song that they messed up yesterday, you find that they're not really playing. But when they jam and they're really playing, the levels are a lot different."

Winning ways

It's clear that Chris' considered approach and skill in communicating with bands have played a huge part in his receiving the Monitor Engineer of the Year award, but when asked about the highlight of his career he's in no doubt whatsoever.

"The Monitor Engineer Of The Year Award was from *Total Production International*, a trade magazine that runs a voted award ceremony. I was very pleased to get the award, because people vote for you. But I was particularly proud of the Emmy award. I was mixing for a special Twin Towers 9/11 fundraising event, where all the money went to the emergency services and families of those who lost their lives. The part from Britain was Sting and U2 doing songs, for which I was mixing the monitors. The show received Emmy awards, and I received an Emmy for Outstanding Sound Mixing for a Variety or Musical Series or Special."

It's now that I realise how much of Chris's day I've taken up, and that it's probably time to let him get back to work. Having had Chris help start me out in the live sound industry all those years back, my final question was if whether he had any advice for aspiring sound engineers trying to find their way in the live sound industry.

"I can't tell you exactly how to get into the business, but whatever break you get, no matter how big or small it is, keep it simple. Sometimes you need masses of gear because you need masses of gear, but don't get lost in technology and equipment. Remember what you are there to do: amplify the band. And don't be scared to tell them that you think it's wrong over there because it doesn't sound right over here. Sometimes that could lead to you

Catching 40 winks in the monitor position at a music festival in Ireland. Obviously, it's not all glamour!

losing your job, but you have to risk that. Something that often gets pointed out about myself is that when I speak up, people do tend to listen, because I believe in what I'm saying. If I get fired, then so be it. I've been fired twice in 20 years. If you have that confidence, of not worrying about losing your job, it gives you the freedom to be able to say what you need to say to people, and allows you to do it in the right way. Just be honest, keep it simple, and put your hand up when you're wrong!" That sounds like good advice to me. SOS

AMT SP25B

Acoustic bass microphone system

Amplifying an acoustic bass on stage is far from straightforward — indeed, the difficulties involved have been known to make players swap to an electric. Could this new AMT mic/preamp system be what upright bassists have been waiting for?

From a live-sound point of view, amplifying electric bass is usually one of the easiest tasks. In smaller venues, the backline is often powerful enough on its own, and even when the bass is DI'd or miked, feedback and clarity are rarely a problem. With acoustic basses, things are by no means so straightforward: the backline amp (if there is one) is often there as a personal monitor for the bass player and doesn't contribute much to the overall live balance, and it's a big problem trying to obtain enough gain on stage without losing the original character of the instrument.

Most bass players of the upright persuasion will have spent much time and money on their instrument, its strings and its setup. Having achieved a great tone and balance, the last thing we need is for the act of amplifying the bass to destroy most of these qualities. So if we want to retain as much as possible of the original acoustic tone of the bass, a good-quality microphone is generally the answer, and that's where the real problems begin in a live situation.

In a recording situation, you can ask the bass player to keep fairly still, and experiment with microphone placement to get the exact sound you want. In a live environment there are serious practical issues that prevent this. For example, the player will probably want to move around on stage, and the mic will have to be close to the instrument to prevent the rest of the band being picked up — and, in general, the closer the mic position, the more adverse is the effect of moving the bass even by a very small amount. Add to these woes the usual lack of setting up time, restricted stage space and so on...

Many players have developed their own ways of miking a bass, including the old wedge of foam rubber or 'cats cradle' of rubber bands in the bridge arch, but they almost always represent a compromise. Here, from specialist microphone designers Applied Microphone Technology (AMT), is a solution that has found favour with top performers: the S25B and SP25B Acoustic Bass Microphone System. Part of a range of specialist acoustic instrument mics catering for everything from harmonica, woodwind and brass to strings, piano and even vibraphone, the AMT SP25B comes in a strong plastic case and consists of the mic assembly on its mounting clamp and a metal-cased preamp. Also supplied is a hex key, for initial adjustment of the mounting clamp, and a brief user guide.

The microphone

The microphone capsule is a very low-mass, tight-cardioid electret condenser design specific to this bass system, and sits within a diminutive shockmount assembly of the metal 'spider ring' type. On the back of the mic itself is a red LED that indicates the presence of battery or phantom power. The shockmount is permanently attached to

At a glance

AMT SP25B

Pros
- Very good mechanical design and build quality.
- Overcomes all the usual bass-miking issues.
- Infinitely variable mic placement.
- Elegant and unobtrusive for live work.

Cons
- Would like a second cover-retaining screw on the preamp, as I'm always losing cover-retaining screws.

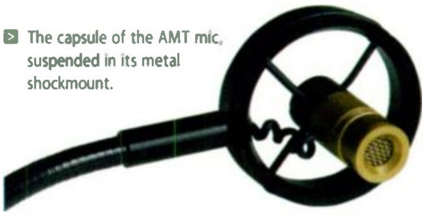
Summary
If you're in the market for a microphone-based bass solution, once you get to know and love this AMT system it should cover all possible 'amplify your bass' situations. It's not cheap, but it absolutely does the business.

Information
£ £752 including VAT. S25B (with more basic preamp) £517. Prices include VAT.
T Affinity Audio +44 (0)1923 265400.
E terry@affinityaudio.com
W www.appliedmic.com

Test info
Reviewed by Mike Crofts
Photographs by Mike Crofts

a slim metal gooseneck section, which is, in turn, mounted on a solid metal clamping bar. The whole thing is held in place on the instrument by two small metal clamp plates, that slide along the bar and clamp on the front and back of the bass; the instrument surface is protected by soft suede-leather pads that cover the inside faces of the clamp plates.

The whole assembly is finished in black, and is unobtrusive and very neatly designed. Once mounted, it feels very solid and isn't in any danger of shifting position or working loose. The gooseneck adjustment is positive and tight, and there is virtually no 'spring-back' when you're making fine adjustments to the mic position. A mic cable is permanently fixed and exits part of the way along the clamping bar, the cable exit being angled back to the rear of the bass and protected by a semi-flexible plastic sleeve. This lightweight cable seems fit for the purpose, but if it were damaged it would be a tricky repair. It terminates in a special four-pole plastic plug (with quick-fit locking ring) that's a special sealed, waterproof design



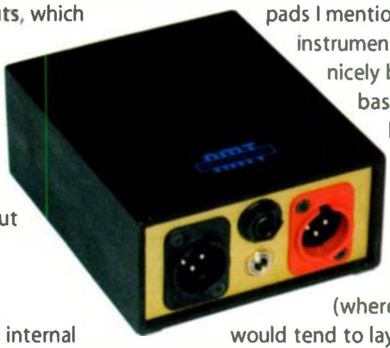
The capsule of the AMT mic, suspended in its metal shockmount.

used mainly in marine communications equipment, and is very securely clamped to the cable.

I like the fact that the plastic plug is very light, as this is the piece which, when disconnected, will hang free, so it won't inflict any damage if it bumps against the body of the instrument. The unique nature of the plug could be a potential issue, as it's not the sort of thing you would want to be repairing at a gig, but on the plus side it does look very well put together and, with care, shouldn't cause any problems. Spare plugs are available from the UK distributors, and I think I would be tempted to buy one anyway, just in case.

The preamp

The AMT preamp is available in two versions: the basic S25B and the SP25B 'pro version' reviewed here, which has an additional transformer-isolated XLR output and a line-driver module. The unit is housed in an enormously strong aluminium-sleeved box that looks as if you could stand one or more elephants on it; the innards slide out (after you undo a single screw underneath) to enable battery changes and allow access to the internal line-level preset adjustment. On one end is a single standard jack input socket and a level control, which controls the signal level from the line-driver output. On the other end are three outputs: a mono jack line-driver output for connecting to a local amplifier on stage, and the aforementioned XLR outputs, which provide a balanced PA or recording feed. On this SP25B model, the red XLR is the isolated (ground lift) output and the black one is 'normal'. This panel also accommodates an input connector for an external power supply.



The SP25B preamp.

The user guide strongly recommends the use of an internal battery rather than either an external or phantom supply, and in fact the preamp is not fully functional unless a battery is present. The isolation transformer stage and the line driver (jack output to your amp) work only on battery (or external supply) power. This is, in part, a precaution against problems which could arise if the unit is connected to a 'weak' phantom supply. The maker's literature suggests that a battery should last around 60 hours, which is quite a few gigs or sessions, and the unit only powers up if the mic plug is connected.

To gain access to the battery, you have to slide the preamp chassis from the outer sleeve by removing a single countersunk screw on the base. (It's a screwdriver job that can't be done with a fingernail or small coin.) I would feel happier if there were two screws holding the sleeve in place, because if you drop this one down the side of the stage there's nothing else holding the unit together for the rest of your gig. The 9V battery holder is the best I've ever seen, and holds the battery with a grip like a fully-grown crocodile. In fact, the overall build quality looks top class. Indeed, I assume the unit is hand-built, as it has a lovely old-fashioned over-engineered feel to it.

Bass benefits

The first advantage of this system is that because the mic is mounted on the body of the bass it can't move relative to the instrument, so it will remain in exactly the same position no matter what the player does. Secondly, the mass of the capsule assembly is so low that it won't sag on the gooseneck even if the bass is moved quite violently — you could probably make it lose position, but only by doing something extreme like dropping the bass from a few inches on to a hard floor, which I have no intention of trying on my beloved instrument!

Mounting the mic assembly couldn't be easier — having made the coarse adjustment to the front clamp, using the supplied hex wrench, you simply offer up the mount to the bass body and tighten the rear clamp using the adjustment knob. There's no damage to the body of the bass, as the clamps are lined with the suede pads I mentioned earlier, and on my instrument the whole mounting sat nicely below the shoulder of the bass, allowing the bass to be laid on its left side (where right-handed players would tend to lay it) without interfering with the mic mounting in any way. However, you can, of course, mount it anywhere on the edge of the body that works best for you. The straight cable can be run around behind the instrument using the rear clamp-knob as a cable guide, which keeps it hidden and helps you avoid treading on it. I would always do this, as it also protects the cable entry point from unnecessary flexing.

Playing around

In use, the SP25B is utterly straightforward: just plug the preamp line-driver output into your stage amp and away you go. There are no 'tone controls' on the unit (you do that bit with your amp), but experimenting with the mic position really brings home just how wide a variation in sound can be achieved by changing where the mic is placed and the distance from the body. I found that having the mic pointing directly into the f-hole from about 4cms (just far away enough to lose the major resonances) gave me a really deep, tubby sound that was ideal for small combo stuff, but for arco work (pit work, for example) it needed to be moved away another centimetre or so, to smooth out the response across the strings

I spent a while fiddling about with the SP25B and recorded different playing styles over a few hours — mainly because it's not easy to be objective about your sound while you're actually playing — and I was extremely happy with the results. What you can't do with this mic is just strap it on and expect everything to sound just like your best fantasy bass immediately. The final results will depend very much on how you use it, and to get the best out of your investment you need to follow the empirical route and try out lots of different placements and settings until it works for you. It's always a good idea to ask other band members or listeners what they think and use the poll results as a starting point, then adjust to taste as you go. However, the sound quality and capability of this system is excellent. I might even sell my bass to get one... SOS

Case Notes

Live equipment flightcasing and protection

Stage gear takes a lot more punishment than studio equipment, so investing a few quid in suitable protection for it has to be worth considering. But what do you need to know before buying?

Once you've spent hard-earned cash on equipment for live sound use, it's the generally accepted wisdom that it will need protecting against bumps and scrapes along the way. Captain Kirk could look after the *Enterprise* by simply saying 'raise shields' but for the likes of you and me, it's 'flightcase'. You might think that the cost of flightcases to properly protect your valuable gear against damage will be quite high (although cases don't actually cost as much as you might expect). However, investing in suitable cases not only offers protection against damage, but has other 'on road' benefits too, in terms of added convenience and flexibility, not to mention the professional look it gives you.

If you try a Google search on 'flightcase', about two million results will be returned, the first page or so of which are likely to refer to flightcases for the entertainment and corporate advertising industries. No matter what the object is, the chances are that someone, somewhere makes a flightcase for it, or is quite willing to do so if asked. In the field of live sound, we need cases in which to hold, operate, store and transport our equipment. The degree of protection required and the overall design and quality of the case will depend not only on the gear itself but also on the intended or potential use. When I started out I couldn't afford much in the way of decent sound gear, let alone cases to put it in, but as I've built up and improved my PA inventory I've tried to ensure that it is packed and protected as well as possible. In choosing which cases are the best for your needs, don't forget that they will add to the overall size and weight of the gear you need to move around; many full flightcases are heavier than their contents, so you may need to consider the individual weight of each item, and perhaps even the load capacity of the vehicle you use to transport your gear.

Let's have a brief look at the cases commonly used for live sound gear, and consider one or two factors along the way.

Info
Feature by Mike Crofts
Photographs by Mike Crofts

Equipment mounting cases

Generally known as 'rack cases' or 'rackmount cases', these are usually made to accommodate standard 19-inch gear, which is permanently fixed into the case and is stored, transported and operated *in situ*. Cases of this type usually have removable front and rear doors, with a centre section that contains the rack gear itself. A few different types are available: you can obtain the traditional plywood cases with aluminium edging and heavy-duty butterfly catches, or there are lighter, moulded cases that are suitable for outboard processors as well as heavier items such as amplifiers. One issue I have with moulded cases is that they often have the rackmounting strips at the front edge of the case, which means that control knobs will protrude and will be exposed when the front cover is removed. The more traditional cases usually mount the gear a little further back inside the casing.

For especially delicate gear, 'sleeved' rack cases are available. The 19-inch rack holding the equipment fits inside an outer casing, or sleeve, and between the two parts is a layer of foam to cushion against impact and vibration (ever sat on the floor in the back of a Transit van when it's negotiating a bumpy road?). Any equipment containing moving parts, such as CD and Minidisc players, or anything containing a hard drive, would undoubtedly benefit from this type of protection.

Prices for rack cases range from around £90 for a budget 4U traditional board case, through standard board and lightweight polyethylene 4U cases at between £90 and £125, to high-protection sleeved 4U cases at around £150. Bear in mind that the larger the case gets, the cheaper it will usually be per 'U'.

A metal 19-inch rack case safely housing a set of radio-mic receivers.

Keeping the gremlins out

Although rack cases are usually built with a sort of aluminium tongue-and-groove arrangement around the lid edges, this is primarily to locate the lid securely and not for environmental protection, unless the case is specifically designed to provide this. Most cases will prevent a small amount of moisture from getting inside, meaning that you can carry them (or, much better, get your roadie to carry them) through the rain, but if you want to keep the elements out completely,

This rack case lid incorporates a weatherproof sealing strip.

the case will have to have a proper environmental rating, such as IP44 — as applied to outdoor mains connectors, for example. Such specialised cases — designed mainly for transporting items like laboratory instruments — tend to be expensive and are not always very robust, but are worth considering for very sensitive gear. I use such cases for my recording equipment, which tends to be carried in cars rather than vans and is only handled by me!

Rack cases are available in different depths (front-to-back), and experience has taught me to use the correct depth if at all possible: too shallow and your equipment pokes out from the back (and is therefore at risk and looks naff); too deep and you'll be forever shining a torch inside and struggling to plug things into the proper holes. A good rack depth will comfortably accommodate the equipment, allow good access to the rear-panel connections and provide enough space to store or permanently install things like power cables. Having said that, if you do keep cables inside the rack (which saves time during setup), make sure that the metal plugs can't roll around inside and damage anything. A power distribution panel is a very useful thing in an effects rack, as it provides a neat and safe solution. Most of these can be mounted on the rear rack strips (if you have them), which means that you're not sacrificing an 'operational' slot in your rack.

One final thought: it's a good idea to label the outside of the case to identify the front and the top, so that it can be transported the right way up, and placed *in situ* the right way around. Saves time every time!

Pack your trunk

Trunk cases can save loads of time and leg-work when you're loading, unloading and setting up. Road trunks come in all shapes and sizes, can be used to transport and protect virtually anything, and cost as little as £150 or so new. The most obvious uses are for cables, microphone stands and the like — in fact, anything which otherwise would have to be carried in small numbers and in lots of trips between van and venue. It's great to roll the trunk right up to the stage area and simply

A cable trunk. It's amazing what you can fit in when you try...

pull out all the cables you need, in the correct order and neatly coiled; I reckon this is the biggest time-saver of all during setup. When packing up at the end of the night you can again save time by just throwing everything in, provided you remember to sort it all out before the next gig.

Equipment transport cases

For live sound equipment, rack cases and road trunks will cover most of the basic requirement, but some kit will require special attention. Up-market backline will often need specially-sized cases for life on the road. For example, a guitar 4x12 could be transported in a bespoke case where most of the height consists of a very deep lift-off lid, so that the cabinet can be left on the shallow base (and the castors) during the gig, if need be. For main PA speakers, very large flight cases would be needed, so unless you're on the road all the time, touring far afield or engaged in the hire business, you can generally get away with padded bags (available for as little as £80 a pair for popular compact PA speakers such as the Mackie SRM450) and a bit of careful packing and handling. Many items of professional audio-visual display equipment are housed in lightweight transit cases, which offer convenience of mobility and enough protection and look like 'real' flight cases, but are not designed to withstand roadie rage. Beware of re-using these cases for heavy items such as amplifiers, because the side panels may not be strong enough to withstand much of an impact.

Speaking of using and re-using, there are a lot of second-hand cases available, some at very good prices. It's worth taking a close look at older ones before purchasing, because it can be very frustrating and time-consuming to repair or replace things like aluminium edging strip, distorted hinges or seized-up butterfly catches. I must admit to having two fairly large and currently unusable cases in my gear graveyard because I just haven't got the »

Trunk call

However useful road trunks are, there's a compromise to be made when deciding what goes into them and therefore what size you need. It's very convenient and very fast to have all manner of bits and pieces in one or two large trunks — just wheel 'em in and away you go — but consider the weight and size of large cases, and the difficulty of handling them. It doesn't take many cables or mic stands to make a road trunk into a heavy and unwieldy object, and you may then need a second person to help get them in and out of the vehicle or into the venue. One way around this is to use what I think of as the Russian doll approach: various bits of gear (for example, microphones, adapters, small signal cables, and so on) can be kept in small cases, and then several of these cases can be transported inside one larger trunk, depending on how much you need to take to the gig, and how many helpers you have. This gives the best of both worlds and also provides two layers of protection. A fully-loaded road trunk can be a difficult beast to control, especially if all four wheels are swivelling castors. I've lost count of the times a slight sideways gradient has given the trunk a mind of its own (the 'Shopping Trolley Effect') and then there's Postlethwaite's Theorem, which states that if one end of a laden castor-equipped trunk is lifted by a person, the opposite end will tend to describe an arc which terminates against an adjacent vehicle.

Mic stands are a real pain to carry when you've got more than two in each hand, and you can get neat little road trunks specially for them. Do watch out when emptying these long, thin cases though. Due to their tall, narrow shape and the weight of the lid acting on one side when open, some of them can be prone to tipping over as you take the last stand out. If possible, stand them up against a wall so that they can't do anyone any harm.

This trunk holds 15 mic stands and also makes a useful trolley.

This flightcase was designed to hold audio-visual display equipment, not PA gear. The damage has been caused by a heavy object moving around inside.

» time to repair them properly, and they're a waste of space and money if they're not working for a living!

How it's all done

On my last trip to Flightcase Warehouse in Tamworth, I took my camera along and had a chat with Jason Furneaux, FW's general manager, and the company's owner and director, Steve Austin, about how the cases are made. Jason talked and then walked me through the whole process of turning raw materials and boxes of fittings into finished flightcases. They're all made from birch plywood, either 7mm or 9mm thick, which is supplied in large sheets and has a coloured (usually black) phenolic surface layer ready-bonded on both sides (they call it 'Hexaboard', but I'm not sure if this is a trade name or a generic name).

The case edging is aluminium extrusion in 7mm and 9mm sizes, depending on the board being used, and the rest of the 'raw' stock consists mainly of fittings (ball corners, castors, handles, hinges and butterfly catches) and the foam used to line the cases, which is cut to exactly fit the equipment going into the case. One case size, for example, can — if 'foamed' to suit — accommodate several different but similar items of equipment.

Flightcase Warehouse make around 120-150 cases every month, nearly all based on specific customer orders (more than half via their web site), so the manufacturing process has had to be streamlined and automated as far as possible, to keep prices down and production up. The various pieces of machinery in the workshop areas are set up to produce whatever model of flightcase is required, and a production run can be for a single case or as many as required to meet a customer's needs. All the specifications are maintained on a specialist piece of design software, which means that the tooling-up and identification of the component parts needed to build a particular size and shape of case is quick and straightforward. An order can literally be in production within minutes of being received.

A cutting machine used on the boards that form the panels of the flightcases.

Case contacts

▪ Flightcase Warehouse
T +44 (0)1827 60009.
W www.flightcasewarehouse.co.uk

▪ R&J Flytes
T +44 (0)1536 723451.
W www.rjflytes.co.uk

▪ The Noizeworks
T 0870 240 3119.
T www.thenoizeworks.co.uk

Being creative

Because of the relatively simple construction of flightcases, they are quite easy to adapt; if you find one with a lift-off lid and shallow base, it can be converted into a cable trunk by turning it upside-down and putting the wheels on what was originally the lid.

You will need to turn any flip-up handles the other way round too, because they are only designed to take the load in one direction, and if you use them the wrong way around they can trap your fingers against the case. If not already fitted, wheel brakes are a good idea too, especially for heavy cases that have to remain upright in transit; having at least one locking castor should prevent too much moving around or the possibility of the case rolling off the tail lift.

This second-hand case was 're-foamed' to fit a Spirit FX16 mixer, and there's plenty of room for the power supply too.

Not all the cases are for the music industry: clients include motor-racing teams, specialist equipment manufacturers (for example, drinks vending machines) and promotional display companies.

Making a flightcase

When an order is received, a job sheet will be raised detailing the model and specification of the case required. The panels and aluminium extrusion are cut to the correct size and the necessary holes and recesses are routed and cut to accommodate the fittings at a later stage.

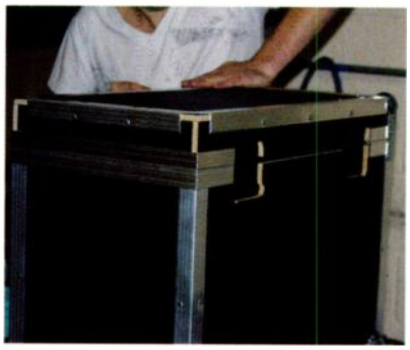
The individual panels are then riveted together to form the basic shell of the flight case. This has been made into a much more efficient process by the introduction of specialised machines, such as the one I'd seen being delivered that very day. Jason explained that it was a 'long-arm riveter', which can punch rivets straight through aluminium extrusion and side panels without the need for pre-drilling. The 'long-arm' part means that the riveter can reach across larger panels and fasten the case together without stopping to re-position the work.

Once the case has been assembled, complete with all the correct openings and recesses, the 'hybrid' sections are attached. These are the aluminium 'mating' edges that

Stencil case!

Don't forget the potential advertising value of your flightcases. They're often quite big, and they'll often be the first thing anyone sees when you roll up to a new venue, so a logo or name stencilled on the outside can give a good impression from the start. Stencilling your cases also adds a degree of security, and it can save time at a gig if the cases are labelled with their contents. However, I tend to use coded language for this, because you don't necessarily want casual observers watching you putting a box labelled 'little, very expensive microphones' into your vehicle. Something like '12 vox h/held' does it for me.

fit around any openings, such as the edges of lids and doors, and must be properly aligned to maintain the case's structural integrity. After this, various other pieces of hardware, such as the metal ball corners, hinges, catches and handles can be added. This job must be done by hand, but because the routing and cutting is done according to the case spec in advance, before assembly, all the openings are exactly the right size for the fittings being used. When the case is complete with fittings, the castors — if required — are attached, either direct to the case, or mounted on an 18mm plywood wheel-board for larger versions.

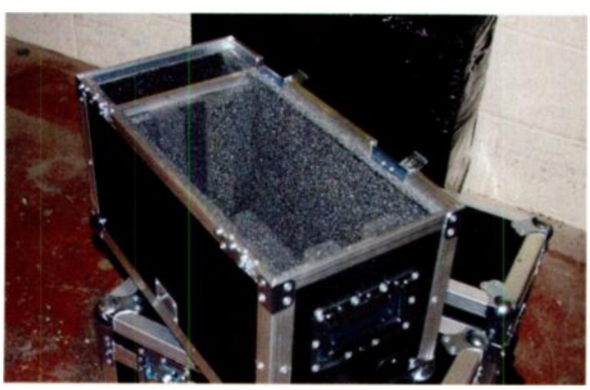

A case awaiting fittings...

Foam if you want to

The final stage is to cut and fix the internal foam lining. Once again, the exact dimensions are taken from the design database and the foam (firm packaging foam called Jiffy foam) is cut to size with an electric knife, then fixed in place with a compressed-air spray-glue gun. This looks like brilliant fun, and you get to wear a cool mask!

The finished cases are checked over and sent to the despatch area for shrink-wrapping or bubble-wrapping, before the firm's two courier services come to collect them every afternoon.

Before leaving, I asked the owner, Steve Austin, whether the company had ever been asked to make anything out of the ordinary — and apparently they have. A gentleman once ordered a flightcase made to fit himself, to be used as his coffin, wheels and all. Now that really is rock and roll! SOS


Internal foam having been glued into place, this case is almost ready to go.

Fohhn Xperience II Powered PA system

This premium German-made PA aims to offer a genuine hi-fi approach to live sound reproduction. But is it worth the investment?

Ever since I reviewed a medium-sized Fohhn PA system, testing it out on a fairly large outdoor event (see the November 2004 issue of *SOS Live*), I've been a big fan of the company's approach to sound reproduction. While Fohhn PAs are by no means the cheapest systems around, they manage to sound more 'hi-fi' than most other PAs, yet they're also relatively compact and light. The Xperience II system represents the smaller end of the Fohhn range but it still packs enough power for smaller band gigs, gigs using full-range backing tracks and discerning acoustic acts. One of my tests was at a fairly loud dance-music gig in a marquee, and although I eventually hit the limiters the level and quality of sound really surprised me. Even more impressive is the fact that the entire system will pack into the back of most small hatchbacks, even without the rear seats folded down.

The basic Xperience II system comprises two passive TOP XT2 speakers and one SUB XS2 powered subwoofer. All the crossovers are passive networks, both within the top cabinets and between the sub and the tops, and the crossover in the top cabinets includes a three-stage tweeter-protection system. A pair of Class-D 500W power amplifiers is built into the sub and these power everything, with DSP (Digital Signal Processing) technology handling EQ and limiting. It's also possible to add two further top speakers and one more passive sub without needing to bring in more amplifier power but, unusually, the system can utilise the full 500W per channel even when working with only one sub and two top cabinets. How the Fohhn designers have achieved this is something of a technical secret, as is the means by which they feed a single voice-coil sub with a mono signal derived from the left and right channels using passive crossovers. This powered passive approach is a departure from the active speaker norm, where every driver has its own amplifier and active crossover, but it certainly works.

A closer look

The SUB XS2 is really the heart of the system but it is surprisingly compact, being based around a single 12-inch, Neodymium, long-throw driver mounted in a vented cabinet built from high-quality birch ply with sensibly placed carrying handles. Measuring just 43 x 39 x 44cm, the sub weighs a very manageable 19kg, partly due to the use of modern Neodymium technology and partly to that of Class-D amplifiers. During the trials, I carried it a considerable distance in one hand with one of the tops in the other, something I definitely couldn't do with my own Mackie

At a glance

Fohhn Xperience II

Pros

- Classy, almost hi-fi sound quality.
- Very portable.
- Impeccably engineered.

Cons

- No input gain control.
- Accessories such as covers, cables and stands are optional.

Summary

If you just want to generate enough SPL, there are far cheaper ways of doing it than with an Xperience II PA, but as a system capable of delivering near studio-quality reproduction in a compact and lightweight package, I've yet to try anything that comes close to it.

Information

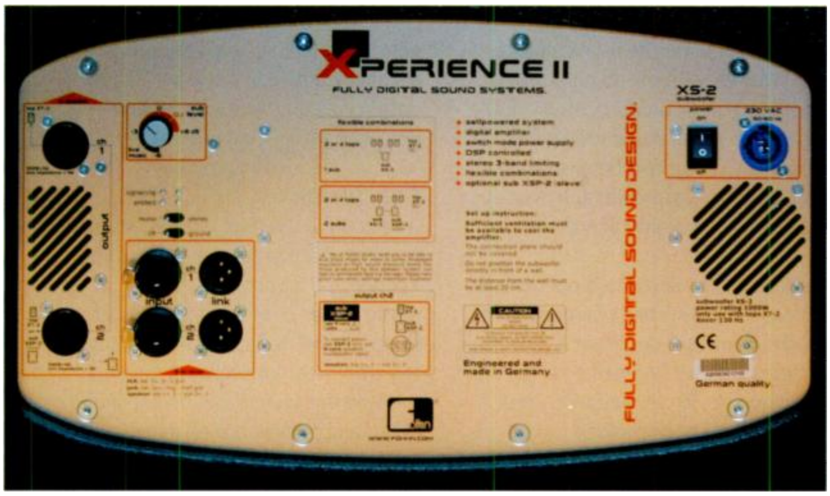
E Xperience II System £2846; Xperience II Power System (two subs and two tops) £3423. Prices include VAT.
T Systems Workshop +44 (0)1691 658550.
W www.systemsworkshop.com
W www.fohhn.com

Digital Signal Processing

Multi-band DSP is built into the sub, to maintain optimum tonal balance at all listening levels and also to provide anti-clip limiting. I asked for more information on this aspect of the system and was told that the DSP provides three-band limiting to cover the frequency ranges of the bass speaker, the mid driver and the HF driver, and ensure that each speaker receives only as much power as it can handle safely. The DSP, which also includes equalisation to optimise the system sound, is claimed by the manufacturers to be the technical equal of the big-name external processors using in large touring rigs. It modifies the system's frequency-response curve depending on the level of the PA, so the tonal balance remains correct whether at a loud gig or a wine bar. As far as I can see, this exploits the way in which the human hearing system responds to different sound levels: boosting the highs and lows at lower listening levels helps maintain a sense of loudness and power. As the system level is increased, this EQ function flattens out to become linear.

system.

All the connections are on the rear panel: Speakons for the feeds to the other speakers and XLR/jack combi sockets for the stereo line inputs. A Powercon cable is used for the mains feed, which has the advantage of being locking, but if you forget to pack it you're less likely to be able to drum up a spare at short notice than with an IEC cable. The fact that the connector is a locking normally the system is run in stereo, but if you were using two systems you could run both in mono mode and have one for each channel. A threaded pole-mount is fitted into the top of the sub and a mounting pole is included, to support one of the top speakers. The other speaker mounts on a conventional stand, unless you add a second passive sub, in which case the second speaker can be pole-mounted above that.



The subwoofer is the business end of the Xperience II system, containing a pair of 500W amplifiers and the DSP for EQ and limiting. The rear panel of the sub features all inputs and outputs, Powercon mains power inlet and power switch, ground-lift switch, and LEDs to show signal presence/clipping and whether speaker protection has kicked in.

one is a double-edged sword, too: the cable won't get dislodged during a performance, but if someone trips over the cable, the plug can't come out, meaning that the cable itself might get damaged, or even pulled out of the plug. Because Class-D amplifiers are so efficient (around 95 percent), the rear panel serves as an adequate heatsink, and even the amps run surprisingly cool, at a maximum of only 14 degrees centigrade above the ambient room temperature.

A single rotary control sets the relative subwoofer level, with the centre position being nominally flat. A slide switch moves from mono to stereo operation:

Although they're deceptively compact, at 32 x 51 x 29 cms, the 8Ω TOP XT2s have a power rating of 300W each, with a short-term peak-handling capacity of 600W. These are two-way speakers comprising a 10-inch woofer teamed with a one-inch, constant-directivity, horn-loaded HF driver. The two drivers are time-aligned for phase accuracy and are very efficient, being quoted at 99dB (1 Watt at 1 metre). They can generate a maximum SPL of 127dB, with a coverage angle of 90 degrees horizontally and 40 degrees vertically, and since their frequency range is 65Hz to 20kHz they could work as a part of a vocal system without the sub (whose frequency response extends from 38 to 130Hz), providing they were teamed with a suitable power amplifier. They also have an angled enclosure (55 degrees), »

Test info Reviewed by Paul White

Fohhn Xperience II

Powered PA system

» allowing them to be used as stage monitors, and may also be used as side-fill monitors. Again, they're made of plywood, finished with a very durable textured black paint, and are virtually resonance free. Weighing just 15kg each, they're built like tanks, with a metal protective speaker grille topped with a thin layer of acoustically transparent foam. Carrying handles are built in, as are fly mounts, and there's a standard pole-mount socket in the base.

Each TOP XT2 has two Speakon connectors on its recessed connector panel, to facilitate daisy-chaining to a second cabinet where required. Optional protective covers are available for both the tops and the sub.

In use

I found the Xperience II system very portable and quick to set up. It also has a soft-start, switch-mode power supply, so that there's no switch-on thump. As soon as the power has come up to the required level, green LEDs on each channel (on the rear of the subwoofer) show that the system is ready to use. These are dual-colour LEDs, which also flash red when the limiter is active. If anything happens that puts the amplifiers into self-protect mode, such as a speaker-cable short circuit, red LEDs show on the affected channels. However, there are no visible limit LEDs on the front of the system, where the front-of-house engineer could see them. For situations where ground-loop hum is a problem, there's also a ground-lift switch, although I found no need to use this, as everything was perfectly quiet.

The maximum level achievable by this system is just a few dBs short of what I get from my Mackie system. When you consider how much smaller the Fohhn system is, that's pretty impressive. It doesn't have quite the same depth of bass, but it still has enough to deal with even quite punchy material in small to medium-sized venues. Part of my test was running full-range backing tracks through the system, and the only time I've ever heard these sound better was when I was mixing them on studio monitors! Most PA systems do a pretty perfunctory job on recorded music, with the mid-range often being particularly weak, but the Xperience II system behaves more like a loud hi-fi system, providing a clear and crisp mid-range with plenty of transient detail.

The real test, at the local pub open-stage night, confirmed the system's credentials. It was brought in and set up in just a few minutes, after which it handled a number of acts, including solo singer/guitarists, acts with backing tracks and recorded background music. Not only did it sound clean and articulate, it also showed better-than-average resistance to acoustic feedback. At very low levels, the sound is brighter and tighter than you'd normally expect, no doubt due to the on-board DSP equalisation, but it doesn't sound unnaturally hyped, and the effect isn't so strong that it has you reaching for your mixer EQ. On the contrary, I found that most signals sounded best mixed flat, including the vocal mics. In fact, the only shortcoming I found was that because the system has no input-level control, you sometimes have to drive your mixer fairly hard to get enough level in. I remember this being an issue with the larger system I tested last year, so it's important to use a mixer that has +4dBu outputs, as the consumer -10dBu level may not drive the system adequately hard.

Jargon explained

Class-D amplifier: A Class-D amplifier is one in which the output transistors are operated, effectively, as switches, being either fully off or fully on, with the audio signal modulating (controlling) the switching action. A Class-D amplifier design is very efficient, allowing smaller power supplies to be used. The 'D' is sometimes mistakenly said to stand for 'digital' when in fact it was simply the next letter in the alphabetical series after Class A, Class B (and A/B) and Class C.

Crossover: A crossover is an electronic circuit designed to separate high- and low-frequency signals from each other so that each can be fed to speakers that are properly optimised for the role — ie. large, robust speakers for bass, and small and light (and therefore fast) speakers for high frequencies. If a crossover is placed between the power amps and the speakers (usually built into the speakers), it is said to be 'passive'. An 'active' crossover is used to divide signals before the power amps, so separate amps are then used for each band, further adding to efficiency. Crossovers can be two-way, simply splitting highs and lows; three-way, adding a mid band; and ocasionally four-way.

Neodymium: Very high strength 'rare-earth' magnet type made out of neodymium, iron and boron. Its use in a loudspeaker allows the coil attached to the diaphragm to be made smaller and therefore lighter, thereby increasing the speaker's efficiency.

SPL: Sound level calculated in decibels compared to a reference sound pressure, commonly 20 micropascal (20 uPa), defined as the threshold of human hearing (0dB SPL). The human ear is capable of hearing an enormous range of sound pressures — the ratio of the sound pressure from the minimum up to damaging levels from even short-term exposure is more than a million — necessitating a logarithmic scale, which also corresponds roughly to our psychological perception of loudness.

Time-aligned: When any part of an audio signal is reproduced by more than one drive unit in a cabinet, any difference in the time taken for the audio to reach the listener from each source, due to the different positions of the drivers, will inevitably degrade the sound. Designers seek to nullify this effect either by physical alignment, such as sloped or stepped baffles, or through electronically delaying the signal to the driver with the shorter path length to the listener's ears.

Conclusion

As a compact system capable of producing hi-fi like audio in smaller venues, the Xperience II is very hard to fault. Acoustic music comes over with pristine clarity and vocals sound as natural as you could hope for, with great intelligibility. I'm impressed at the way the system can be expanded using additional passive boxes: although this particular system is probably too small for bands who want to mic their drums and DI their bass (except in small venues), it can be expanded by adding more tops and/or another passive sub, to add up to another 6dB to the overall level. And if you need more volume or more depth of bass, Fohhn produce a range of sound systems, of which this is probably the smallest.

The Xperience II system is also a joy to transport and to set up, but all these positive attributes come at a price. Because of its not inconsiderable cost, my guess is that the Xperience II would be best suited to club and pub performers who are working regularly and need to combine the best possible sound quality with ease of transportation. It's also ideal for acoustic bands who need a bit more up-front level, or acts using good-quality backing tracks or sequences. Similarly, electronic acts working smaller venues will find that the Xperience II can recreate the sound achieved in their studio recordings very accurately, and with enough low end to do justice to almost any type of material. SOS

As you'd expect, an angled shape allows the top cabinets to be used as stage monitors.

classifieds

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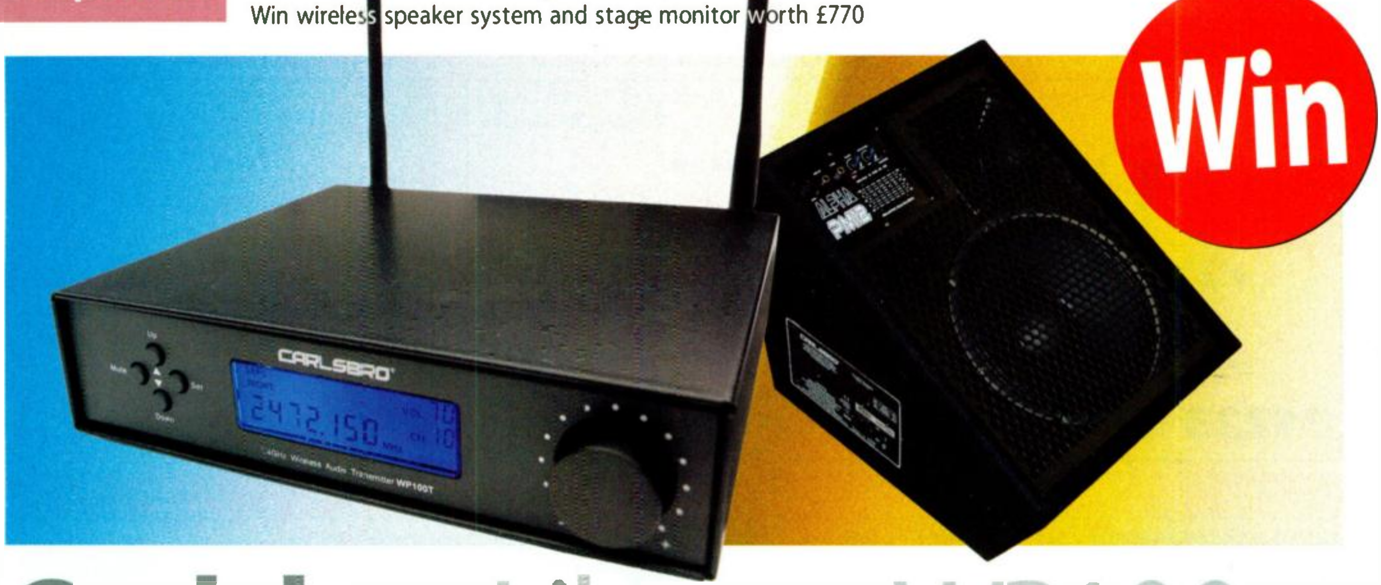

Carlsbro Liberty WP100

Mention the word 'wireless' in the context of PA and everyone immediately thinks of wireless microphones. Wireless, or 'radio', mics are understandably popular, providing good audio quality these days and dramatically reducing on-stage cable clutter, as well as giving performers more freedom of movement. But even if all your input sources are wireless, you still need lengthy cabling to return the mixer's output to your power amps and speakers, unless... Yes, you've guessed it. Why shouldn't the mixer output be wireless too?

Carlsbro's Liberty Wireless WP100 system is designed to take the stereo output from your mixer and transmit it to two or more individual receivers placed by your power amps or self-powered speakers. While some microphone systems use compressed audio to overcome radio bandwidth limitations, the Liberty has a 'mission-critical' application, with the entire audio chain passing through it, and clearly needs to be engineered to a high standard. The system uses a high-quality, uncompressed audio signal, transmitting in the licence-free 2.4GHz band (2400-2482 MHz). The receivers feature a dual-diversity twin-antenna system, which will automatically switch to follow the strongest signal path available, making the radio link as robust as possible — the last thing you want is drop-outs or glitching in a unit that is connected directly to your power amps and can't be reached from your mixing position!

The system's receivers have an automatic frequency-scan function that enables them to find and lock on to the transmitted signal. The transmitter offers balanced XLR and jack inputs and features an intuitive user interface with a large, backlit LCD showing which of 50 user-selectable channels is in operation, and also displaying signal strength. An output level control allows further adjustment of drive level between mixer and power amps.

In a music context, wireless mics and a wireless return signal mean goodbye to heavy, expensive multicores, but there are other applications too: think about theatre audio — hiding a small speaker on stage for a spot effect is often much easier than hiding the cable that feeds it — or sports events, where the same signal often has to be routed to far-flung locations.

Carlsbro's Liberty Wireless WP100 system is a truly innovative application of modern wireless technology that brings the convenience of totally wireless audio distribution within reach of the gigging musician, and in this issue of *SOS Live* Carlsbro are giving readers the chance to win one, worth £499. They're also offering a runner-up prize of an Alpha PM12 powered monitor wedge worth £269. The PM12 is a fully featured foldback monitor with mic and line inputs, input gain and volume controls and a six-band graphic equaliser to deal with any feedback problems. This extremely useful bit of kit offers no less than 200W of power and even has an external speaker output for driving a second monitor.

If you'd like a chance of winning one of Carlsbro's great prizes, fill out the entry form and post it to the address on the coupon. Please make sure you answer all questions and complete the tie-breaker. We also need your full address, including postcode, and **daytime** telephone number. The closing date for entries is 30th April 2006. Good luck!

Info

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The small print

1. Only one entry per person is permitted. 2. Employees of SOS Publications Ltd, Carlsbro Ltd and their immediate families are ineligible for entry. 3. No cash alternative is available in lieu of the stated prize. 4. The competition organisers reserve the right to change the specification of the prize offered. 5. The judges' decision is final and legally binding, and no correspondence will be entered into. 6. No other correspondence is to be included with competition entries. 7. Please ensure that you give your DAYTIME telephone number on your entry form. 8. Prize winners must be prepared to make themselves available in the event that the competition organisers wish to make a personal presentation.

Questions

How many channels does the WP100 have?

- a: 12 ☐
- b: 16 ☐
- c: 24 ☐
- d: 50 ☐

How much is the WP100 worth?

- a: £4.99 ☐
- b: £49.99 ☐
- c: £499 ☐
- d: £4999 ☐

Which of these is not a feature of the PM12?

- a: Input gain control ☐
- b: Output volume control ☐
- c: Six-band EQ ☐
- d: Undo ☐

Carlsbro Liberty System Tie-breaker

If you were 'at liberty' to do whatever you wanted with a wireless speaker system, what would it be and why? Answers in 30 words or fewer, please.

....

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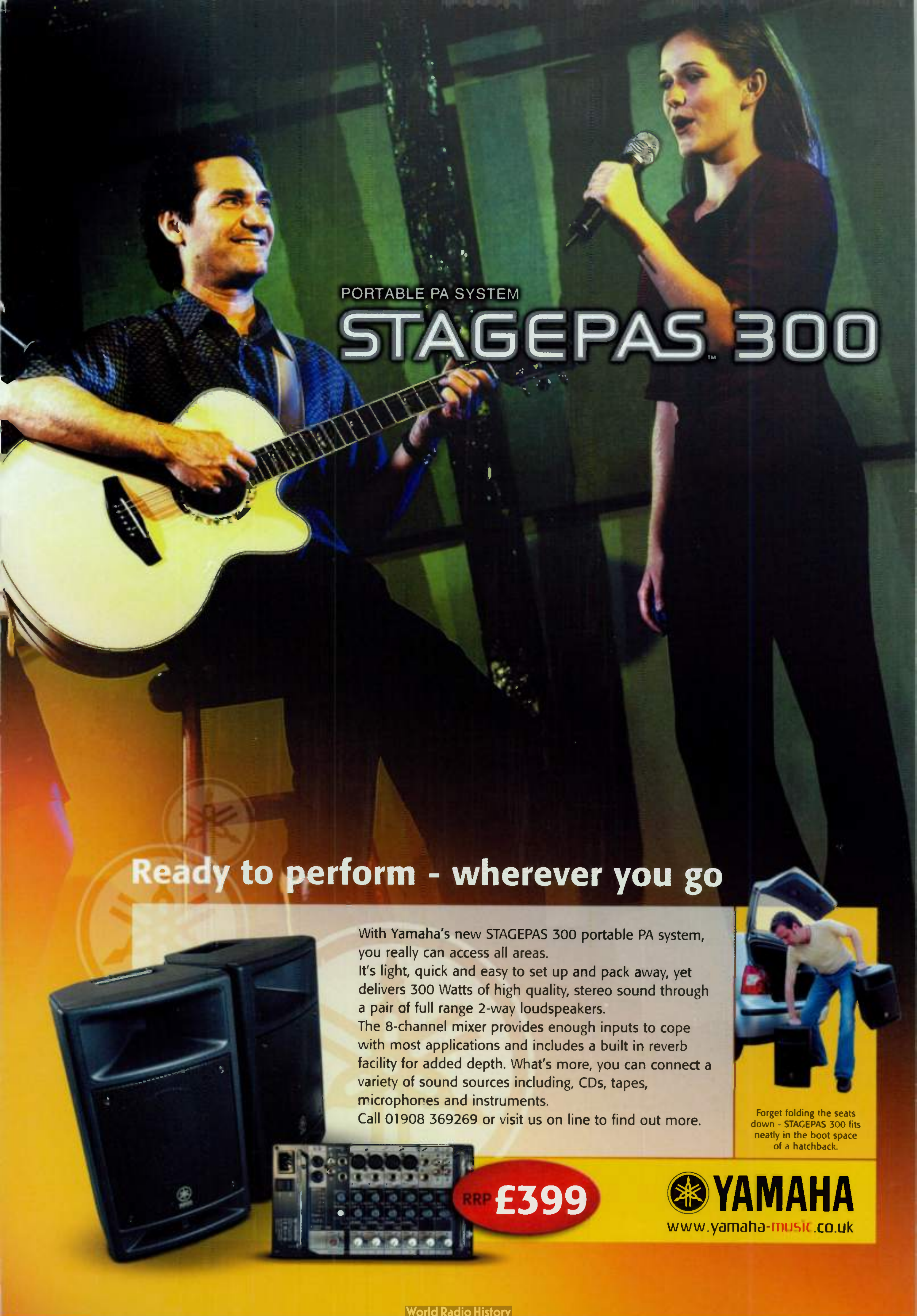
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