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SOUND ON SOUND™

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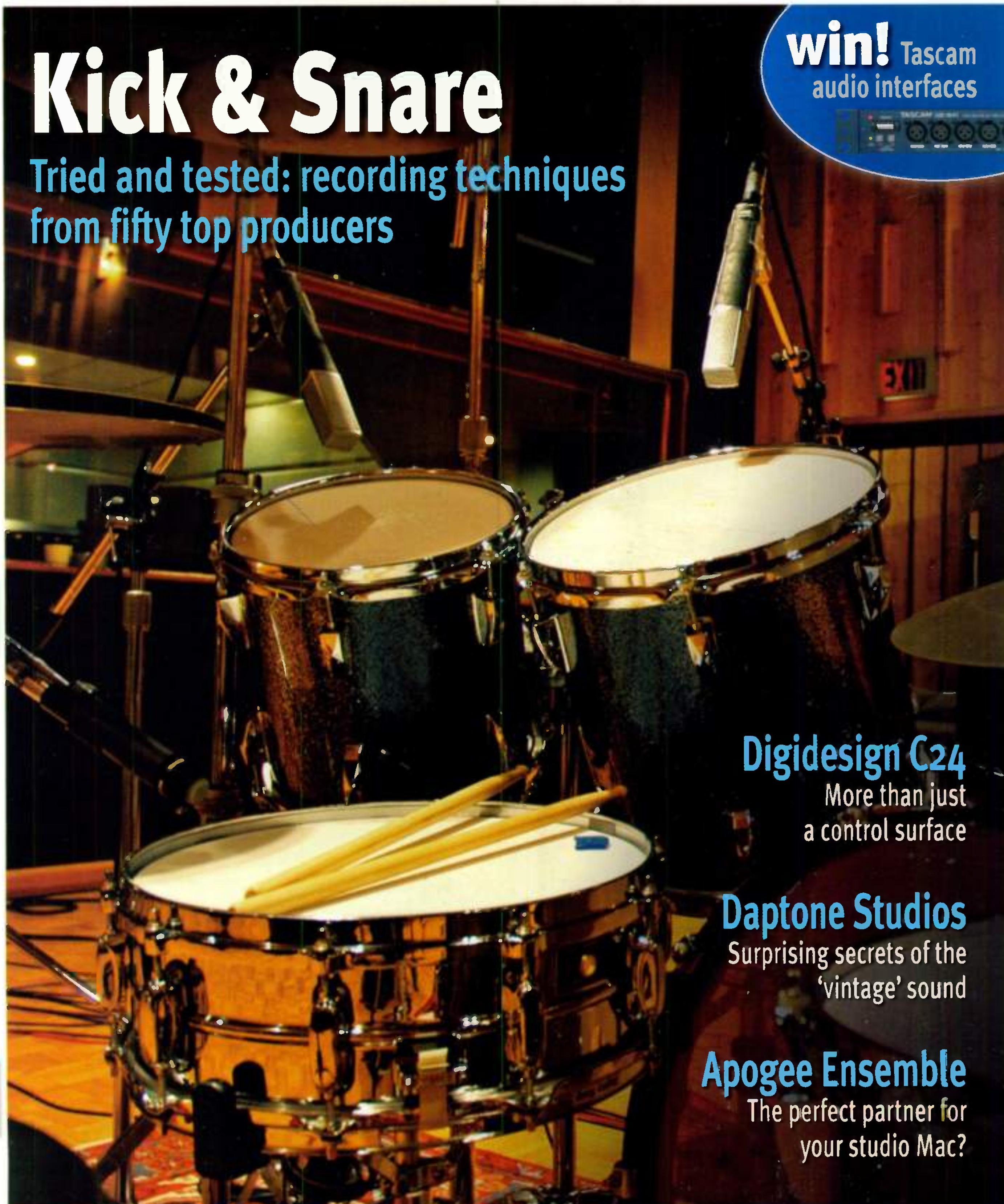
June 2008 £4.99



win! Tascam audio interfaces

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Tried and tested: recording techniques from fifty top producers

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Apogee Ensemble
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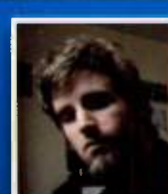
Mike Chapman on Blondie
The legendary producer behind *Parallel Lines*

Is it a summing amp? Is it a mixer?
Is TL Audio's Fat Track for you?

Pendulum
Drum & bass goes global



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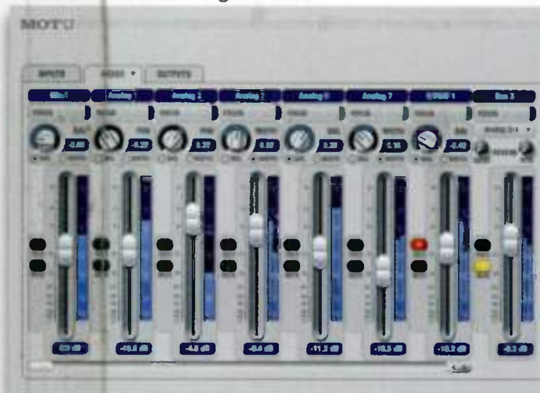

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Question Everything

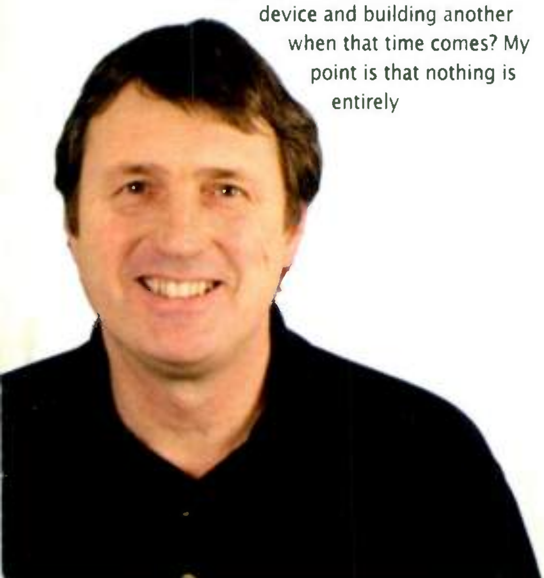
A lot of what we're expected to take as being 'obvious' or accepted wisdom isn't always quite so clear-cut when you think about it more deeply. Take energy-saving light bulbs, for example. There's no doubt that they use less energy than a conventional bulb, but in the nine months of the year when most British households need to run their central-heating systems to keep our bracing climate at bay, any 'wasted' energy from light bulbs and gadgets left on standby simply contributes to the overall domestic heating. Assuming you have a room thermostat, this means that you burn a little less gas to compensate. Energy-saving bulbs are a good idea (or they will be when they generate anywhere as near as much light as they claim to), but they don't save as much energy as you may have been led to believe.

What has this to do with recording? Well, we're always being told that switching off computers and other equipment (as opposed to leaving it in standby or sleep mode) when it's not in use will reduce our energy consumption, but on the other hand it can also shorten the life of many non-mechanical components. It's the thermal stress of starting up from cold that causes the most damage to integrated circuits and transistors. And how much energy is spent disposing of one old device and building another when that time comes? My point is that nothing is entirely straightforward, and the same is true of accepted wisdom when it comes to the recording process.

For example, when mastering, most of us assume that we always have to apply compression, limiting and EQ to make things sound 'better'. But why? There are times when it's a good idea, but it shouldn't be regarded as compulsory — a good mix can sound great just as it is. In these days of digital downloads, we can't even use the excuse of applying processing to make a tune sit nicely alongside other tracks on the album, because the end user might well have his or her iPod set to Shuffle! And why must everything be processed to sound louder at the expense of audio quality? Surely a species that's evolved enough to record and appreciate music should be able to use a volume control?

The same goes for choice of equipment. There are certain things we aspire to own because we are told that they are industry standards, the inference being that we can't make hit records without them. But, in reality, we need to identify the weak link in the chain — whether it be knowledge, aptitude, musical ability, room acoustics or choice of equipment — and then aim to redress the balance by the most appropriate means. We're always being told to 'think outside the box' but maybe now the time has come to recycle the box and just think!

Paul White *Editor In Chief*



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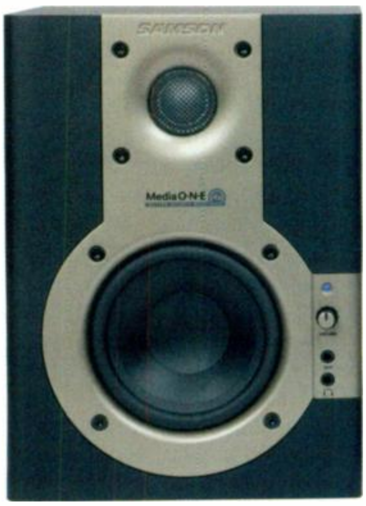

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The Sound Recording Technology Show

Don't miss it!

S*ound On Sound* are sponsoring Sound Recording Technology (SRT), the best recording show the UK has ever seen. On 12th June, four action-packed days of music technology madness will commence at the Excel exhibition centre in London's Docklands! The Sound Recording Technology show, part of the London International Music show, is the only place you can see all the leading recording and audio technology products from all the major manufacturers under one roof. Experience the demos and get hands-on with all the new gear for 2008 for the first time in the UK. Whether you're a tech-savvy laptop muso or you have your own project studio setup, SRT will have something for you.

Sound On Sound will be running a full programme of seminars and panel discussions on the Friday, Saturday and Sunday of the show. *SOS*'s Editor In Chief Paul White and Technical Editor Hugh Robjohns will be on hand to answer questions, along with many more of *Sound On Sound*'s editors and contributors (check out the timetable on the opposite page to see when you can join in with our Q&A sessions).

What's more, your SRT ticket is also valid for the London Guitar Show, Drummer Live and the new Unplugged acoustic instrument show, which are all running concurrently! Guest stars at the shows will include UK jazz legend Courtney Pine, bassist Billy Sheehan and guitarists Joe Satriani, Paul Gilbert, Richie Kotzen, Guthrie Govan and Jamie Humphries. Also, the RD Crusaders, starring Roger Daltrey and Lulu, will perform during the show. What more could you want?

www.londoninternationalmusicshow.com

SRT 2008
the sound recording technology show
In association with
SOUND ON SOUND

London International Music Show 08

DRUMMER LIVE 08

unplugged 08

LONDON GUITAR SHOW 2008

Also At LIMS

Running alongside SRT are three other shows: Unplugged, the London Guitar Show and Drummer Live. Your SRT tickets will give you access to all three shows, which are all taking place under one roof at the fabulous Excel Exhibition Centre.

Drummer Live

Now in its fifth year, Drummer Live is the place to be if you want to see and hear the hottest stuff in drums and percussion, and meet some of the drum world's biggest names. This year, visitors will get the chance to see drummers including Chad Smith and Jason Bonham play, and sit in on seminars delivered by Mike Dolbear and his team (mikedolbear.com). Just remember to bring your ear plugs!

Unplugged

Your chance to see brass, woodwind, pianos and acoustic guitars from the world's leading manufacturers and distributors, and sift through the huge array of printed music on show from the best music publishers. The place to be if you need some peace and quiet!

The London Guitar Show

The biggest names in guitar manufacture and retail will join together for the largest public guitar show in Europe. Guests this year will include none other than Joe Satriani, who will be playing a live gig during the show (additional tickets required).

Sound On Sound's Dream Studio Giveaway

Entry to SRT also allows you to take part in the massive Dream Studio giveaway prize draw, where a huge home studio bundle based on the Apple 'Dream System' is up for grabs. The system is based around an eight-core Mac Pro running Logic Studio with Apogee interfaces, while additional prizes from manufacturers including Roland, Edirol, Shure, SSL, Allen & Heath, Rycote, TL Audio, Studiospares (and many, many more) will complete the prize line-up.

In addition, the winner will have their studio installed by the Studio SOS team, and will receive Logic training from Alchemea.

A full list of equipment can be found on pages 154-155 of this issue, with regular news and updates posted on the *SOS* web site. Remember that you can only enter the prize draw by attending the show; you can't do it on-line or by post. So be there!

Sound On Sound Seminars at SRT

Time	Event	Description
Friday 13th June		
10.00am	Show opens	
11.00am	Apple Presents: Compose With Logic Studio	In addition to checking out the new features in Logic Pro 8, you'll see how other tools on on the Mac can benefit musicians and producers.
12.00pm	Creative Sound Design In Pro Tools	*SOS* contributor Jem Godfrey gets into do-it-yourself sampling and sound design, and discovers how to create new sonic universes with a can of beer and a toy Dalek!
1.00pm	Producer Panel	Today's hit-makers talk tech (and other things) in our open-topic discussion forum.
2.00pm	Home Studio Recording techniques	*SOS* Editor In Chief Paul White and Technical Editor Hugh Robjohns present a home-recording workshop, with a Q&A session to wrap things up.
3.45pm	Operating & Optimising PA Systems	*Performing Musician*'s PA correspondent Mike Crofts guides you through the ins and outs of PA setup, with live band demos courtesy of the *SOS* editorial team.
5.00pm	Show closes	
Saturday 14th June		
10.00am	Show opens	
11.00am	Apple Presents: Compose With Logic Studio	In addition to checking out the new features in Logic Pro 8, you'll see how other tools on on the Mac can benefit musicians and producers.
12.00pm	The MPG Presents: The Loudness War	The Music Producers Guild (MPG) bring together recording and mastering engineers to discuss the topic of loudness in modern music. No doubt the discussion will spill over into the Producer Panel that follows.
1.00pm	Producer Panel	The loudness discussion rages on, with the questions taken from the audience.
2.00pm	Home Studio Recording Techniques	*SOS* Editor In Chief Paul White and Technical Editor Hugh Robjohns present a home-recording workshop, with a Q&A session to wrap things up.
3.45pm	Operating & Optimising PA Systems	*Performing Musician*'s PA correspondent Mike Crofts guides you through the ins and outs of PA setup, with live band demos courtesy of the *SOS* editorial team.
6.00pm	Show closes	
Sunday 15th June		
10.00am	Show opens	
11.00am	Apple Presents: Compose With Logic Studio	In addition to checking out the new features in Logic Pro 8, you'll see how other tools on on the Mac can benefit musicians and producers.
12.00pm	Recording Vocals	*SOS* contributor Mike Senior shows you how to perfect your vocal sound, with live vocal demonstrations and mixing tips.
1.00pm	Producer Panel	Today's hit-makers talk tech (and other things) in our open-topic discussion forum.
2.00pm	Home Studio Recording Techniques	*SOS* Editor In Chief Paul White and Technical Editor Hugh Robjohns present a home-recording workshop, with a Q&A session to wrap things up.
3.45pm	Operating & Optimising PA Systems	*Performing Musician*'s PA correspondent Mike Crofts guides you through the ins and outs of PA setup, with live band demos courtesy of the *SOS* editorial team.
5.00pm	Show closes	

Antares Avox 2

For all your vocal-production needs

Software supremos Antares, of Auto-Tune fame, have announced an update to their Avox vocal toolkit (which was reviewed in *SOS* January 2006, and on-line at www.soundonsound.com/sos/jan06/articles/antaresavox.htm). With five brand-new plug-ins, Avox 2 is bound to appeal to engineers who need all the polishing tools they can get, for when the vocalist goes home!

Included in the Avox 2 bundle are the same five plug-ins found in the original toolkit — Throat, Sybil, Punch, Duo and Choir — which together can change the character of the voice, remove sibilance, and layer multiple voices on top of the source vocal. But the new additions will give users even more flexibility in the studio.

Harmony Engine is a four-part vocal-modelling harmony generator capable of simply thickening a vocal, or generating a virtual ensemble of backing vocalists. Each harmony line can have a different tonal character (thus avoiding the 'robot' effect), and can have different amounts of vibrato. Presets enable the user to easily create harmonies according to the root key of the incoming vocal line (and any chord they wish to 'play'), and Humanise controls allow for variations in pitch and timing, again, helping to make the output sound as realistic as possible.

Aspire is a noise processor, which can add qualities such as 'air', 'rasp' and 'smokiness' to a vocal, while the Warm plug-in adds tube tones to a vocal, creating that pleasing-to-the-ear 'warmth'.

If you're after some more obvious and unusual vocal effects, two new additions to the Avox 2 bundle could prove useful. Articulator is, according to Antares, "a modern-day version of the venerable talk box", while Mutator, Antares' 'Extreme Voice Designer', is capable of creating weird and wonderful alien-like voices from source vocals. Just select the vocal range of the incoming audio, choose how much Mutation and Alienisation you wish to add, and stick it in your signal path.

Antares' Avox 2 is available now, and at £349 is £50 cheaper than the original bundle, despite the added content. The included plug-ins are compatible with Macs and PCs, and work inside RTAS, VST and Audio Units-compatible hosts. There's also a version of the package that includes Auto-Tune, called Avox 2 AT. This is available in two versions, Native and TDM, which cost £479 and £599 respectively.

Sonic 8 +44 (0)8701 657456

www.sonic8.com www.antarestech.com

Multi-pattern Phanthera

Brauner unveil 'V' revision

High-end German mic manufacturers Brauner have launched a new version of their widely respected Phanthera studio condenser microphone, which was reviewed in *SOS* August 2007 (and on-line at www.soundonsound.com/sos/aug07/articles/braunerphanthera.htm). The new model, the Phanthera V, is essentially the same as the standard mic, but it features variable polar patterns (the V stands for Variable) and a pad, where the original model was limited to a single, cardioid polar pattern and had no pad.

The new mic is virtually identical to the original Phanthera, apart from the addition of the pad and pattern-selection switches. It sports the same yoke-style shockmount as found on many other Brauner mics, such as the Valvet and Phantom.

The Phanthera V is set to cost £2110 when it ships in the summer.

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www.sea-vertrieb.de

http://braunermicrophones.de

Big Fish Audio make a break for the East Coast

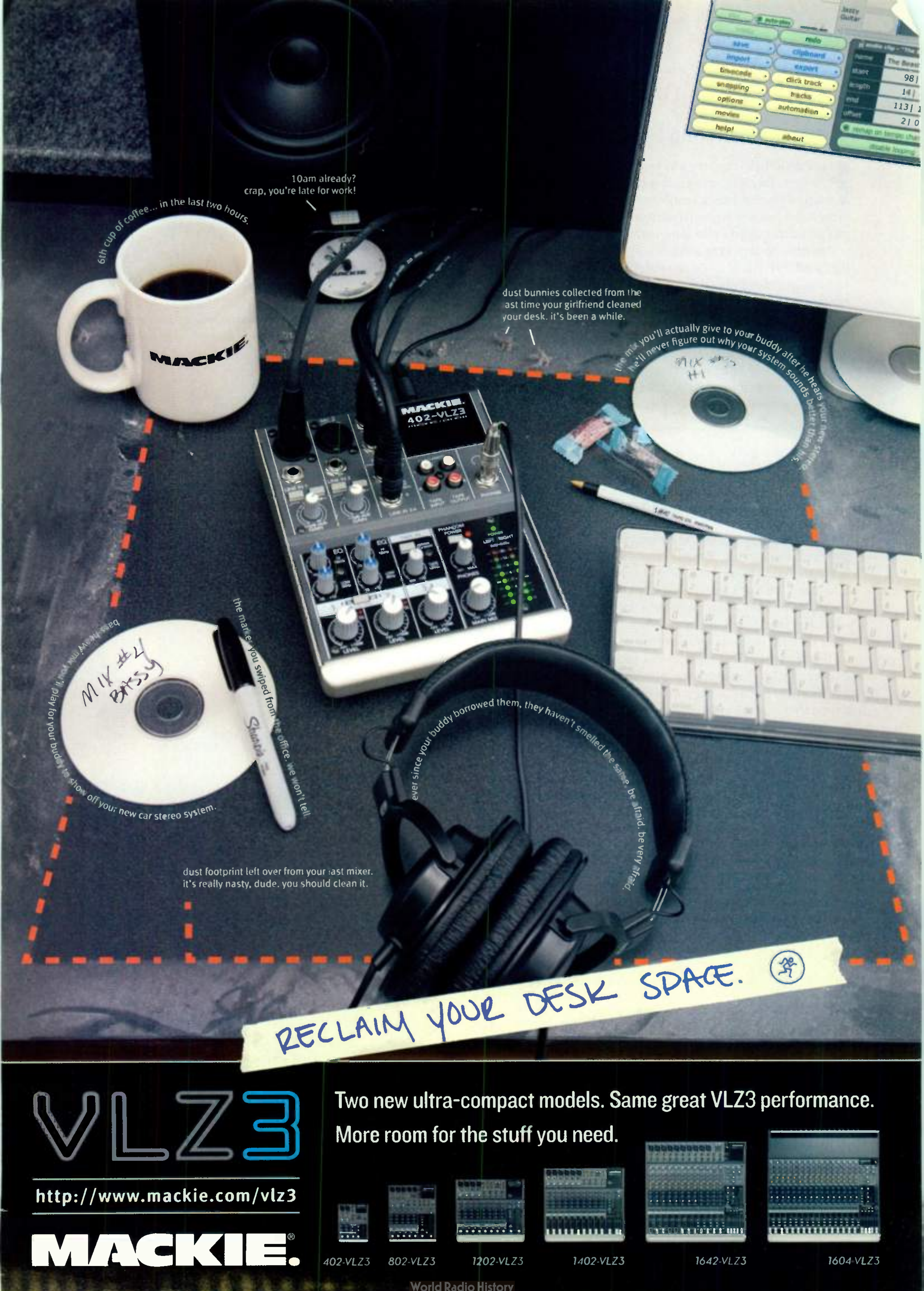
Sample creators Big Fish Audio (**www.bigfishaudio.com**) have released a new sample library dedicated to the sounds of East-coast hip hop. *Straight Outta NYC*, as it's called, contains construction kits comprising samples and loops of everything from beats and basses to horns and keyboards.

In total, there's almost 4GB of sample data in a number of formats, with over 1.5GB of raw WAV audio alone. Usefully, the drum hits from the loops are listed individually, so users can mix and match elements of different kits.

Straight Outta NYC costs £40 in the UK from Time + Space (**www.timespace.com**).

Focusrite team up with Ableton

UK-based manufacturing and distribution company Focusrite (**www.focusrite.com**) have announced that they will be handling the UK distribution of Ableton Live. The move brings Focusrite their first software title, which will complement their current line-up of hardware from themselves and Novation. Chris Gooddie, Focusrite's Sales Director commented: "integrating the best available hardware and software puts more power into the hands of the creative musician".


10am already?
crap, you're late for work!
6th cup of coffee... in the last two hours.
MACKIE
dust bunnies collected from the last time your girlfriend cleaned your desk. it's been a while.
the mix you'll actually give to your buddy after he hears your new stereo.
he'll never figure out why your system sounds better than his.
MIX #3
#1
MACKIE
402-VLZ3
MIX #4
BASSY
bass-heavy mix you'll play for your buddy to show off your new car stereo system.
the marker you swiped from the office. we won't tell.
ever since your buddy borrowed them, they haven't smelled the same. be afraid. be very afraid.
dust footprint left over from your last mixer.
it's really nasty, dude. you should clean it.
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Focal CMS50 & CMS65

Compact Monitoring Systems

Focal are a French company who have been making monitors for over 20 years. They're incredibly well respected in the hi-fi world, and car audio buffs often look to Focal for top-notch in-car audio. But it's only in the last decade that they've turned to the pro-audio industry, and so far they've had their sights aimed at the top end of the market.

The latest products to be announced by Focal are the CMS50 and the CMS65 (CMS stands for Compact Monitoring System), which are both two-way, bi-amped nearfield monitors. They're aimed at home- and project-studio users, and are priced accordingly, but are apparently made to the same rigorous standards as Focal's other products. The smaller of the two, the CMS50, features a five-inch woofer, while the CMS65 (pictured) has a 6.5-inch driver, and both sport one of Focal's signature aluminium/magnesium inverted-dome tweeters.

Like almost all other nearfields, the CMS monitors have a range of controls to tweak their frequency response. On their rear panels there are high- and low-frequency shelving filters, a high-pass filter, and a 'desktop notch' function that helps to suppress troublesome low-mids (centred at 180Hz) when the monitors are used on a flat surface, such as a desk.

Both CMS models feature die-cast aluminium enclosures that have been acoustically treated to remove the unwanted resonance that Focal say is "frequently encountered on that type of cabinet". They're available through SCV in the UK.

SCV London +44 (0)208 418 1470
www.scvlondon.co.uk www.focalprofessional.com

Studiologic Numa

Controller legends announce new master keyboard

Keyboard and MIDI controller specialists Fatar have announced a new master controller in their Studiologic range. The manufacturers say that the new controller, the Numa, is "the most pianistic and flexible MIDI controller ever designed", which is quite a claim. Unlike many of its competitors, the Numa doesn't have masses of knobs or faders: instead, attention has been paid to making the keyboard feel as much like a real piano as possible. All of the 88 hammer-action keys are solid (Fatar say that the black keys of some competing controllers are hollow), and the keyboard is graded, so the response of the keys gets progressively lighter further up the keyboard.

The Numa has a touch-sensitive user interface, complete with a large LCD that displays whichever parameter is being programmed. Four hot keys, arrow pads and a scroll wheel are used to enter and edit information, and a function called YouPlay guides the user through a process that configures the dynamic response of the controller to best suit their playing style. A pitch-bend wheel is situated, rather unconventionally, on the side of the device, above sockets for MIDI, USB, foot pedals and a power supply.

The body of the Numa is made from a single piece of moulded ABS plastic, which helps to keep the weight of the controller down. Usefully, it has a sliding top panel, which extends to enable the user to place a laptop, sound module or controller on top of the keyboard. Currently, it's only available in 88-note format, although smaller versions may be in development. The Numa costs £900 including VAT, and should be available by the time you read this. Check out the full *Sound On Sound* review on page 164 of this issue.

Arbiter +44 (0)208 207 7860 www.arbiter.co.uk www.fatar.com

ESI Duafire DC power supply

In last month's What's New section, we stated that the AC-DC power adapter for the ESI Duafire is an optional extra. It is, in fact, included. Apologies for any inconvenience caused.
www.timespace.com
www.esi-audio.com

Another Black Hole from JZ Mics

Following the announcement in *SOS* April 2008 of the JZ Mics (**www.jzmic.com**) multi-pattern Black Hole and fixed-cardioid Black Hole SE, the Latvian manufacturers have announced the third revision of the large-diaphragm condenser, the Black Hole PE. Like the SE, the new model has a fixed-cardioid polar pattern, but features a pad, which has -5dB and -10dB settings. It costs 1395 Euros, which was £1222 at the time of writing. Coinciding with the launch of the Black Hole PE, JZ Mics have also released a new shockmount and pop shield that fit all the Black Hole models. The shockmount, a conventional elastic-based one, replaces the rubbery mount that shipped with the original Black Holes and is, according to JZ, "devised to use on crazy applications where the included stand holder is not enough to handle all thunderclaps". So now you know. The pop shield and shockmount are bundled together, as the former connects to the latter, using two thumb screws to fix the pop shield in place. The bundle costs 240 Euros (around £195).

New products from KB Covers

KB Covers specialise in flexible, removable covers for Apple computer keyboards, which give the user a map of key commands for specific applications. New from the company are keyboard covers for the latest verion of Logic, and they've announced that the whole range is now available to fit the new, slimline Apple keyboards. For further details, check out **www.kbcovers.com**, where you can purchase the products.

Steinberg announce new hardware

With a little help from Yamaha

At this year's Frankfurt Musikmesse, Steinberg were showing three new hardware products — two interfaces and a desktop DAW controller — designed to provide 'Advanced Integration' (AI) specifically with Cubase, their own DAW software package. All three products have been co-developed and manufactured by Yamaha, Steinberg's parent company.

The two new interfaces, the MR816X and the MR816CSX (pictured below), are 1U rackmountable devices that connect to the host computer via Firewire. Cleverly, like many new audio interfaces coming on to the market, they have on-board DSP, and these run Yamaha's acclaimed REV-X reverb algorithms and the Sweet Spot Morphing Channel Strip plug-in, the latter only being operable in the CSX derivation of the interface. DSP-powered plug-ins show up as VST3 objects, which can be inserted into channels inside Cubase, but they can also be applied directly to the input channels using a bundled software mixer called MR Editor.

Both interfaces have 18 simultaneous inputs and outputs comprising eight analogue inputs and outputs, an eight-channel ADAT bus, and stereo S/PDIF in and out. All the analogue inputs have mic preamps and front-panel gain pots, so you can set the levels of input channels independently (unlike some competing products). What's more, the first two analogue inputs have insert points, so you can apply processing from external hardware without having to route signals through the interface's converters. Another useful feature is the inclusion of a word clock input and output, enabling connection to professional synchronisation equipment.

The third new product is the CC121, a desktop controller that, again, is designed especially for use with Cubase. It communicates with the connected computer using USB, and 'talks' directly with Cubase, requiring no parameter assignment. There's even a 'Cubase Ready' LED that indicates when hardware and software are sync'ed.

The CC121 has a single motorised fader, situated among hardware versions of the controls that you find on a Cubase channel strip, such as mute and solo buttons and a pan control. The central section has a bank of 12 knobs dedicated to controlling the bandwidth (Q), frequency (F) and gain (G) of Cubase's built-in EQ. There's also an eight-button transport section that includes loop, fast-forward and rewind controls. The right-hand side of the device has what's called the AI Knob, an endless rotary encoder that can be used to edit any effect, instrument or channel parameter inside Cubase; just point the mouse cursor at the parameter you want to edit, and the AI Knob takes over.

The announcement of the new gear was, according to the company, "an exciting step towards our long-term objective of providing uniquely integrated hardware and software products". We'll keep you updated with any further developments.

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www.arbiter.co.uk

www.steinberg.net

Loopmasters enlist drum & bass heavyweight

Sample library specialists Loopmasters (**www.loopmasters.com**) have released a new sample pack created by Nu Tone, one of the UK's best respected drum & bass producers. The sample pack, which costs a mere £25, contains 172 samples in a range of file formats, and 21 software sampler patches compatible with Kontakt, Reason, Halion, EXS24, and any SFZ-reading application. Samples include the expected drum loops and bass lines, but also some surprising piano and Rhodes samples, played by Nu Tone, who has an impressive range of academic music qualifications. Nu Tone and Loopmasters say that you don't need much else apart from this sample pack to get the basics of a track together. Visit Loopmasters' web site to download the sample pack, and to watch a video, which gives you an insight into how Nu Tone makes his tracks. Loopmasters' Nu Tone drum & bass sample pack will also be available for download from Time + Space (**www.timespace.com**).

M-Audio UK celebrate 20th birthday

The UK division of gear manufacturers M-Audio have launched a new web site to mark the 20th birthday of the company. Features of the new site, which is found using the same web address **www.maudio.co.uk**, include a Community area, where users can go to get more information on M-Audio's wide range of products, by watching videos and reading artist interviews.

Also, a new M-Audio University will allow users to complete tutorials that should help to improve their recordings and mixes. There are also buyers' guides to help potential customers choose the best gear for their requirements. M-Audio say that general navigability of the site has also improved, so pages can be found more quickly than before. If you get a spare moment, why not check it out?

Genelec miniaturise

'First Step' products launched

Genelec have released a new monitoring system designed specifically for desktop use. The 6010A two-way nearfield monitor and 5040A small-scale subwoofer both employ active designs, and are aimed at users who want to take their "first step into the world of professional audio". The monitors resemble Genelec's 8000 series in terms of looks, with the rounded corners of the trademarked Minimum Diffraction Enclosures (MDE), metal grilles and cute Isopod feet, which help to decouple the monitor from the surface on which they're placed.

As with the monitors in the 8000 series, the 6010A has its connections on the rear panel, but there's no front-panel volume control or power switch, as with the 8020 or 8030. Genelec say that they're capable of handling frequencies between 74Hz and 18kHz, producing peak SPLs of 102dB at a distance of one metre using a pair of 12-Watt amplifiers. The input is on RCA phonos, and there are controls for input sensitivity, bass tilt and engaging a Desktop mode that optimises the frequency curve for use on tables.

The frequency response of the alien-landing-craft-shaped 5040A subwoofer stretches down to 35Hz, and a 40-Watt amplifier enables the 6.5-inch woofer to kick out peak SPLs of 96dB at one metre. It has separate inputs for all six channels of a 5.1 surround signal, with link outputs that can feed the five satellite speakers. This arrangement enables a remote volume knob, connected via a cable to the subwoofer, to control the level of a whole six-speaker system. So, using the input sensitivity control on the 6010A to select the optimum relative levels of the system, you can 'set and forget', knowing that your system will sound the same every time you switch it on.

The 5040A and 6010A are both magnetically shielded, so can be used next to CRT monitors and TV screens. The 6010As retail at just under £200 each, with the 5040A subwoofer costing around £400.

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Cool front

Rain Recording's Solstice

American PC manufacturers Rain Recording have been key players in the audio computer market for some years now. Starting out in the US in 2001, they immediately targeted customers wanting stable, 'turnkey' solutions that work out of the box. In October 2006, they spread their wings and flew across the pond to form a UK office, where they're starting to get a loyal following.

Their latest offering is the Solstice, the most affordable 'workstation' machine in their range. It's available in two different dual-core configurations, both with AMD's Athlon processors, and a mighty quad-core setup, which features an AMD Phenom-series CPU. As with most computers, buyers can specify additional RAM and hard-disk space when purchasing, and also retrofit parts at a later date, but the standard specification should be good enough for most applications.

All configurations of the Solstice come with seven USB 2.0 ports (including one mounted internally for hiding valuable dongles) and two Firewire 400 ports. There are slots for two PCIe cards and two standard PCI cards, and built-in Gigabit Ethernet enables connection to a fast network. The Solstice range starts at £849 including VAT in the UK.

Also new from Rain is an update to the Element system, their mid-range workstation PC. Element machines now feature processors with Intel's latest Penryn technology, and can have up to six Terabytes of hard disk space installed. For more information on all the new products and to buy direct from the manufacturer, head to Rain Recording's web site.

Rain Recording +44 (0)845 094 3964
www.rainrecording.co.uk

TCM Mastering goes on-line

Kent-based TCM Mastering have been in operation for over 15 years now, mastering records for the likes of EMI, Sony, Warners and Universal. Founder Ted Carfrae has worked with a range of artists from Marc Almond and Liberty X to Katherine Jenkins and David Cassidy; he even landed the incredible job of restoring and remastering the BBC Record Library. Ted's latest venture is an on-line service for TCM Mastering that allows customers to upload their material to be mastered at the TCM facility. Single tracks can be processed for £25, or albums of up to 14 tracks can be mastered for £250. Ted explains the reasons for the new enterprise: "Professional, quality mastering is no longer just for the big guys, I know first hand how hard this business is and encouraging new talent is something that I am passionate about". For more information, head to the TCM web site.

TCM Mastering +44 (0)870 034 9887 **www.tcmmastering.com**

Tascam USB 2 interface updates

Tascam (**www.tascam.co.uk**) have updated the drivers for their US122L, US144 and US1641 USB 2 audio interfaces. Version 1.01 of the driver brings support for the latest Apple operating system, Leopard (Mac OS v10.5), and improves performance when operating at 88.2kHz and 96kHz sample rates. Latest drivers can be downloaded from the Support pages on Tascam's web site. The company have also upgraded the included version of Cubase that ships with the US-series devices, so new customers will now get the latest LE4 revision bundled with their purchase.

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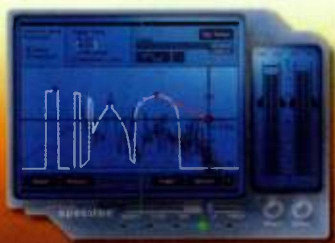

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Charter Oak's saplings

M900T & SCL1 join the arboretum

US-based Charter Oak Acoustic Devices have announced a new microphone, the M900T. Like the standard M900, it's a small-diaphragm end-fire mic, but it features a tube-driven head preamplifier, along with a US-made Cinemag output transformer, rather than employing solid-state, transformerless circuitry. It also comes with the same three capsules as the M900, giving the mic omnidirectional, hypercardioid and cardioid capabilities. Situated on the mic's body are a -15dB pad, which can be handy when using the mic on loud sources, and a high-pass filter, which attenuates sounds below 75Hz and will help to reduce stand-borne rumbles.

Also new from Charter Oak is the SCL1, a solid-state compressor/limiter, which is apparently capable of providing a constant output level that's free of 'pumping' artifacts, regardless of the input level or frequency. That's about all we know about the SCL1, as it's due to be formally announced during the Amsterdam AES Show in mid-May. We'll keep you up to date.

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www.asapeurope.com www.charteroakacoustics.com

Realize Music Challenge

Notion Music, the developers of Notion composition software, are running their Realize Music Challenge again this year, after the success of last year's contest. The grand prize is to have your composition played by the London Symphony Orchestra and recorded at Abbey Road studios.

Like last year, there will be three categories: Young, Emerging and Professional, from which the grand prize winner will be picked. There will also be section finalists, who will receive a prize of $1000, with a further nine runners up winning a Notion software product of their choice. As well as having their composition professionally played and recorded, the grand prize winner will be flown to London and put up in a hotel, so they can sit in on the session. They'll also receive a cash prize of $2500.

A panel of distinguished composers and conductors will judge the entries, which must be between five and 10 minutes long, and must be compsosed within Notion software. For full details, visit www.notionmusic.com/contest. The deadline for submission is the 31st May 2008, so you'd better get writing!

www.notionmusic.com

Izotope ANR-B

Software company enter the hardware domain

Izotope will be known to many as software designers. Their range of plug-in effects is found in studios around the world, and their noise-reduction processors are used for many applications, from audio restoration to album re-mastering. But they've just made the leap into the world of hardware, with the introduction of the ANR-B noise-reduction processor.

The ANR-B is a two-channel device that can work in dual-mono or stereo mode. It scans its audio inputs for broadband noise, hum and other unwanted artifacts, and suppresses them in real time, improving the intelligibility of the source. Front-panel controls are simple: each channel has a single suppression knob (ranging from minimum to maximum), which allows the user to vary the amount of noise reduction taking place. Surrounding buttons enable the user to bypass the noise-reduction circuitry, preview the noise that's being removed, and command the unit to learn and reduce the level of unwanted sounds according to a 'noise profile'. A final button marked 'adapt' puts into operation Izotope's acclaimed adaptive noise-reduction algorithm, which reacts to changing noise over time, allowing for automatic operation with little or no input from the user.

The Izotope ANR-B is designed mainly for the broadcast market, where live-to-air telephone feeds, for example (such as those used in radio talk shows) need to be clean and intelligible. But it has uses in the recording studio. For example, imagine you get the perfect vocal take, but the booth air conditioning was left on, and there's hum, whine and hiss all over the audio. Simply run the vocal through the device and hey presto: take restored!

Rear-panel connections include analogue line inputs and outputs, AES-EBU I/O, and various remote-control options. Unfortunately, as you'd expect, it comes at a price. Izotope's ANR-B costs just under $5000 in the US (just over £2500 in the UK at the time of writing). It is due to become available later this summer.

In other news, Izotope have ported their iDrum virtual drum machine, until now a Mac-only application, to the PC platform. The latest revision (version 1.6) is compatible with Vista- and XP-based machines, and works as a stand-alone virtual instrument or as a plug-in inside a VST or RTAS host. A library of samples is included with iDrum, along with a number of preset drum kits. What's more, users can import their own samples into iDrum, allowing for custom kit creation. Distribution for iDrum is through M-Audio (www.maudio.co.uk), while the ANR-B will be handled by Izotope themselves.

www.izotope.com

Q&A

Your technical questions and queries answered

Q How can I use my sampler with Logic?

I've recently upgraded to an Intel iMac with Logic Express 7, and have been experiencing a few problems while transferring onto this format. My previous setup was an old Mac Performa running Opcode's Studio Vision. I was using this with an Emu ESI2000 sampler. I had been running all of my audio in the sampler using samples, recorded audio and various instrument and synth presets. Working this way I developed a style and sound that I really like, especially with drums, but I had real problems mixing and finishing tracks.

I'm still using the sampler to make beats and then I'm trying to transfer them into Logic. I'm finding this a really awkward process, as my audio always plays back in a different place. So, if I record different tracks individually, it's always difficult to get the timing spot on. How can I change this? Also, is there a way that I can record my drum samples into Logic, and then play them in Logic as MIDI, such as I would when using a soft synth or internal instrument? I don't seem to have a software sampler with Logic Express, so maybe I need the new Logic 8? The problem at the moment is that I want flexibility to edit the MIDI data right up to the final stages — editing audio is obviously much more restrictive with things like beat programming. I also want to be able to use the production facilities on offer in Logic while using MIDI programming.
Adam Hurst

The ESI2000 (top) can be fitted with a Turbo board to add extra outputs, if you want to record multiple instruments at once, but a soft sampler, such as Logic's own EXS24 or Native Instruments' Kontakt, can be much easier to use and more powerful than older hardware samplers. You can easily trigger imported samples from such soft samplers.

Reviews Editor Matt Houghton replies: If you're finding that the timing is not sufficiently consistent to record the different parts in separate takes, you could try recording the different sampled instruments simultaneously to different tracks in Logic. If you do that before you add other elements to your project, they'll all be in time and you can calculate the tempo and adjust the project in Logic to suit before adding other elements. This will require your sampler to have sufficient outputs (there are four, but it is expandable to eight, with further digital channels available, if you have the Turbo board installed), and you'll need a soundcard with the same number of inputs.

I'd recommend getting a soft sampler, though, as you'll eventually find it much easier to use, and much more powerful than your old system — and if you want to you can import all your old samples from the Emu into the soft sampler and trigger them in there instead.

Logic Express 8 includes the EXS24 soft sampler, whereas v7 only includes the cut-down EXS24P (which will only play back pre-programmed libraries). EXS24 is also part of the 'full fat' version of Logic, so you'll ▶

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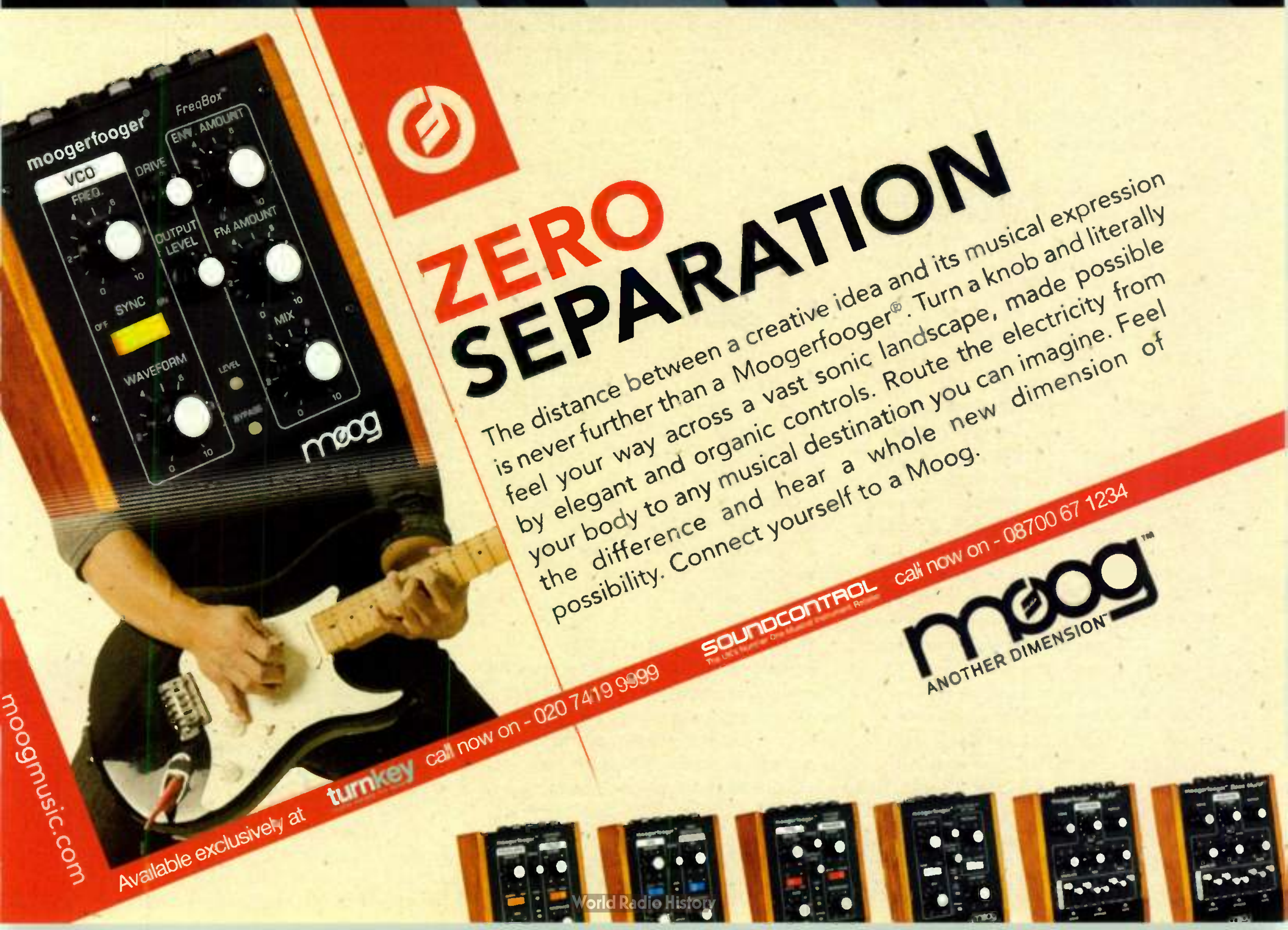

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Q&A

need to upgrade to Logic Express 8, Logic Pro 7 or Logic 8 if you want to use EXS24. Alternatively, you could buy a third-party soft sampler like Native Instruments' Kontakt, or maybe even download a freeware sampler: try the database at www.kvraudio.com if you want to search for one.

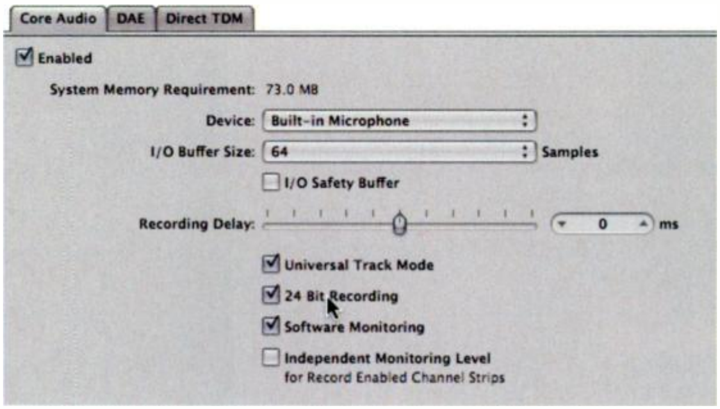
In most soft samplers, you need to record your samples in as WAVs or AIFFs using a program like Logic (any audio recording/editing software will do) and then load the samples into the sampler. We've run a number of workshops on EXS24 in the past, and the basics haven't changed a great deal, so you might find it helpful to read a few of them if you're new to all this. Try this one to get started: www.soundonsound.com/sos/aug02/articles/exs24.asp.

Q Do I really need 24-bit recording?

Is 24-bit recording any better than 16-bit recording in a home studio, given that the only qualitative difference is (apparently) the noise floor? In 99.9 percent of home studios, this difference will be masked by the ambient noise of the studio, and by preamp or mic self-noise. From what I've learnt from Hugh's articles on digital audio, I'm sceptical. Is it really worth me switching from 16-bit to 24-bit recording?
***SOS* Forum Post**

Technical Editor Hugh Robjohns replies: The characteristic defined by the word length is the available dynamic range, or where the noise floor sits in relation to the peak level. A properly dithered 16-bit system provides a dynamic range of around 93dB, while a mid-budget 24-bit system should be able to deliver something around 115dB or so. The implication, then, is that you could work with about 20dB more headroom in a 24-bit system without any increase in the apparent system noise over the 16-bit mode. For most people, the additional computer overhead of operating with 24-bit word lengths instead of 16 has a negligible effect on processing power, and the additional file-storage requirements are inconsequential, given the size of modern hard disks, so the lower system noise and greater headroom margins are a welcome benefit for a negligible cost.

Having said all that, you are right in that for most people recording at home, the recording noise floor is almost always defined by the ambient room noise rather than the digital system noise. Therefore, you



Recording at 24-bit increases the available headroom in the system, although this may not offer any practical benefits for signals recorded with microphones in an average home studio.

can usually work with reasonable headroom margins at 16-bit word lengths while maintaining the ambient noise floor comfortably above the system noise floor. In other words, the dynamic range of the source recordings is typically far less than the capability of a 16-bit recording system. Where that changes is if you record electrical instruments such as guitars and keyboards via DIs. In these cases, the noise floor is often far lower than that possible when recording with microphones in a home studio, so the dynamic range of a 16-bit system may start to become a limitation.

The bottom line is, if you are happy with the results you get with 16-bit, there may be no benefit in changing to 24. However, most people do find that 24-bit working enables greater headroom margins without noise penalties, and that in turn makes recording less stressful and mixing rather easier. Personally, I can see no negative side to recording at 24-bit, and only positive benefits.

Q Do you have any soldering tips?

I have a vast amount of mic cable and a load of three-pin Neutrik NC3FX female connectors ready for soldering (there is no need for male connectors as the other ends are going to my patchbay). I've done a limited amount of soldering in the past and just wondered if there were any handy tips? I've got all the tools, including a 'desolderer' and a multi-clamp to hold things in place.
***SOS* Forum Post**

Technical Editor Hugh Robjohns replies: Wiring XLRs is one of the easier DIY challenges to tackle once you have a little experience under your belt. The solder contacts are large and fairly tolerant of over-heating, while cable trimming and dressing aren't too critical either. It's a good way for novices to get started and build their confidence in soldering and cable preparation.

The first step is to thread the cable through the strain-relief collar. It's immensely frustrating to have to unsolder a perfect connection just to put the collar back on, so don't forget! If you're planning to use slide-on cable identification markers, then put these on before the collar.

With the cable threaded through the collar, strip off the outer sheath and prepare the individual wires. It's best to use a proper wire stripper for this, and it's a good investment to buy the best self-adjusting type you can find. Using the wrong tool could damage the insulation and/or nick the wires. The former risks short circuits, the latter risks breakages and open circuits.

With the wires trimmed to an appropriate length and stripped, it helps to 'tin' them. Twist the strands tightly together, heat each wire, and feed in a little solder so that it flows and coats the metal. Don't get it too hot, though, as the insulation is likely to melt or shrink.

It's good practice to then introduce some insulation sleeving over the screen wire, to prevent short circuits inside the connector body. You can buy bags of pre-cut rubber sleeves that make the job neat and tidy, and they also allow for easy re-working if you need to repair the connection at a later stage. Heat-shrink sleeving can be used, but makes the job of repairs slightly more difficult. Also, at this stage, decide which of the two core wires will be denoted as 'hot' and which as 'cold' — make a decision and stick to it!

Next, fix the connector into a small vice

It always helps to tin any wires before soldering joints. Use a little solder on the twisted strands to lightly coat the wire.

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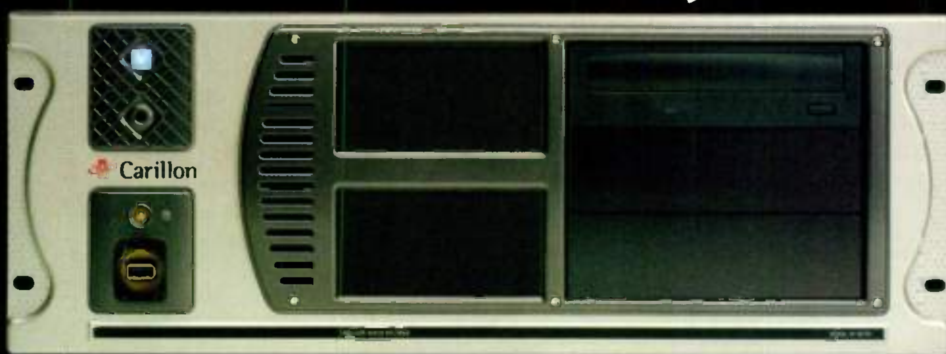

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Make sure that your connector is securely fastened in a vice or clamp to stop any movement while you solder.

or multi-clamp in such a way that allows you to access the three pins, and then tin each of them. If you don't have a vice or clamp, then you could use a cable tester, or a similar device (the heavier the better to stop things moving around too much while you are working), as long as the connectors are fitted somewhere convenient. Also, I find it helps to point the female end of the connector slightly uphill, otherwise the solder disappears inside the terminal!

Identify the pin numbers (these are usually embossed beside each pin on both the inside and outside of the connector) and remember that the male and female connectors are mirror images of each other. For that reason, it makes sense to wire all the connectors of one sex first, and then all those of the opposite sex. If you alternate between wiring male and female connectors, it is inevitable that you will end up wiring at least one backwards!

Once you have correctly identified each pin, offer the 'cold' wire from the cable to pin 3 (it's the the one in the middle). Hold it steadily in place while you heat both the wire and the terminal, and then feed in some solder. As soon as the solder melts, remove the heat and hold everything still until it cools and the solder sets — it should be just a few seconds.

Leave any mistakes to cool before using a desolderer to clean the joint of any old solder, and then start again.

The key is to be be as quick, clean and precise as possible. Don't keep jabbing at the solder joint. If it isn't right, do the next joint and come back to any problem later once it has cooled. Use the solder-sucker (desolderer) to clean the joint of any old solder and start again.

Repeat the process for the 'hot' wire on pin 2, and then the screen wire on pin 1. I don't recommend linking pin 1 to the connector shell as it usually causes more trouble than it's worth, but there is a central tag for that purpose if you should require it.

If you're happy with the solder joints, fit the plastic insulation and cable clamp sleeve. Slide the whole thing into the metal shell (being careful to align the rib and notch), ease the release button into place and screw the end-gland up tight. Always test the cable to make sure that it's wired correctly (with particular attention to pin 1/pin 3 reversals). In the case of patchbay wiring, it pays dividends to label the cable ends in some meaningful manner, making it easier to replug things later.

Q Where should I put my violas?

I'm going to be recording a string section. I'll be recording the section using both close-miking and a Decca tree in order to have maximum recording flexibility. We'll be doubling a lot of sections and, further down the line, adding a 10-piece brass section and some samples to make the recording sound fuller. Can you please advise me as to how I can place my 14 string players in order to get the most out of them? I have five first violins, three second violins, two violas, two cellos and two double basses. As I only have two violas, I'm worried that placing my players in their normal orchestral positions will leave a big gap in the middle of the formation, where the violas would be placed. I'm trying to emulate a large orchestra and have been mulling over whether a wider spread of players will help or not. The danger is that the players will not blend, and instead sound like individual players, which must be avoided. What's your advice here?
Via email

***SOS* contributor Mike Senior replies:**
If this orchestral arrangement is designed to complement some kind of pop or rock track, then leaving a bit of a hole in the middle and over-emphasising the width of the section might be just what you need. This will keep the strings out of the way of other more important parts of the track, such as lead vocals. Plus, with everything else going on, you probably won't notice a few blend problems in the grand scheme of things.

However, it sounds to me more like you're trying to mock up an orchestral score for a film or something similar, in which case the situation's a bit different. You might find that you have mix-balance problems with only a couple of violas. A lot depends on the specific players, instruments, and arrangement, but the numbers are a little against you. If it's just the stereo image you're worried about (and you're not happy for this hole to be filled by the winds and brass), then move the violas a little closer to the cellos, and bring the whole string formation tighter together. If balance is still the problem, then you're right that pushing the close mic nearer to the violas, to compensate for their minority, could easily affect the blend. If you have time, you could try overdubbing the violas as a separate part to give you a fall-back option. (If the lack of other parts would make it impossible for the violas to coordinate their part solo, then put in a token player on whichever other sections are required to make it possible.) The important thing to remember when doing this is to record the Decca tree alongside the viola close mics: this will give you more viola ambience to help gain the illusion of more viola players. That said, if you're going to be including samples anyway, you might find that you can deal with any imbalance at that stage. A pair of violas, especially if doubled, should be enough to disguise quite a lot of extra sampled viola parts. If there is an exposed

With only two violas in the section, there could be problems with regards to the mix balance. If you have time, you could try overdubbing extra violas to make up for the shortfall.

viola line that you're worried about somewhere, but you still want to prop things up with more samples, then adjust the arrangement a little in that spot to double the violas with some of the violins or cellos. If all else fails, and you have to turn up the viola close mics in a way that affects the blend, then you might be able to compensate with a little added reverb on those mics.

However, If you're aiming to capture a very natural sound, more akin to a classical-style ensemble, then you have to ask yourself how much real stereo positioning you'd actually hear from a couple of dozen rows back in a concert hall. I'd question the idea of widening things too much if that's your target sound.

Sound Advice

#11 How do you record a choir?

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

Hugh: The country is full of choirs, with varying levels of skill and musicality. There are many different styles of choral music, and many different performance spaces, so there is no 'one way' to record them. Every single case will be different and require different approaches. Also, choral music is, in my opinion, one of the hardest genres to record, and it can show up any weaknesses in equipment or technique remarkably well!

The first consideration has to be finding a suitable acoustic space in which to record. With conventional or traditional choirs, I have found that the best approach is to keep it simple, with just a stereo pair of microphones, and perhaps some outriggers or spaced mics if the situation calls for it. That means the acoustics of any performance space are critical to the overall sound, so choosing a good venue is paramount.

Paul: On the other hand, if you have access to a project studio, you could try to record a choir there. But space will be limited, so think about recording small groups in stages. Of course, the choir may not be comfortable with this, and even if they agree, you'll have to supply headphones for everyone who is overdubbing parts. On the upside, though, you can pan the different parts separately and also add stereo reverb to the end result, giving you more control over the overall sound.

Hugh: I'd always go for a 'real' choral setting if possible, and there are plenty of great-sounding churches and halls around. Although finding one that is well away from traffic, flight paths and football playgrounds can be a challenge! Bear in mind that if you are recording a gospel choir, they tend to be recorded closer — sometimes even with individual mics — so excellent room acoustics aren't quite so vital. Even so, a reasonably good sound in the room minimises colouration problems and helps the choir to perform well.

Paul: Where you have control over the way the choir is arranged (not in the musical sense), you may be able to put the loudest and the least good singers furthest from the mics, or adjust the number of rows to change the width of the choir. This is something I often had to do with school choirs.

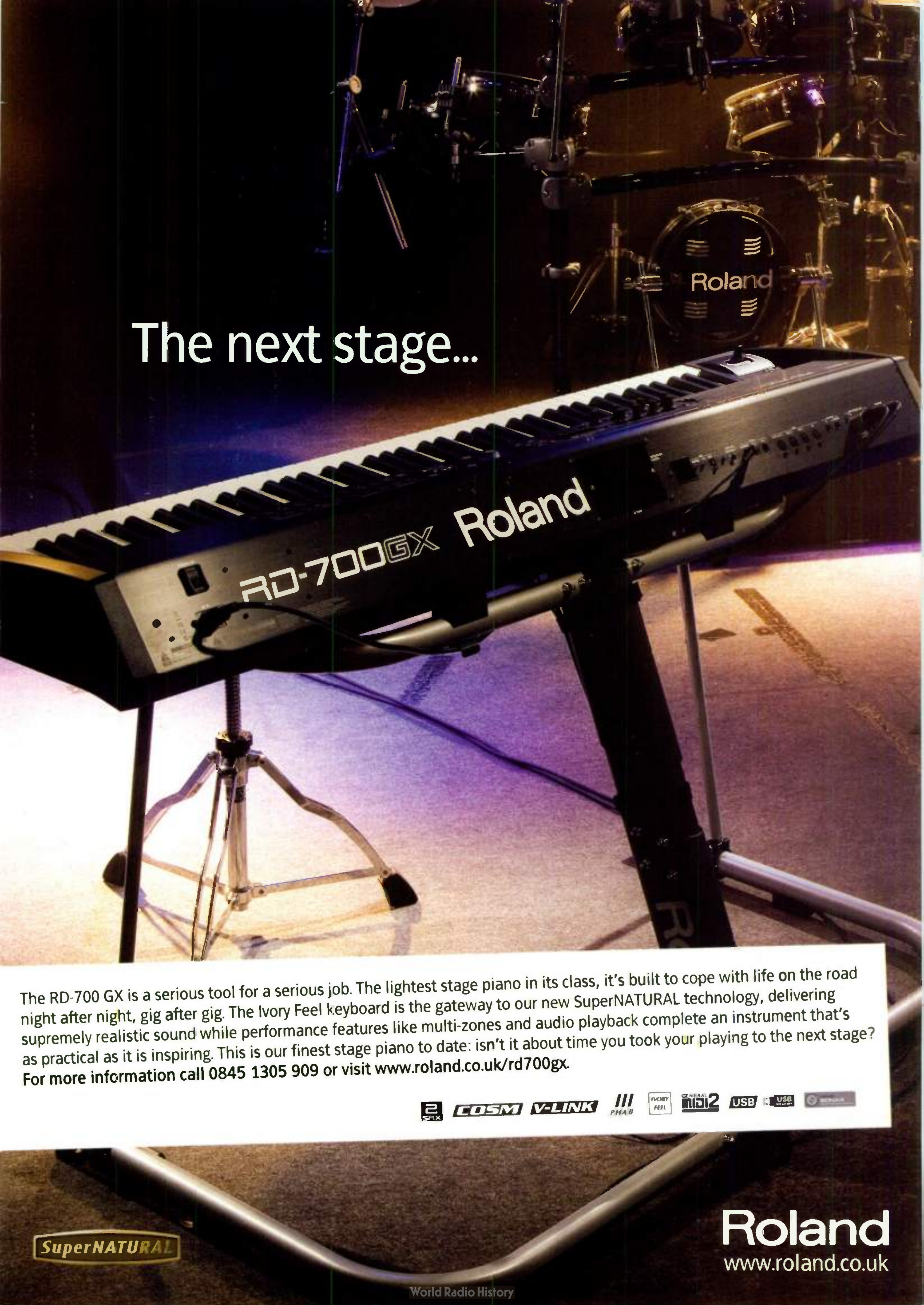
There was also one recording I had to make in an empty converted chapel where the acoustic was actually too lively to give a good result. So, we set up a folding table-tennis table behind the choir to form a vertical screen, then draped it with gym mats and clothing to damp down the acoustic a little bit. You'll probably not find many gym mats in your typical church, but if you scout out the venue before the recording session, you'll be able to come prepared with some duvets and tall lighting stands for a similar effect.

Hugh: Microphone setups will depend on acoustics, available equipment and personal taste. I usually start with either an ORTF arrangement (cardioids angled at 110 degrees and spaced 17cm apart), or a coincident pair — often crossed figure-of-eights — if the acoustic will allow. Small-diaphragm mics are generally better for capturing a choir as they suffer less coloration for off-axis sounds, and there will be a lot of that! With a large choir, adding omni outriggers can sometimes help to increase the spaciousness of the recording (especially if using a coincident pair). You can also use outriggers, positioned further back from the choir, to enhance the reverberation pick-up if the space is a little dry and confined. If there is an accompanying instrument, such as a piano or organ, it's usually necessary to add a spot mic or two to help focus the sound and stabilise its position in the image, but be careful not to overdo it and make it stand out.

Paul: While Hugh favours coincident mics, I tend to use spaced cardioids (or omnis where the acoustic is sufficiently sympathetic). I space these apart by around half the width of the choir and set them up on stands at least a metre higher than the front row of the choir (assuming there are high ceilings). This provides a relatively even coverage over choirs that are two or more rows deep, and also ensures the singers at either side aren't left out. The mic distance from the choir will need to be adjusted to acheive a good balance of direct and reverberant sound, though you can always err on the side of 'too dry' by keeping the mics close (one to two metres in front of the front row) and then add additional, artificial reverb if necessary.

Hugh: I always record flat, with no EQ or compression. It simply isn't needed and, if it's required at all, is far better to judge in a known acoustic. Depending on the complexity of the work and the intentions for it, I would either track each mic separately and work on a mix later, or I would mix straight to stereo in one go. In my broadcasting days, I would have pushed and pulled the faders as necessary, controlling the major dynamic changes to suit the restrictions of live broadcasting. With modern recording equipment there is plenty of dynamic range to capture the whole recording with fixed levels.

Paul: It pays to leave a lot of headroom when recording choral material. Not only will the volume increase through the early part of the day as the choir warms up, but the complex interactions of lines and harmonies can create complicated beat patterns that benefit from plenty of headroom to breathe. With 24-bit recording systems there won't be any problem with the noise floor, so don't be afraid to leave 10-15dB of headroom on peaks.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Powered Monitors

Samson MediaOne 4a

Obviously, a good multimedia system is better than a bad one, but will a good one do in a home studio?

Paul White

If you're running a serious project studio with the aim of producing release-quality material, then you need the best monitors you can afford. However, if you're more concerned with songwriting or making basic demos and you currently use off-the-shelf multimedia desktop speakers with your computer, you could still benefit from upgrading to a better multimedia speaker. Samson's MediaOne active monitor range is aimed at exactly this market, and the company's expectation is that the same speakers will be used for recording, music playback and gaming.

> SOUND ON SOUND
>
> **Samson MediaOne 4a £129**
>
> *pros*
> - Good-quality desktop speakers.
> - Reasonably priced.
>
> *cons*
> - There's only so far you can go with multimedia speakers in a studio.
>
> *summary*
>
> The Media One 4as straddle the gap between budget multimedia speakers and dedicated studio monitors, and are a worthy upgrade from the former if you're not ready for the latter.

Overview

The MediaOne 4a monitors reviewed here are essentially passive speakers, in that they use passive crossovers set at 4kHz, but one of the speaker cabinets also houses a 20 plus 20W stereo power amplifier that drives both channels — which means that the more correct description for them is 'powered', as opposed to 'active' monitors. The latter term generally describes systems where separate amplifiers drive the woofer and tweeter, each fed from an electronic crossover.

The rear-ported and rather stylish cabinets play host to a copolymer-cone four-inch woofer and a one-inch, ferrofluid-cooled, Neodymium-powered silk-dome tweeter, set in a waveguide to control its dispersion. There's a front-panel volume control for the power amplifier, as well as a headphone output and an additional 3.5mm auxiliary input for connecting a personal MP3 player or other line-level source. The main inputs are on the rear panel and (for convenience of interfacing with consumer computer soundcards) these are on RCA phonos, rather than the more usual jacks and XLRs that we've come to expect. There's a further spring-clip terminal link to the unpowered speaker. A two-metre cable to link the speakers is also included in the kit.

The power switch and AC power connection are located on the rear panel of the powered-speaker housing, and the rear panel itself is made of sheet steel, with a small heatsink mounted on it to help cool the power amplifiers. The cabinets, which measure 168 x 194 x 232mm and weigh 16lbs (7.3kg) per pair, are conventional boxes, each with two rear-facing ports, and have a black satin finish with an attractive trim around the drivers. Magnetic shielding is incorporated, which means that the

Alternatives

When it comes to quality multimedia speakers, there are a few other worthy contenders, such as those by Blue Sky and Event. If you're currently working with cheap multimedia speakers, you should notice a significant improvement by moving up to any of them, and they won't break the bank.

speakers can be used in close proximity to CRT-style monitors. The frequency response is quoted as being 65Hz to 23kHz (there are no limits specified), and the graph printed on the rear panel shows a fairly brisk fall-off below 100Hz.

Impressions

Given their size, these speakers deliver a fairly punchy sound, though the 100 to 150Hz region has a slightly boxy character — no doubt due to some enthusiastic cabinet tuning being used to maximise the low-frequency extension. A bit of loose cotton wool stuffed into the ports helps to tame this, as does mounting the speakers on resilient pads or solid stands. As these are desktop speakers designed for close-up work, there's no lack of level when they're only a couple of feet from your head, and the sound stays tight unless you really go over the top. The silk tweeters deliver a creditably smooth high end, and the relatively small woofers keep the mid-range reasonably clear and controlled. While it would be unfair to compare the Media One 4as with more expensive studio monitors, they behave much better than many multimedia speakers, and with the 'cotton wool in the ports' modification the bass end tightens up significantly too. It's also worth noting that in small spaces or untreated rooms, small speakers often behave better than their full-range counterparts, as they don't excite the low-frequency room modes so strongly.

The rear panel of the right (master) speaker.

Conclusion

Ultimately, these speakers achieve what they set out to do, which is to plug the gap between typical multimedia speakers and studio monitors. If you have the space and the budget, I'd recommend that you buy something better, but if your studio lives on a computer desk and you simply want to improve on the nasty speakers that came with the computer, without spending a fortune, then the Samson Media One 4as are certainly up to the challenge. SOS

information

- £ £129 per pair including VAT.
- T Sound Technology +44 (0)1462 480000.
- E info@soundtech.co.uk
- W www.soundtech.co.uk
- W www.samsontech.com

Aphex Headpod

Headphone Amplifier

Paul White

Unhappy with what was commercially available, the Aphex boffins designed the Headpod for their own use — and they liked it so much they decided to release it.

Headphone amplifiers: boring or what? Well, you'd better hope so, because if a headphone amp attracts your interest, it probably because it is doing something wrong. A good one, on the other hand, will soon be taken for granted and you'll forget all about it. There are dozens of headphone amps on the market (if you want an alternative to the model reviewed here you could try Samson's C Que 8, for example), but because headphones come in a wide range of impedances, driving them all at an adequate level with enough headroom to avoid clipping isn't as straightforward as you might think — which means that headphone amps don't all work as well as they should.

Overview

Aphex excel at bringing something new to the party, whether it is a mic preamp, an Aural Exciter or a headphone amp. The circuitry of the Headpod that's reviewed here was originally developed for internal use by Aphex because they weren't happy with what was on offer elsewhere —and, pleased with the result, they decided to turn it into a product. They're aiming the Headpod at a wide-ranging market, covering both pro and semi-pro users, and including computer-based recording, composing or editing, sound for picture work, location recording and broadcast.

This innocuous-looking headphone amplifier is ruggedly built from heavy steel and finished in an attractive white paint finish. It has four TRS stereo outputs, and two balanced TRS inputs that allow the unit to be fed from a mono or a stereo signal. There's also a separate TRS stereo input that can be fed from a headphone socket, with a slide switch to select between the two input types. All of the jack sockets are of the tough, metal variety, and the aforementioned circuitry, which is powered from an included mains adaptor, has been designed for very low distortion, a wide frequency response and bags of headroom.

Each of the inputs has a 20kΩ impedance if used balanced, or 10kΩ unbalanced. Input levels of up to +24dBu can be accommodated, where the frequency response is an impressive 10Hz-120kHz ±1dB. The signal-to-noise ratio is typically 100dB, with crosstalk below -80dB at 1kHz or -70db at 10kHz, and there's a typical THD-plus-noise figure of better than 0.001 percent when driving 100mW of power into 25Ω.

An overall level control acts on the input signal before it is split to the four independent stereo power amplifiers that feed the headphone outputs, so as to eliminate interaction. In total, there's a maximum available gain of 35dB — so it is advisable to turn the input right down when first powering up, to avoid a nasty (and very loud) surprise!

Despite all this functionality, the Headpod is conveniently small, measuring only 5.5 x 1.9 x 4.25 inches (138 x 48 x 114mm), and weighing approximately one pound.

On Test

My first shock, on powering up this headphone amp, was the intensity of the white power LED — which is so bright that it prevents you from seeing the knob positions clearly if you're looking down on the unit... but on the plus side, at least you can use it as a surprisingly effective torch when working in the back of your rack (I'm not kidding!).

Physically there's little to criticise: the size makes it very convenient in most situations, and there's an optional mic-stand mount. Sonically, the Heapod is noticeably tighter and cleaner than my existing headphone amp, especially at high levels — and you can get a lot more volume than is good for you if you turn the wick up. I tried the Headpod with various headphone models, having a wide range of impedances, and it gave a good level for them all. The only thing I'd add is the ability to switch off one side of one output for singers who like to work with one phone off.

Conclusion

The Headpod is very quiet, and that being the case, I'm glad to say that (other than the integral searchlight) it is precisely as boring as I had hoped: it does exactly what it claims to do and it does it beautifully. In fact, I'm already taking it for granted! SOS

SOUND ON SOUND

Aphex Headpod £115

pros
- Exemplary sound quality.
- Compact size.
- Reasonably priced.

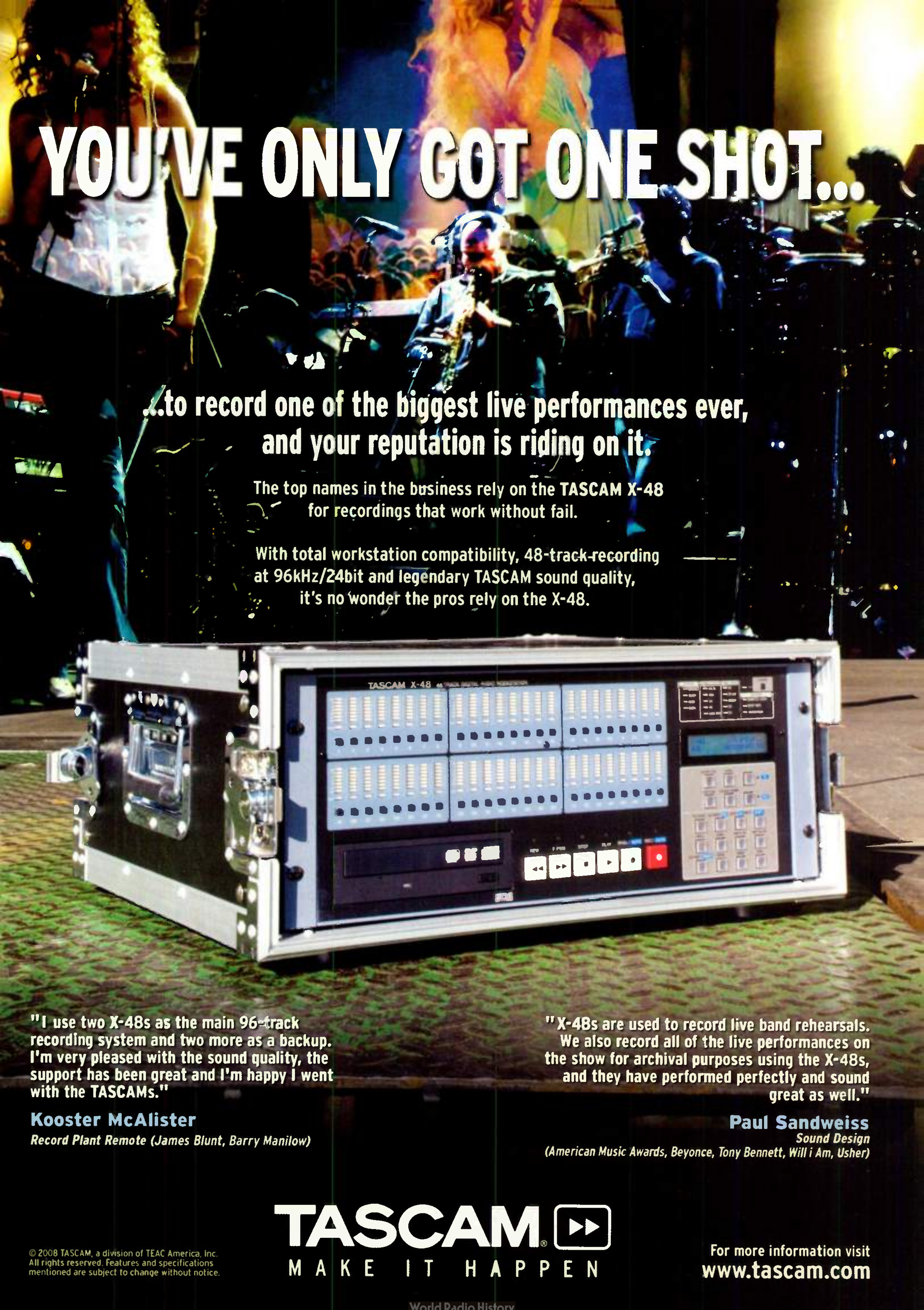
cons
- Sonically none, although I still have spots in front of my eyes from the bright LED!
- Not able to mute one side of the stereo signal.

summary

This is a very good basic headphone amp, with four independent output channels, each with its own amplifier.

information

£ £115 including VAT.
T SCV London +44 (0)20 8418 1470.
E sales@scvlondon.co.uk
W www.scvlondon.co.uk
W www.aphex.com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Kooster McAlister
Record Plant Remote (James Blunt, Barry Manilow)
"X-48s are used to record live band rehearsals. We also record all of the live performances on the show for archival purposes using the X-48s, and they have performed perfectly and sound great as well."
Paul Sandweiss
Sound Design
(American Music Awards, Beyonce, Tony Bennett, Will i Am, Usher)
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Waldorf Blofeld Digital Synthesizer

In 2004 it looked like game over for Waldorf, but the German company are back, and with a hardware synth that more than lives up to their former reputation...

Paul Nagle

Named after James Bond's pussy-fondling antagonist, the Blofeld is the first product from a revitalised Waldorf Music GmbH. Unlike Donald Pleasance's portrayal of the sinister Spectre overlord, Waldorf's Blofeld is benign and refreshingly free of cat hairs, thus should be welcome around your house, studio or secret base.

In 2004 Waldorf ceased trading and it seemed this would be the end of the line for its distinctive brand of hard-sounding, wavetable-grunging, bass-bin-shaking and often garishly-coloured synthesizers. Then rumours began to circulate that all was not lost, that something might be salvaged by the enthusiastic employees of this quirky company. Ultimately, Joachim Flor, one of Waldorf's former sales representatives, collected some people and some money to build a new company with the same name and most of the key players of the previous company. To cut to the chase, after a long gestation period, Waldorf are shipping hardware once more!

Plundering Q's Laboratory

Although judging good looks always involves a fair measure of personal taste, the Blofeld's white, slimline, metal exterior, shiny aluminium encoders and clear graphical screen won't fail to impress anyone who appreciates quality workmanship and stylish design. Internally, the Blofeld represents the ripest cherries plucked from the Microwave 2 and the Q and Micro Q series. I should stress that this synthesizer isn't simply a rehash of existing ideas; the Blofeld is a new creature with a fresh identity — it even manages to surpass the power of those classic instruments in some cases.

The Blofeld has three oscillators per voice, two of which are capable of delivering wavetable synthesis or analogue modelling, while the third is devoted to analogue emulation only. Each oscillator may be routed freely through two filters; these are selected from a diverse collection of filter types and topped off with filter panning and drive options that take sonic mangling to another level. Three fully-featured LFOs,

four snappy envelopes and Waldorf's ever-powerful modulation matrix are on hand for when you need controlled complexity in your patches. Further delights include 16-part multitimbrality, a programmable arpeggiator and more than 1000 sounds to use or to overwrite. Polyphony can be up to 25 voices, depending on DSP load. Oh, and did I mention that there are effects too? Clearly, this would be an impressive synthesizer in any class but is all the more so when you take a look the bottom line.

There are limits to Waldorf's generosity, of course, and some of them become obvious when you take a look at the rear panel. The presence of just two audio outputs is probably to be expected at this price point, but it is the single MIDI port that caused me the most immediate concern. The Blofeld has MIDI In only — there's no MIDI Out and definitely no Thru. This means that if you wish to offload any of your patches for external storage, you must use a computer and USB. Fortunately the USB port provides full MIDI I/O, so if you are a PC or Mac owner you do have options — but if your studio is hardware MIDI-only, you'll have issues.

A headphone socket and connection for the external power supply complete the rear-panel offerings — and I sincerely hope the wall-wart supplied to UK buyers is different to the one I received. I've really had my fill of perching Euro connections into wobbly shaver adaptors — I thought there was some kind of law against this sort of thing!

Getting Started

Just for a change (no, not really!), my first job prior to making any noises was to upgrade the Operating System and factory sounds. Waldorf recommend Windows 98, Windows XP (or higher) or Mac OSX 10.3.9 (or higher) as supported platforms. Upgrading the OS proved perfectly straightforward: I connected the synth to my Windows XP PC via USB, at which point the Blofeld was recognised as a generic USB Audio device. This device appeared in the MIDI port list in my sequencer program, making it a simple matter to send the appropriate MIDI file. The whole OS updating process took about 11 seconds and probably left me feeling rather ▶

SOUND ON SOUND

Waldorf Blofeld £349

pros

- The essence of Waldorf's digital synths squeezed into a slimline and handsome metal container.
- Microwave and Micro Q power you can put in your pocket!
- Up to 25 voices, effects, over 1000 patches, programmable arpeggiator, that PPG filter, deep modulation...

cons

- No MIDI out port.
- Currently incomplete Multi mode.
- Receiving patch dumps is a little hit and miss.

summary

The Blofeld represents an amazing comeback from Waldorf, who have plundered their vaults of accumulated wisdom to give us the best of their digital synths in an attractive and inexpensive package. Although the matrix method of parameter access won't be loved by everyone, it's nicely implemented, and since it keeps the price down I can't see any reason for these not to fly out of the door.

over-confident about the prospect of updating the factory sounds.

I needed to upgrade these because initial versions of the soundset reportedly suffer from drastic variations in volume and I wanted to start with a clean sheet. As it happened, this updating procedure proved rather less painless than the first one. Firstly, sending the data produced no indication that all the sounds had been correctly loaded. There was an occasional message that SysEx was being received but afterwards I could spot no obvious differences — my data load had clearly not worked. I experienced a vague sense of *déjà vu*, remembering similar issues with my Microwave XT, which at least offered the clue that it was reorganising its memory during a receive, meaning that you had to try again. Sometimes again and again, watching the damn thing throughout! Anyway, in the end I completely re-initialised all patches and reloaded from scratch, this time sending the MIDI file at a very slow tempo. Thankfully, this appeared to work, although I thought some patches still sounded a tad quiet. I can only hope that Waldorf manage to improve the process in the future, perhaps providing some kind of simple software application to do it via USB.

Something also worthy of note, if you have purchased a Blofeld but are unable to find the latest soundset, is that the printed 'Quick Start' manual points you to the wrong Waldorf web site — the correct one is documented in the full manual to be found on the supplied CD.

The Bloefeld's rather spartan rear panel — and yes, that is just a single MIDI input port.

Enter The Matrix

It was in 1996 that I first encountered a Waldorf synthesizer that employed the matrix method of parameter access. This was the powerful analogue monosynth they call the Pulse, and it used a matrix instead of a panel of dedicated knobs. The system worked pretty well, saving considerable cost by allowing just a few physical controls to operate a wide array of parameters. Time passed, and further refinements came along in the form of proper displays and continuous encoders. The Microwave 2 possessed what I still regard as the fastest, most intuitive variation on this theme.

The Blofeld's matrix is a two-tier system based around seven perky encoders, a delightfully sharp graphical display and a series of buttons and LEDs. At its simplest level, the menu system is navigated via the four buttons for oscillators, filters, modulation and so on. Where there are several options, multiple button-presses step through them, the position at all times shown by an LED. The matrix rows are labelled in a small neat font that my ageing eyes can just about cope with, and having done your button pushing, you simply grab the relevant encoder and tweak away. For most quick editing purposes, you might never need to go much deeper.

Beneath the display are two encoders that line up with items appearing on the display screen. These items don't necessarily correspond to the current selection in the matrix, which confused me occasionally; it is implemented in this way so that you can use the matrix encoders independently of the display encoders.

When exploring the menu system further, the selection dial is used to navigate through the edit pages. Knowing where you are in this 'deeper' realm of the menu system is aided by the 'page of pages' area of the display, in the top right-hand corner. You soon learn that there are lots of these pages, many of which are populated by excellent graphics, illustrating oscillator and LFO waveforms, envelope shapes and so on. At any time you can return to play mode by hitting the Play button, then use the selection dial to select patches and the encoders beneath the display to change banks and perform category searches. The latter is essential when you consider that there are eight banks of 128 patches to choose from.

The selection encoder is notched for ease of use (all other encoders are smooth in operation) and when initially clicking through the preset banks I was overwhelmed by the sheer number of patches, even though they are conveniently organised into 13 different categories. There was no way I could meaningfully audition them all and still get a review written, but I soon realised they represent the sort of sounds I've grown to love from Waldorf — hard, biting basses, mad sound effects, evolving wavetable pads, brassy leads and more. I also thought I noticed a warmer, more fuzzy character than I am familiar with, which was intriguing.

I decided to pitch in and program some patches myself from scratch, perhaps using the ever-useful randomise function to generate some starting points. The Utility menu contains patch store, initialise, randomise, and various dump options. Some of the patches thrown up by randomise were astonishing, although I concede that very few would be classed as 'normal'. Randomise gave me strange, eerie and

Note The Arpeggiator...

A programmable arpeggiator stored with each patch is something we encountered on the Microwave 2, and the Blofeld's has grown considerably in stature from those early beginnings. Needless to say, it can cope with all the standard up, down, and up/down directions, curiously lacking only a random option. There are 15 ROM patterns that use gaps to break things up, or you can make your own pattern, adding glide, shuffle, pauses, accents and more to build something truly unique. The arpeggiator uses a 'note list' — up to 16 input notes that it memorises and from which you can choose when building user patterns. As each step in a pattern can specify a random choice from this list, you should soon find a way to simulate any 'Rio'-type arpeggios you might need. There really isn't space to cover every option in glorious detail so I'll just say that layering up a few arpeggiating patches in Multi mode proved a fine way to occupy a rainy Saturday afternoon, even if I had to generate my input notes on several MIDI channels simultaneously.

downright unsettling tones, all ready to tweak and develop further if necessary. I wanted to start overwriting many of those factory electric pianos, organs, basses and soft, girly pads right away! Having concluded that every synth should have randomise, I then pondered how much nicer it would be if degrees of randomisation (or some other way of taming the results) were offered for those times when you want to generate something you can actually play.

Oscillator Action

I kicked off by exploring the oscillators, which instantly impressed me with their range of 128' to 1/2' — that's before I remember from the Microwave, but personally I prefer the Blofeld's versions, especially as there is an oscillator brilliance control that can bring back the grittier, scratchier sound if needed. Some of the tables have 'gaps' in them, and these, when swept, suggest those Microwave sounds we know and love, just somehow 'nicer'.

One feature that works differently to earlier Waldorf synths is that after selecting a new wavetable you must then play a note before the new table can be heard. Similarly, adjusting brilliance or selecting a new FM source for the oscillator appears to do nothing until, again, you play a new note. This is

"Sonically recognisable as distinctly Waldorfian in character, the Blofeld is also capable of smoother, lusher tones. It can do convincing analogue and wavetables at the same time — quite a combination!"

modulation! This is a synth that could seriously disturb any bats you have in the attic! The Blofeld includes all the wavetables from the Microwave 2, plus the two 'Alt' tables from the Q series and, although there is no provision for adding your own tables, you can have two different wavetables active at once — something not possible on the Microwave. This is mightily cool for creating complex, evolving pads, especially when you draft in some of those LFOs and envelopes to modulate each wavetable independently.

The wavetables sound slightly different to the Microwave 2's implementation; they are expanded to 16-bit and use the same high-quality blend algorithm found in the Q and Micro Q. Even the 'speech' wavetables (namely: '1-2-3-4-5' and '19-20') sounded less glitchy (and also less intelligible) than

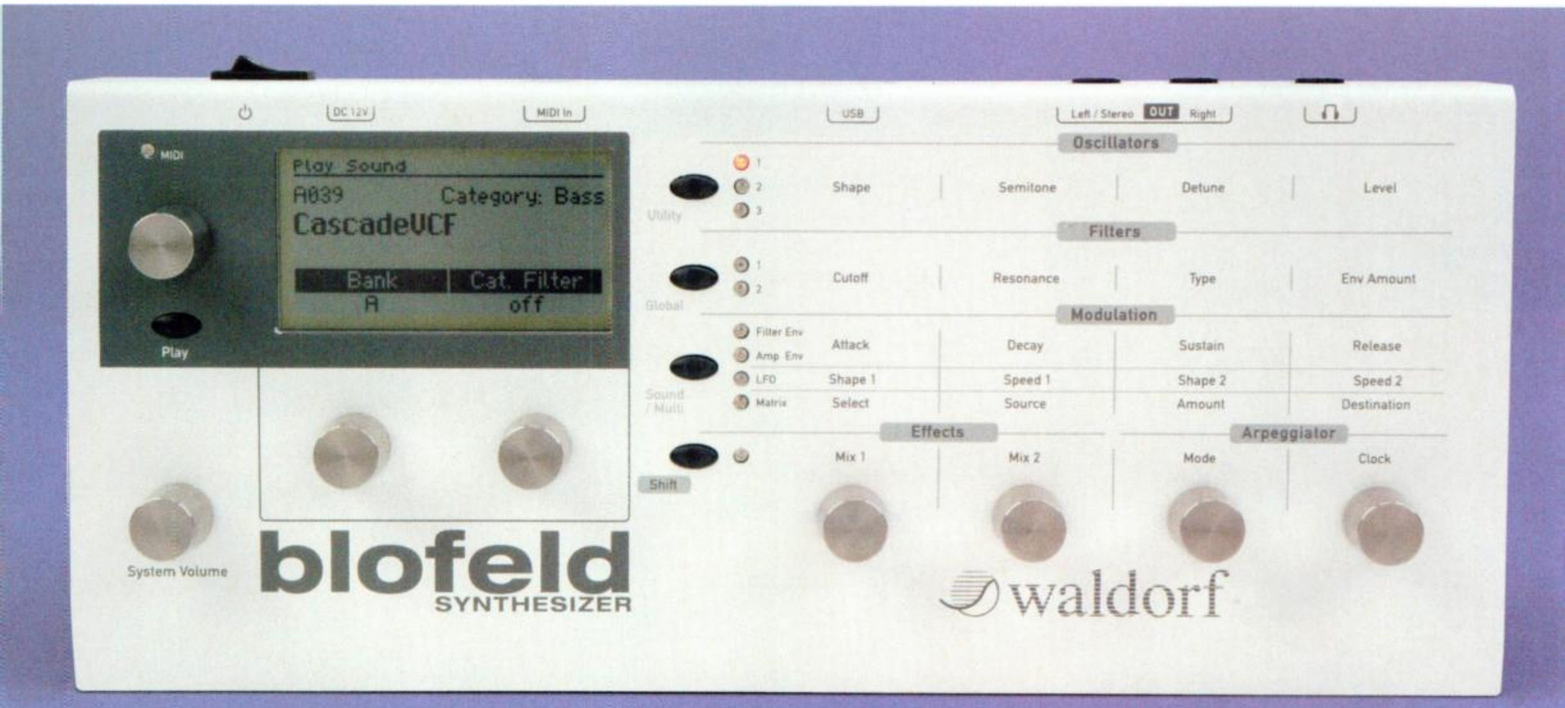
something to be aware of if you are scratching your head or trying to extract imagined earwax, as I was.

When I selected the modelled analogue waveforms, I thought I noticed something interesting occurring. One thing I never imagined I'd write about a Waldorf synth is that the oscillators, when detuned, can sound lush, swimmy and 'analogue'. But that's exactly what I thought here. Perhaps the degrees of detune are simply finer or there's some other German magic going on; either way all the expected analogue waveforms are present and correct and can be delivered by all three oscillators (only oscillators one and two can handle wavetables). Setting brilliance to 64 gives an oscillator sound very like the Micro Q, whereas 128 more closely matches the Q. In other words, analogue modelling with balls. My only wish was

Multitimbral Operation

When programming, you can save DSP power by turning off unused oscillators or filters. This could be very important if you wish to use the Blofeld multitimbrally, although this is one aspect of the synth that's incomplete. In fact, it rates just one page in the manual, where Multi mode is shown as a means of selecting patches on 16 separate parts, each part corresponding to a MIDI channel.

Multi mode is engaged via shift and the Sound/Multi button. You can then pick a part to edit by holding the Play button while turning the selection encoder. It's a bit primitive right now, and you can't even layer sounds on the same channel or do keyboard splits, for example, nor can you name and save Multi assignments. Nevertheless, each patch keeps its own Effect 1 setting and there's a mix level available to all parts for the common Effect 2. This would typically be reverb or delay and is sourced in whichever patch you pick for Part 1. I am assured that Multi mode will be refined in a forthcoming Blofeld software release.



The Blofeld's austere front-panel design belies the complexity of the synth engine beneath.

▶ for a single page to show the levels of all three oscillators, plus the noise source and ring modulator. Fortunately the MIDI spec is such that you could assign these source levels to a series of sliders on your favourite controller, which would also speed up programming considerably.

Pulse width may be adjusted either for the wavetables (in which case it sweeps the wavetable) or the square wave, where it performs as you'd expect. The oscillators can serve as audio-level modulation sources too, plus there's dirty, evil sync available between Oscillators 2 and 3. Sync even works with wavetables, rather than being restricted, as on some synths, to only the analogue waves.

For those Incredible Hulk moments, there are several unison options on tap for multiplying and detuning each voice up to six times. Or, if you prefer to show your more lyrical side, portamento and glissando are present for those expressive 'glidey' and 'steppy' sweeps between notes. Finally, and possibly the most versatile oscillator feature of all, there's the balance control, which routes each oscillator freely between the two filters.

Filter Tips

Configurable either in series or in parallel, the twin filters may be picked from an impressive list of filter types. The list includes the usual 2-pole and 4-pole low-, high- and band-pass options, plus a notch filter, positive and negative comb filters and the PPG low-pass filter, modelled on the 24dB SSM2044 chip of the PPG Wave. While it may be unfair to single out any for special praise, I'm going to do it anyway. This filter sounds fantastic. I had always wanted to combine it with the wavetables of the Microwave 2; finally I was able to do so — and I was not disappointed. Similarly, with no more than a couple of detuned sawtooth waves as a source, the PPG filter yielded some of the best pads I've heard from a Waldorf synth.

It's easy to get diverted by the number of ways you can use a filter arrangement this versatile; I found myself connecting a flanging comb filter in series before the PPG, then controlling the filter balance with velocity, so that quieter notes passed through both filters, the loudest going only through the PPG, those in between varying position according to note strength. In just a few minutes I had a patch in which I could articulate notes to rise out of a flanged mush purely by how hard I hit the keys.

As well as a selection of filter types, there are a number of drive curves that drastically affect the filter's output and tone. These include tube and pickup simulations, audio level modulation (by Oscillator 1), a rectifier, clipper, sine shaper and more. Each one imparts its own tonal changes to the selected filter, and as you crank up the drive for the first time you really begin to grasp the enormity of what is concealed in this tiny white synth. Each filter has its own panning control too, so you can move the filters around dynamically with creative use of modulation.

Modulate Your Tones

It's often at this point in a review of any Waldorf gear that I recall why their synths stand out from the crowd: very few other manufacturers let you tinker with such minute details of sound sculpting. In common with earlier products and alongside many built-in modulation routings — the (variable) source connected to oscillator pitch, pulse width and FM — the Blofeld has 16 completely free modulation slots. With these you can connect any of 29 modulation souces to most of the destinations you might want. To give you a flavour, these destinations include envelope stages, cutoff and resonance of both filters, and the level and filter balance of every sound source; they encompass oscillator and filter FM amounts, filter drive amounts and panning, LFO speeds and more. The destinations that I found absent include the myriad parameters of the arpeggiator, oscillator detune and filter envelope amount.

Various Waldorfian modifiers are also present for when you wish to perform mathematical operations on the modulation signals. So if, for example, you wanted to 'unsmooth' some modulation, perhaps to recreate the steppy wave changes of the Microwave, the 'OR' and the 'AND' modifiers are your friends.

About Effects

To finish off, where would we be without effects? The Blofeld has two sets of effects for each program: Effect 1 and Effect 2.

Effect 1 consists of either chorus, flanger, phaser, overdrive or 'triple FX', the last a slightly simplified combination of chorus, sample and hold and overdrive. Effect 2 adds delay, clocked delay and reverb to the list. With its maximum delay time of 557ms, the Blofeld can't produce delay lengths as long as the Q or even the Micro Q, but at least it offers MIDI-clockable delays. These have a range starting at a very fast 1/96 division right up to 10 bars, although goodness knows how high your bpm would need to be to achieve that! I also noticed a few audible clicks that occur when you adjust delay time or feedback during play; I mention this because the manual specifically claims it won't happen.

Finally, the reverb really doesn't sound too bad in context. There are a number of tweakable parameters that help you get more mileage from it, including diffusion, size, shape, decay and damping, with the reverb's character further adjustable using low- and high-pass filters. Overall it's a welcome, if unspectacular, inclusion.

Licensed To Thrill?

Blofeld was 007's chameleon-like adversary and an evil genius to boot, so finding holes in his fiendish schemes is no easy task. In fact, most of the shortcomings I unearthed can probably be attributed to cost saving. Admittedly, just two audio outputs isn't ideal for a multitimbral synth, and the matrix method of parameter access can never be as fast as a panel full of knobs, but I think this latter issue is minimised due to the logical way the matrix has been implemented. Given the Blofeld's size, there's a hell of a lot packed in here yet at no time has either build or sound quality been compromised. As you'd expect from Waldorf, extensive MIDI control is available, so you can choose your favourite remote controller and program it appropriately, at which point things start to look much more accessible.

Loss of MIDI ports I found slightly harder to forgive, but I guess most of us probably have some USB devices kicking around, so connecting up the Blofeld for the sake of saving our patches shouldn't be an impossibility, even if it is occasionally inconvenient. If, in future, Waldorf find a way to offer USB audio functionality as well as MIDI, the Blofeld could make an irrisistible companion to a laptop-based setup.

The current Multi mode is unfinished (see the 'Multitimbral Operation' box earlier in this article) but it is, at least, usable, even if you can't save individual assignments yet. With over 1000 patches supplied, you have a huge amount of exploring to keep you busy until an update percolates through.

Sonically, although recognisable as distinctly Waldorfian in character, the Blofeld is also capable of smoother, lusher tones. It can do convincing analogue and wavetables at the same time — quite a combination! In fact, the ability to mix these forms of synthesis freely and route the results through some seriously good filters (and, indeed, some seriously wacky ones) already puts the Blofeld beyond most hardware synthesizers made today, let alone at this price point. With that extensive modulation matrix, versatile arpeggiator and built in effects too, I have to admit it's difficult to do anything but gush. If any miniature super-villain is destined to rule the world, Blofeld gets my vote! SOS

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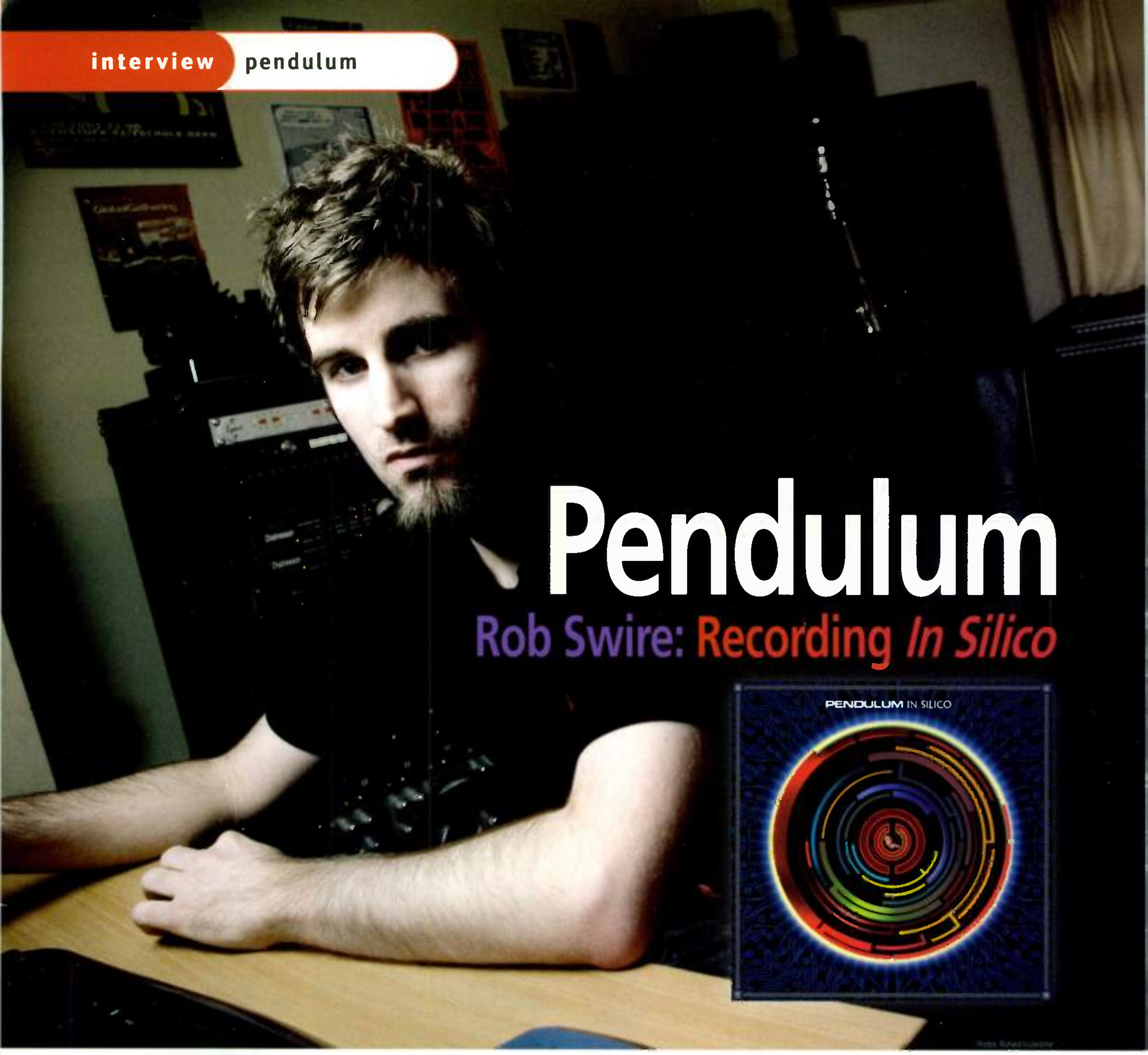


Pendulum

Rob Swire: Recording *In Silico*

From their roots as a drum & bass act, Pendulum have transformed themselves into a dance-rock powerhouse.

Sam Inglis

Drum & bass has proved one of the most enduring genres of electronic music, despite — or perhaps because of — remaining largely outside the mainstream music industry. If anyone can drag it above ground, it's Pendulum, whose debut long player *Hold Your Colour* is widely cited as the best-selling drum & bass album of all time.

That album showed a band developing at a furious pace, and in the three years since its release, things have continued to move on. Pendulum have signed to major label Warner Brothers, recorded and scrapped a follow-up to *Hold Your Colour*, transformed themselves into a powerful live band, and at the time of writing, are finally poised to release their second album. *In Silico* has the potential to become one of those elusive dance-rock crossover hits, blending electronica with metal guitars and live drums to create a sound that is brutal yet thrilling, complex yet immediate. Above all, it's a proper album that's satisfying to listen to from beginning to end. How have Pendulum made the transition from DJs and remixers to songwriters and album artists?

"It was hard," admits Rob Swire, "because the tracks which we did to play in clubs in the beginning ended up with a lot of people listening to them in their cars and shit, which we didn't really take into mind. We were trying not to really shift from that too much, I just wanted to include more of the band side of things — the acoustic drums and the vocals and the guitars. We knew we wanted some tracks that weren't just the drum & bass one minute 30 intro, 64 bars of drop, 32 bars of breakdown, 64 bars of drop, 32 outro, so we did have in mind that we wanted to try to put together something with proper lyrics, proper vocals, proper parts and everything. It was a bit harder than I think we had in mind."

Swire is the brains behind Pendulum's studio operations, and as well as writing and producing all the tracks on the album, he's also become the band's main vocalist. The decision to have a single lead vocalist on all the tracks was, again, prompted by the desire to make a consistent-sounding album. "We wanted to make sure of that, unlike the last one where we approached it more producer-like and got anyone that we could." This, however, added to his burden as producer. "Since I was doing the vocals, it's a bit harder to keep the objectivity on the engineer's side, because I don't think anyone likes hearing themself back, and when you're trying to mix that, it adds a whole different, annoying dimension to it."

Pendulum Remixing Pendulum

The process of developing the album material reflected both Pendulum's reinvention as a live band and their love of remixing. "We basically put together all the tracks as very shitty demos, because I had this idea that it'd be quite good, since we love doing remixes, to have these demos of the songs, go and record them as a band so we get the parts for them, and then come back with the parts and tear them apart like we were doing a remix of someone. So we recorded the drums straight after we'd done those demo tracks, so we'd have something to play along to. They're mostly complete performances, apart from tracks where we didn't know what we were doing and we just recorded as many different things as we could."

Thereafter, says Rob, "It's pretty much constantly back and forth [*between him and the band*] until we get approval from everyone. We try not to sample as much as we can. We usually just start with us tearing some sound apart and then work it from there. There's a breakbeat in the second track that we used because we needed an old hardcore-style vibe, but we were going into this album saying 'If we need a breakbeat, we'll record it. If we need a high-pitched, happy hardcore-style vocal, we'll record that, and try to do it ourselves.'

"A lot of what we were trying to do with this album was listening back to bands, new shit and old stuff, and hooking into a particular sound that gave the song a particular vibe or style, and then being like: that, but with our usual stuff. We wanted to hear that combination, and we'd go out and try to recreate it."

As Bonham As Possible

The emphasis on creating fresh recordings was particularly important with the drums, which were tracked by engineering legend Dave Bascombe at Olympic Studios. The point, as Rob explains, was to give Pendulum access to "something that no-one else has, basically. If you're using breaks, it's probably a break ▶

Little Boxes On The Hillside

Pendulum's studio occupies one room of a nondescript house in the London suburbs. There's not much by way of acoustic treatment, but as Rob Swire explains, that's partly because of their unique approach to house-hunting. "We actually looked at quite a few houses before settling on this one. We took a speaker with us in the car, and set the speaker up in the room and played test tones. The estate agent was trying to sell us on kitchens, and we were like, 'We don't care, dude! — whoooooooooooo [*he does an uncanny impression of a sine sweep*]'. For some reason, sitting right here it did actually work the best. I think that can only be because of the cupboard, which is full of tiny wooden squares that go all the way to the back. I don't know what the fuck they're for, but somehow I think it evened up the bass!"

Though there's a pair of Mackie speakers stacked on the floor, and a subwoofer under the window seat for testing mixes intended for club PAs, the main playback system is a pair of Mitchell & Todd active monitors. "I got introduced to them by Simon Askew, the Maida Vale guy we stole to do our live sound. He was using a pair, it was the model below this, and the PMCs. I heard these and I said 'What the fuck is that, it's like everything you want in a monitor!' It's Dynaudio drivers, but the Mitchell & Todd guys designed the enclosures and the amps for them. They're really flat. The guys actually came down here and set them up themselves. Something was a bit weird with the tweeter and he came down and replaced it, so they're good like that."

Pendulum's studio occupies one room in a suburban house. The acoustically treated area to the left of the window is where Rob Swire's vocals are recorded, and the pink cupboard visible at the right is "full of tiny wooden squares that go all the way to the back", somehow evening up the bass response!

RECORDING *IN SILICO*

All the vintage synths in the Pendulum studio are still in showroom condition, notably this Sequential Prophet 10.

that everyone's used millions of times, and we wanted to go in and say 'Well, no-one's touched these. We've got fresh breaks, it's our album.'

"It ended up being quite flexible, because we got him to do dynamic multisamples of each kit that we got [*drummer Paul Kodish*] to play, and then loaded that into Kontakt, and had both the live recorded audio and the audio from Kontakt coming out of about 32 different channels into 32 groups, and we just mixed them from the groups. So any time if we played something and we were changing the song, at any point we could just switch the live drums out and play it in Kontakt. It was quite cool mixing through the groups, because you couldn't even tell that it was switched. Literally, it just sounded like the same take.

"For the live drums, we went as John Bonham as possible, with the big 26-inch bass drum and Ludwig Black Beauty snares. We found some raw John Bonham drums and decided they were the easiest in the world to have against electronic stuff, but the sound was a bit harder to replicate than we thought."

Rob has a theory as to why the Led Zeppelin drum sound works so well in an electronic context: "I think it's because of how huge the sound is, without having to rely on the clicky transients that generally you can't really put into a dance music mix. Like, if you listen to a Nickelback track and listen to the kick, there's no way that's going to fit into any dance record ever. It's too low, for one, and there's too much pop, there's too much click. Something like the Bonham drums, the transients are all so soft, but the sound is still so big, it's really easy to use."

The live drums were frequently layered with samples after the fact. "We pretty much just have a folder of as many kicks as we can find. If we're stuck for something we go through and raid that, or start combining things, or tearing bits of the samples apart and using different parts of them."

A Warm Reception

Pendulum's recordings make extensive use of both vintage analogue synths and software instruments, neither of which are renowned for their reliability on stage. Rob Swire's solution? "For the live stuff we just sample everything into Kontakt and load it up onto Muse Receptors. They're the only things that are stable in a live environment, because if those things crash, two seconds and they're back up. With a laptop running Kore or something, if one thing goes wrong you're fucked.

"Since I've found an FBI disk-cloning tool that backs up the hard drives, I love them. Before that, when you had to send the drives to Muse to get backed up, they were probably shaving five years off my life."

Soft Guitars

Guitarist Perry ap Gwynedd has seen his role in the band grow since *Hold Your Colour*, and the instrument is now central to Pendulum's sound. On the new album, the guitar sounds were created using IK Multimedia's Amplitube 2 and TC Electronic's TC Thirty plug-in. "There's a lot of guitars in there, and our guitarist was a bit pissed that we wouldn't record an amp, to begin with. We wanted to keep it software-based, just because you don't really know what sort of sound you're going to end up with. For what it is, even including a guitar in the first place is going to throw our fans off like nothing else, so for us to have that sort of last-minute 'That doesn't sound right, let's use this...' We just use them as walls of sound alongside the synths. They don't tend to feature by themselves too much at any point. With one of the tracks we tried to do a guitar solo as Aphex Twin might have done it, which I liked the idea of. The brilliant thing

Rob Swire has high praise for his Mitchell & Todd monitors.

anything else that does it so well."

I was curious to see Rob making extensive use of the Gforce ImpOSCar plug-in, given that he has a real OSCar behind him. Does the emulation not sound the same as the original? "Not at all, and that's why I like it. The OSCar is possibly the dirtiest, most depraved-sounding synth ever — it's fucking gritty — and the ImpOSCar doesn't sound like that at all. It's clean and trance-like, which is probably why I like it for leads. I find the actual OSCar is shit at leads and good at bass."

Other hardware in the studio includes an Alesis Andromeda analogue polysynth, of which Rob says "I love how much you can do with it. It's like the Reaktor of analogue. It is a bit of a headfuck, with that interface. Leaning over that small [*display*] thing trying to dial in plastic knobs that seem to go a bit dodgy over time isn't really ideal. It splits its messages over multiple NRPNs, I think, so if you use the ribbon controller you can see about four different controllers coming in.

"For pads we usually use the Andromeda or try to layer something with the Prophet. With the Prophet there's a sort of *Blade Runner*-esque thing you can't really get from anything else. I use a lot of weird little freeware shit. [*Krakli Software's*] Cygnus is the best thing ever for spacey pads. It's like Absynth but just made for those sort of sounds in the background. If something that looks good comes out I just buy it!"

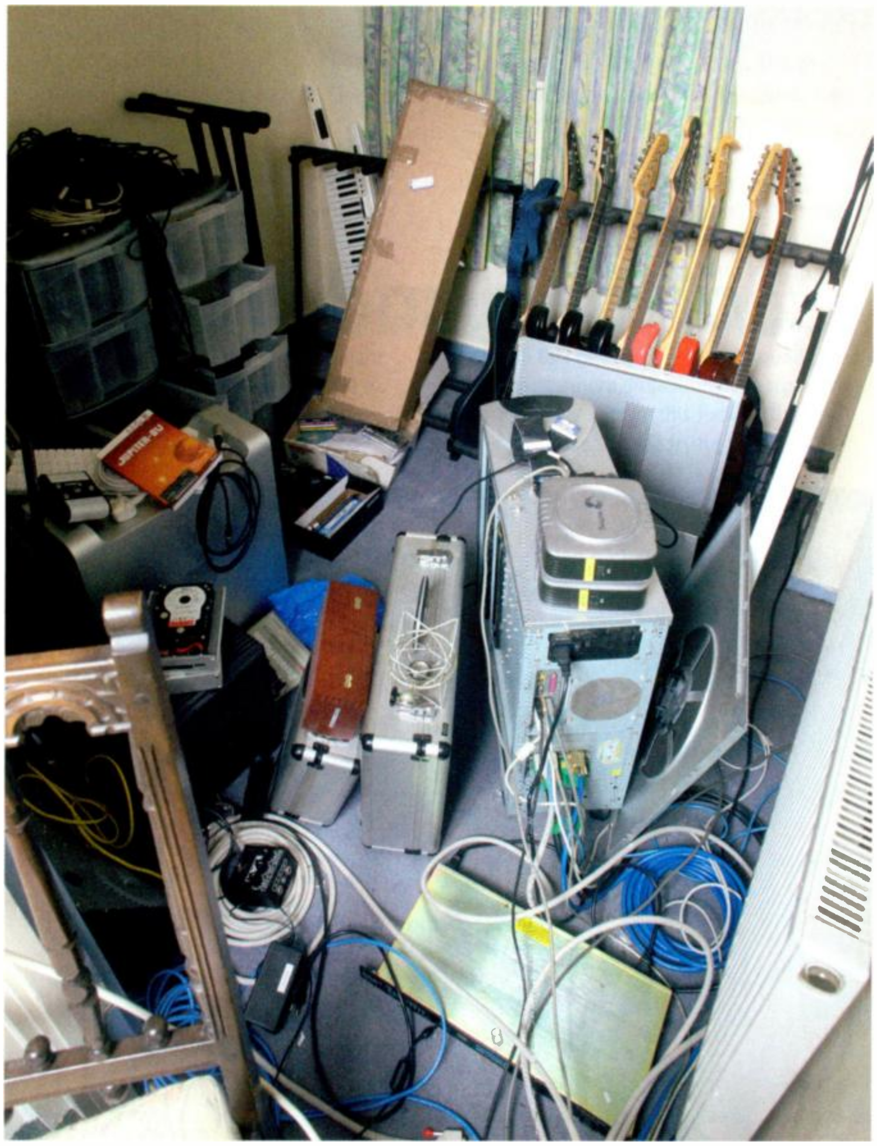
Digital Heart

The heart of Rob's studio is Steinberg's Nuendo 4 DAW ("I switched over to this from Logic, I think when Nuendo 1 came out. The layout of it feels a lot better than Logic did — I hated the Environment"), running on a Windows PC. "Macs don't work all the time like everyone claims! The software available... I wouldn't say it's better than what the Mac has, especially since the Mac has Metasynth, but once you get stuck into an OS, with the amount of plug-ins we use that are just PC, it'd be too hard to change over."

The PC's own processing power is augmented by TC Powercore and Focusrite Liquid Mix DSP units. Does Rob find much use for these? "The Powercore, definitely, mainly for the VSS3. I can't find a better reverb than that without buying something stupid like a Lexicon 480L. I find it does the job, and it doesn't tax the processor. There's usually quite a bit of stuff going to the VSS: drums, just to fill out snares, or pads or vocals. Little things like trying to emulate the Killers' Elvis effect."

There's also a small but well-used selection of outboard processing gear, which is integrated into Nuendo "with the [*Lynx*] Aurora [*interface*] and the External Effects thing in Nuendo, although Nuendo has a bug where it keeps on trying to reassess the delay compensation, and it's really fucking annoying. So as soon as we find something we like we print it to disk and get the settings if we need to."

There's also a Neve 8816 rackmounting mixer, which is used purely for mixing line-level sources when required, as Rob is not a convert to the idea of analogue summing mixers. "I just don't buy it. I've heard examples, people have posted up examples around the 'net, and every time they do it, they go 'Listen to how much better that sounds when it's summed,' and I've always preferred the digital mix. So I think I'm just going to stick with that."



The small room adjacent to the studio, which serves as a machine room-cum-junk store.

Pendulum Live

Pendulum's decision to remodel themselves as a live act has as much to do with the changing face of the music business as with artistic fulfilment, and it's a decision that seems to be paying off. On the day this interview takes place, the band are jetting off to their native Australia for the first leg of a punishing summer schedule that encompasses virtually every UK festival from Download to Creamfields, as well as numerous headline dates of their own. "The way the music industry is headed, it's a lot more important on the live side than it seems to be in the studio, which is a big shame," admits Rob Swire. "There's not the perfectionist, tweak-one-synth-for-a-month element with the live stuff; you've got to accept that some things aren't going to be perfect and move on, that's the nature of live. It's going to be different every night, and that's the good side to it, I think."

Meanwhile, *In Silico* itself was released on May 12th. It's been a long time coming, perhaps thanks to Rob Swire's perfectionist tweak-one-synth-for-a-month element, but the wait has definitely been worthwhile. "The label were saying 'When's it going to be finished?'" he laughs. "We're like, 'When it's good.'" If only more artists adopted that philosophy... SOS

IJData BodiLizer

Acoustic Imaging Plug-in

When it comes to recording acoustic instruments you've got two choices: pickup or microphone. Recent hardware developments have set out to overcome the shortcomings of piezo pickups by using DSP to impose the 'acoustic image' of a particular guitar (essentially a multi-thousand-band EQ filter setting) on the output of an undersaddle pickup.

The filter setting is developed by simultaneously recording the outputs of a pickup and a microphone and analysing the result to create the filter that can be applied, using DSP, to the output of the pickup to make it sound like the output from the microphone. Since the output of the microphone is delayed with reference to that of the pickup (by virtue of the fact that it takes soundwaves approximately one millisecond to travel one foot from instrument to microphone) the process comes with its own, built-in processing time.

BodiLizer puts similar ideas into VST plug-in form for the PC. Unlike other incarnations of this technology, it covers acoustic jazz, steel- and nylon-strung guitars, a mandolin, a violin and a Norwegian Hardanger fiddle, thus giving you a fair palette of images (stereo or mono) to use with your chosen instrument. Although several of the chosen steel-strung guitars are not exactly the greatest-sounding guitars in the universe, they do have very distinctive characters, and the level of control that BodiLizer gives you more than makes up for their sonic shortcomings.

The rather charming control panel, as well as enabling you to select the image you want, also carries eight virtual rotary controls that allow you to precisely tailor BodiLizer's effect on the track. Width increases the apparent width of the signal; Low allows you to emphasise lower frequencies; Size gives the impression of an increase in body size by moving the filter resonances towards lower frequencies; Tilt simply tilts the frequency spectrum upwards to give a brighter overall sound; Hole and Body set the level of the chosen instrument's soundhole and body resonances respectively; Dyn determines how much filtering is applied overall and, finally, Gain controls the overall gain in BodiLizer. There's also a Mono switch that is used when only a mono input is present.

In use, I loved it! The amount of tweaking that you can do and the range of available images means that you can get really creative — for example, just try applying the Hardanger fiddle image to a DI'd acoustic guitar... great fun. If you record your acoustic guitar with a pickup, buying this plug-in is pretty much a no-brainer, and there's even a demo download available on the IJData website if you want to try before you buy.

In an ideal world I'd really want BodiLizer to allow me to make images of my own instruments, because no-one, apart from me, is going to want an image of a 1920's B&M Lyrachord banjo-mandolin. Sadly, due to current levels of software piracy, it seems unlikely that IJData (being a one-man company) will be able to justify the investment in time and money needed to develop BodiLizer further. That, my friends, is a real shame — all the more so given that the 40 Euro download price hardly breaks the bank. *Bob Thomas*

SUMMARY

BodiLizer is a surprisingly effective VST plug-in for Windows that uses filters to make an acoustic instrument recorded with a pickup sound like an acoustic instrument recorded with a mic. It's easy to use and very cost-effective. Thoroughly recommended.
www.ijdata.com

TIPS & TRICKS YouLube

If your guitar is set up and strung correctly and you still have tuning problems (particularly on a model fitted with a vibrato unit) friction at the nut and string saddles is the most likely cause.

Of course, there are plenty of commercial products out there, especially in the US, that can be used to address this problem, and I've also discussed some DIY versions in these pages in the past (*SOS* March 2007). I've recently discovered a better recipe — and for a few pounds you can make enough lube to last you and your friends a lifetime. The ingredients are Vaseline (available from any pharmacist) fine graphite powder (which is sold for lock lubrication — I found several suppliers on eBay) and PTFE spray lubricant (which you can get from most machine supply companies, including Screwfix in the UK).

The exact ratio of the ingredients doesn't seem to be vital, and if you start with a small jar of Vaseline and a heaped teaspoonful of graphite powder you won't go far wrong. Stir these together in a suitable container and then spray in a teaspoonful or so of the PTFE spray and keep stirring until you get a creamy consistency. It is important that you put the graphite in before the PTFE because the PTFE spray and the Vaseline prove very difficult to mix on their own. The resulting black goo can be put in a jar, ready for manual application, or you can store it in one of those syringes used for filling inkjet cartridges (without the needle).

You only need the tiniest amount of this lubricant to fix most string-friction problems, so having wiped it onto the bridge saddles using a cotton bud, you then wipe most of it off again. You can also apply it to nut slots using small cocktail sticks, although disposable tooth-flossing gizmos (available from your local pharmacist) are better, because they will let you work the lubricant right down into the nut slots before fitting the strings. Again, wipe off any excess with a tissue or rag before fitting the strings and you're done. Not only does lubricating the friction points on your guitar reduce tuning problems but — importantly — it also helps reduce string breakage. *Paul White*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Apogee Ensemble

Audio Interface For Mac

High-end digital converter manufacturers Apogee are Apple's audio partner of choice, and their latest Firewire interface is aimed squarely at Mac users with a taste for quality.

Mark Wherry

It's hard to believe that it's been over two years since Apogee first announced their new line of products aimed at desktop musicians and audio engineers. While Apogee were already well known in the industry and acclaimed for high-quality converters, master clocks, and the UV22 dithering algorithm, the company had arguably only dipped their toes into the world of computer-based recording, with products like MiniMe and various Firewire or Pro Tools expansion card options. This changed at the 2006 NAMM show, when Apogee announced Symphony, a PCI-based system for handling a large amount of I/O with low latency via the company's AD16X, DA16X and Rosetta-series converters, and Ensemble, Apogee's entry into the Firewire audio interface market.

We looked at Symphony in last September's *SOS*, and although it might seem a little late in the day to be looking at the Ensemble, Apogee have actually been consistently improving the product since its original release (which included beta drivers) last year. For example, they've released a firmware update that allows the Ensemble to work as a stand-alone converter when not attached to a computer. The Ensemble's drivers have also been regularly updated, with the current release from last December (on which this review is based) being fully compatible with both Leopard (Mac OS 10.5) and Tiger (10.4).

Think Apogee

The Ensemble is designed to work exclusively with Mac computers, and, as such, has a brushed-metal appearance that sits quite happily alongside a Mac Pro, Cinema Display, iMac, or MacBook Pro. The 1U enclosure is about the same size as an AD16X or DA16X converter, and although this is a really minor point, I wish Apogee had made the rack ears removable (as with other interfaces such as the RME Fireface 800 or Prism Orpheus). This would have made the unit look a little more at home on your desk, if it's not in a rack, and stop it from snagging on a bag when being packed up for mobile usage.

Speaking of mobile usage, while the Ensemble has a built-in power supply with a standard IEC connection, last October Apogee introduced the Ensemble Mobile, which is functionally identical to the regular Ensemble in all respects, except that it lacks the built-in AC power supply. Instead, the Mobile features an XLR4 port so that it can easily be connected to a DC power source, and is conveniently supplied with an external AC to DC power supply having a regular IEC connection for when you're not in the field.

In terms of audio, the Ensemble has eight analogue inputs, and the first four can accept either a line- or mic-level signal, thanks to the built-in preamps. All four inputs have both XLR and high-Z (high impedance) TRS connectors, although the high-Z inputs for the first two channels are usefully located on the front of the Ensemble, which makes room for the additional send and return insert TRS connectors Apogee have provided for the first two channels on the back. The remaining four inputs (channels 5-8) support line-level signals via TRS connectors, and an

SOUND ON SOUND

Apogee Ensemble £1404

pros

- Good-quality audio converters and preamps, as you would expect.
- Integrates perfectly with your Mac.
- Built-in monitoring capabilities that are surround capable.

cons

- Not compatible with Windows, which could frustrate Boot Camp users, or indeed PC users.
- No built-in MIDI interface.
- Some driver issues persist.

summary

The Ensemble brings Apogee's hardware and software technologies to an easy-to-use, high-quality Firewire audio interface. If its feature set is what you're looking for, Ensemble is an ideal centrepiece for any Mac-based desktop or mobile studio, especially if you're using Logic Pro.

additional eight TRS connectors provide eight analogue outputs.

For digital connectivity the Ensemble has input and output TOSlink optical connectors for either ADAT or S/PDIF input and output, in addition to regular coaxial S/PDIF, and there are also word-clock BNC connectors, along with a push switch so that the Ensemble can optionally self-terminate an incoming word-clock signal. Technically, the Ensemble can work at sample rates of up to 192kHz (and even offers hardware sample-rate conversion), but only the analogue and coaxial I/O are supported once you work with rates above 96kHz. S/MUX mode is supported for ADAT operation with the TOSlink connector, which can provide four ins and four outs at 88.2/96kHz.

A Clear Front

The front panel is clearly laid out and features 10 plasma-style meters. The first eight of these can show either the level of the analogue inputs or outputs, while the last two are labelled D1 and D2, and show the presence of signal in either the coaxial or optical digital I/O respectively. Two white LEDs to the right of the meters indicate whether the metering represents either the audio input or output of the interface, and it's also possible for the meters to be disabled if they should become distracting.

Also featured on the front panel are two rotary encoders, one for controlling input levels and the other for output, and each show level via a ring of bright white LEDs. The input encoder allows you to set the gain for the mic preamp on the currently selected input, and you can set the current input by pressing the input encoder, which toggles through a set of blue LEDs to the right that indicates the selected input. There's also a corresponding set of red LEDs to show whether phantom power is enabled on each of the four mic pres, although the power itself needs to be turned on or off remotely from the host computer.

The output encoder sets the level for the main output, and a really nice touch is that the Ensemble's software (see the 'Maestro' box) lets you configure whether the Main output is stereo, 5.1, or 7.1. For stereo, the output encoder controls the levels of the first two analogue outputs, whereas in 5.1 or 7.1 mode the encoder sets the level of either outputs 1-6 or 1-8 respectively. A 'line out' mode is also available, which effectively disables the main output volume functionality and makes the eight analogue outputs behave as independent line outputs.

There are also two quarter-inch headphone jacks on the front panel, and, as with the input encoder, pressing the output encoder allows you to set whether you're adjusting the level for the Main output, or one of the headphone outputs (as indicated by another set of blue LEDs). What's especially neat about the headphone outputs is that they're completely independent, both in terms of level and in terms of which pair of the Ensemble's outputs they are assigned. For example, headphone port one could be set to the output of analogue outs 1 and 2, while port two is set to 3 and 4. This makes it very easy to create headphone mixes without the need for any additional hardware.

Pressing and holding the output encoder for a few moments will mute the Main and headphone outputs, and the LED to represent whichever output was selected will start flashing to indicate the mute status. Pressing and holding the encoder again will release the mute. ▶

The rear panel sports sockets for the Ensemble's eight analogue inputs and outputs, along with connections for ADAT and S/PDIF, word clock and Firewire.

While this is a somewhat pedantic point in respect of what is a pretty well-designed front panel (especially the colour-coding of the LEDs), I really wish it was possible to configure the Main output encoder to mute on a simple press, rather than a simple press switching the selected output for level adjustment. Although this behaviour keeps consistency with the input encoder, I found the press-and-hold action for mute annoying, because I would use mute all the time, whereas selection of headphone outputs was something I rarely (if ever) needed to do. Some people might find the selection behaviour more useful (if they use headphones), but I wish Apogee would provide a quick mute option for those who might prefer it.

Working With The Ensemble

To get started, simply install the Ensemble's software — although, as always, it's probably a good idea to make sure you download the latest version from Apogee's web site, rather than using any supplied versions. You can then plug the Ensemble into your Mac via one of the two Firewire ports found on the back panel. The second port is handy if you want to daisy chain other Firewire devices.

The Ensemble goes into standby mode when it's connected to a power source, and, perhaps unsurprisingly, pressing the power button on the front panel powers up the unit for operation. A status LED on the front of the Ensemble lights up in blue if the device is successfully connected to your computer, or green if it's working in stand-alone mode without a computer connection. The status LED also has the potential to illuminate in red if you're really unlucky, and this can happen if you plug the Ensemble into your computer and a successful connection can't be established. In this case your Mac will probably kernel panic and the Ensemble will go into stand-alone mode.

I found that having a kernel panic when connecting the Ensemble was not a completely uncommon occurrence with my MacBook Pro running Mac OS 10.5.2, although I admittedly don't restart the computer that often and usually have quite a number of applications open. Apogee's helpful manual shows the Ensemble being connected to a Mac prior to being powered up, although it doesn't seem to explicitly warn against doing this the other way around, which is what I would normally do, as I'd leave the Ensemble in stand-alone mode when not connected. According to Apogee, the majority of Ensemble users don't experience this issue, but Apogee are aware of it and are working with Apple to find a resolution.

Once successfully connected, you can set the Ensemble to be the default audio input and output for your Mac in either Audio MIDI Setup or the Sound System Preferences Panel. When you adjust the volume using the Mac keyboard or the menu bar extra icon, you're then actually adjusting the Main volume on the Ensemble, and the Ensemble's output-level LED ring on the main output stays in sync with your Mac.

Like Symphony, the Ensemble is also supplied with Apogee's Maestro software, which is used to configure the onboard routing and settings. Although the Ensemble works with any Core Audio-compatible software on Mac OS X, Logic Pro users are especially lucky as Logic incorporates an Apogee Control Panel (accessible from the Options / Audio menu) that allows you to adjust the Ensemble's settings within the application. While this is functionally no different to using Maestro in conjunction with any audio application, the neat thing is that the settings for the Apogee Control Panel in Logic are stored with your Project. This means that you can easily recall whatever mic-pre settings you had for a given Project, by simply clicking the 'Recall

Maestro, If You Please...

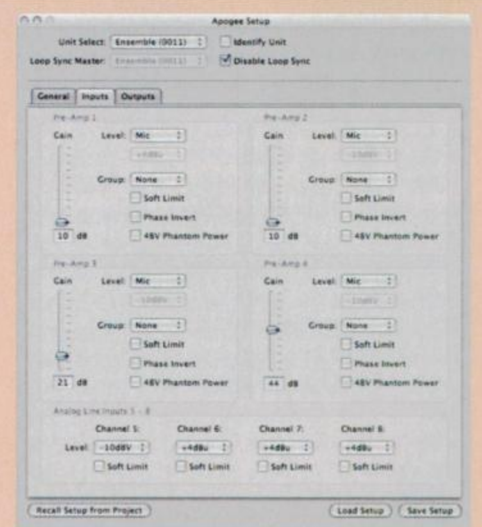
The supplied Maestro software serves two purposes: firstly, to provide hardware mixing and routing capabilities so that hardware inputs can be routed directly to hardware outputs; and secondly, to allow you to access and configure all of the Ensemble's internal settings.

The mixing and routing aspects of Maestro were covered in the Symphony review, so it's worth referring to that article if you're unfamiliar with this functionality. One thing to bear in mind, though, is that while Symphony and the Ensemble both make use of the Maestro software, certain Symphony features, such as the V-Bus routing, are not available on the Ensemble. However, this isn't a big deal, since most native audio applications, such as Logic Pro 8 and Cubase/Nuendo 4.1, are now able to support this style of routing (where audio buses can be used as the inputs to audio tracks) without hardware assistance.

One routing improvement that's been made since our last review is that it's now possible to route one hardware input to multiple software inputs, and, similarly but perhaps more usefully, it's also now possible to route one software output to multiple hardware outputs. This is great if you want to create different headphone mixes, for example.

The Maestro Control window allows you to configure all aspects of the Ensemble's hardware setup, and this is where you have full control over the built-in preamps. Each preamp has a gain control, and a particularly useful feature is the ability to assign each preamp to one of two groups, enabling gain adjustments to be linked between preamps assigned to the same group. There are also phase-invert and 48v phantom power toggles, and you can also set whether to disable a preamp and use line level for an input instead.

Each input and output level can be set to use a +4 or -10dB line-level signal (instead of a preamp on the first four inputs) and Apogee also provide SoftLimit, an analogue limiter that allows an extra 4dB of headroom, which can be enabled on any of the eight analogue inputs.



Logic Pro includes an Apogee Control Panel where you can access and store the Ensemble's settings directly from your Logic Project.

Setup from Project' button in the Control Panel once you've loaded up a previous Project.

In terms of audio quality, you would expect the Ensemble to be of a high standard, and it certainly doesn't disappoint. If you're looking for some unquantifiable adjectives, I would say the converters maintain a good image, the bass frequencies sit well, and there's clarity in the high end. While the difference was admittedly subtle compared to my trusty Fireface 800, the Ensemble seemed to have a little more depth, with fractionally more space at the top.

The only real problem I encountered when testing the Ensemble was when I decided to see if the record offset latency was handled correctly. On connecting one of the Ensemble's analogue outputs to one of the analogue inputs and recording a test signal (a click), I noticed that the input signal would be placed as if it was recorded before it was technically played. Obviously, this is physically impossible, so I contacted Apogee's technical support to see what was going on. Apogee's Director of Technical Services, Roger Robindoré, replied that "Apogee and Apple are aware of the issue, and have been working together to resolve it."

The Highest Point?

The competition in the Firewire audio interface market is fierce these days, no matter what your budget, and the most obvious competition for the Ensemble is perhaps RME's Fireface 800, Metric Halo's Mobile I/O and Prism's Orpheus. The Fireface 800 is slightly cheaper, but offers 10 analogue inputs and outputs, along with a second ADAT input and output, while the Orpheus is basically Prism's answer to the Ensemble, but with a price tag equivalent to more than the costs of two Ensembles. Finally, Metric Halo's Mobile I/O 2882 is a bit cheaper, but is the oldest of the interfaces and doesn't support 176.4/192kHz sample rates. A version with onboard DSP effects is also available for roughly the same money.

One area where the Fireface and Orpheus have a distinct advantage over the Ensemble is that they also offer Windows support, which could be important if you run Windows on your Mac via Boot Camp, or have additional Windows-based computers to which you'd like to attach a Firewire audio interface on occasion. However, if you exclusively use Macs, this obviously won't be an issue.

A second point on which the Fireface and Orpheus score is that both offer a MIDI input and output port as well. Again, this might not be important to you, and Apogee's focus is clearly on providing best-in-class audio functionality; but it does seem a shame that for mobile use you may need to pack a MIDI interface alongside the Ensemble.

Ultimately, though, despite a few remaining driver issues, Apogee have done a great job with the Ensemble, incorporating high-quality audio components into a Firewire interface that's easy to use, integrates well with Mac OS X (and Logic Pro), and offers flexible routing via the Maestro software. Although there are cheaper alternatives, the Ensemble certainly isn't overpriced, and I think Apogee have found a comfortable price point for the features and level of quality being provided. SOS

information

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BIAS Master Perfection Suite

Mastering & Mixing Plug-ins For Mac OS & Windows

BIAS's cross-platform plug-in bundle offers some new thinking on dynamics, EQ and pitch correction.

Paul White

The BIAS Master Perfection Suite of plug-ins, which also comes as part of the BIAS Peak Pro XT package, comprises key mastering tools such as multi-band compression with limiting and parametric EQ, but there are other tools that can be used both for tracking and mastering. These include a very flexible expander/gate, a pitch-correction system that can also be used to make voices and other sounds deliberately weird, and a fingerprint EQ system that can be used to match the spectrum of one piece of audio to that of another. The icing on the cake is an incredibly comprehensive metering system that shows the history of a piece of audio in both the time and frequency domains.

The Master Perfection Suite is a boxed product, but all the component plug-ins are also available separately from the BIAS web site. They are compatible with Audio Units on the Mac and RTAS/Audiosuite and VST on both Mac and PC, and can get by on a Mac G4 or an Intel Pentium III, but a 2GHz or faster processor is recommended. Supported operating systems are listed as Windows XP Home/Professional with Service Pack 2 or Windows Vista for PCs, and Mac OS 10.4 or later on Apple machines. Installation requires 300MB of available hard disk space and authorisation is via the BIAS web site, using the serial number from the software package.

SOUND ON SOUND

BIAS Master Perfection Suite £419

pros

- Excellent sound quality.
- Intuitive interface design.

cons

- No side-chain filter on gate.

summary

A great plug-in tool kit well suited to, but not limited to, mastering and mix polishing applications.

GateEx

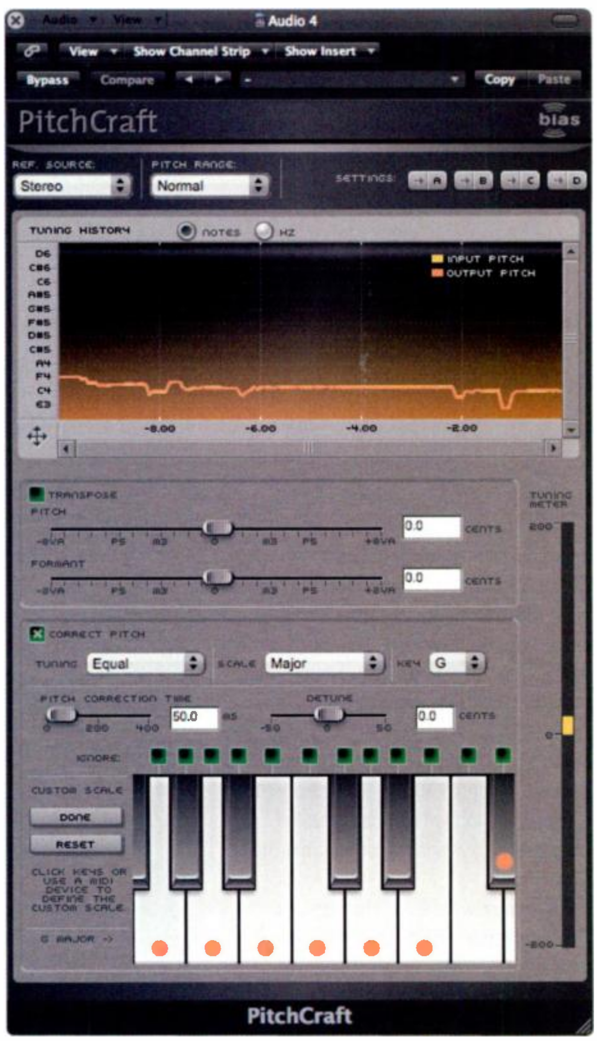
GateEx, as its name suggests, is an expander/gate, and includes high-resolution input and output meters, a waveform display of the input relative to the current threshold setting, and a graphical view of the gate threshold and 'hysteresis' settings. The latter is useful, as it effectively allows the threshold at which the gate closes to be offset to be a few dBs lower than the threshold at which it opens. This can be useful in stopping gate 'chatter' — rapid repeated opening and closing that can occur when the decay envelope of a sound isn't as smooth as it might be.

As well as adjustable hysteresis and look-ahead, GateEx has the expected attack, release, hold and depth controls, and in expander mode the ratio may also be set by the user. As with the other plug-ins included here, up to four snapshots can be saved and called up for instant comparison, and parameter values may be adjusted with the on-screen controls or typed directly into parameter boxes. You can save or load settings at any time in the usual way.

The gate works fine and the waveform display really helps to show you what is above and below the threshold. However, there's no side-chain filter, so you can't make the gating frequency-sensitive.

PitchCraft

Like several other pitch-fixing plug-ins, PitchCraft adopts the paradigm first developed by Antares with their ground-breaking Auto-Tune plug-in, displaying the musical scale as a one-octave section of piano keyboard. Pitch-detection and shifting algorithms are then used to nudge errant notes towards the nearest 'legal' note in the scale defined on this keyboard. The pitch range over which correction is



PitchCraft can be used for basic pitch correction or more creative mangling.

applied can be set by the user, and shifted notes can have their formants preserved automatically, to avoid unnatural changes in character. Of course, if you want unnatural, you can opt to modify the formants or pitch (plus or minus one octave) to morph voices from demonic at one extreme to cartoon mice at the other. You can also adjust the pitch-correction speed to provide slow, natural corrections or fast, vocoder-like note changes that strip away all vibrato and feeling. Where the host program allows MIDI data to be sent to a processing plug-in, new notes can be added to the user scale in real time to control more musically complex passages. As with all such plug-ins, a fairly clean monophonic input is required.

The keyboard displays the current scale, while a history window shows the originally performed pitches and the scale-corrected pitches in a graphical form, so you can see how much work the software is doing and on which notes the biggest problems occur. There's also the familiar tuning meter (vertical this time), which shows in real time how much pitch correction is being applied. However, you don't get the keys lighting up to show you the notes being detected, which I find useful on other plug-ins.

In practice, the pitch-fixing works at least as well as Logic's Pitch Correction, and as long as the original vocal isn't too far out and you pick the right scale notes, it sounds pretty natural. The formant control only sounds natural over a very short range before moving into special-effects territory, and much the same can be said of the pitch-transpose function.

Repli-Q

Repli-Q is a 'fingerprint' EQ and works in a very straightforward way, though you can get your hands dirty and tweak the EQ curves it comes up with if you want to. In essence, it can 'learn' the average audio spectrum from one audio file, save it, then use the information to EQ a second audio file to make its spectrum match that of the first reference file. Repli-Q works over a huge number of frequency bands and so comes up with a very detailed correction curve. This type of plug-in can be useful for making your mixes sound more like those of your favourite producers, but only if the music is broadly similar. For example, any attempt to make a solo flute track sound like a drum and bass mix will simply try to boost frequencies that were never present in the audio being processed, and will invariably sound dreadful. A slider allows the user to vary the amount of spectral correction applied (using less than 100 percent often sounds best), and there's also a smoothing fader to round out some of the finer peaks and holes in the resulting EQ curve that might otherwise create audible 'peaky' resonances.

The spectrum display shows the spectra of the two pieces of audio plus the spectrum of the second piece of audio after equalisation, each of the three spectrum lines being colour-coded. There's also an EQ view mode that shows the correction curve; this can be edited by dragging with the mouse to cut or boost sections in a parametric EQ kind of way, with the width, amplitude and Q of the edits definable by the user. Automatic gain compression and soft clipping is available to take care of any high peaks caused by the EQ.

In practice, Repli-Q works very much like other fingerprint EQ plug-ins I've tried, and produces similar results, which can vary from the spectacularly useful to the disappointing, depending on what you expect it to achieve. If you're doing a cover version of an existing song, for instance, you can get your mix to sound more like the original, though often 75 percent correction is enough in this application. When matching mixes, it can help if the songs are in the same key, as the ▶

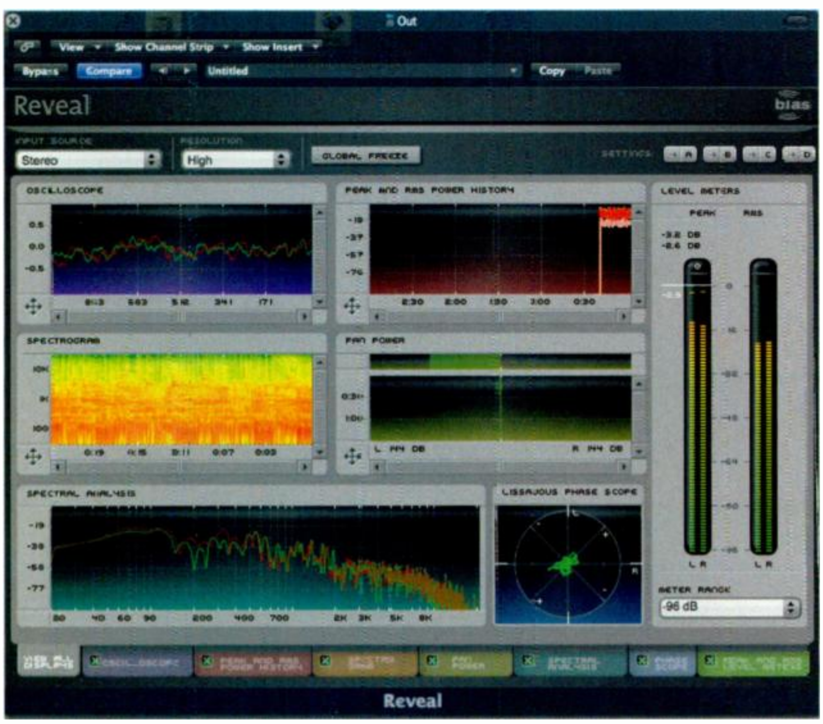
BIAS MASTER PERFECTION SUITE

Repli-Q allows you to impose the frequency profile of one track on another.

▶ fine 'spiky' EQ curve detail often tends to relate to the musical key. Yet another use is to more closely match the sound of mixes done at different times or in different locations so they sit together more comfortably on an album. It can also be used to make DI'd acoustic guitars sound more like miked-up samples of the same instrument, which is a neat way to achieve a nearly miked sound, but with the benefit of the separation that's only possible from DI'ing.

Reveal

Reveal provides a set of meters in a similar format to the respected Spectrafoo plug-in beloved of Pro Tools users. Its various windows offer Peak/RMS level meters, a Lissajous 'jellyfish' phase meter for evaluating the stereo components of a mix, a waveform oscilloscope, a Pan Power meter that looks at the level of non-centred sounds, ▶



Reveal is a comprehensive metering plug-in.

and a Phase Correlation meter. The plug-in also hosts a Peak/RMS power history, a high-resolution spectrum analyser, and a spectrogram that maps out the energy in different parts of the frequency spectrum. The Level Meter range is user-adjustable, and a global Freeze button stops all the displays for analysis. Tabs beneath the multi-meter display allow any individual meter to be viewed full size. The high-resolution meters and spectrum analyser are practical tools with intuitive results, as is the phase meter, which can tell you a lot about stereo width and symmetry, but some of the other meters need more knowledge to interpret. It is worth persevering, though, as the meters can tell you a lot about both your own mixes and commercial mixes, which may help you refine your mixing technique.

Already the high-resolution meters have solved a problem for me, because they showed a background noise level of -75dB when my sequencer wasn't running. This was traced to the live sequencer channel input from my faithful old Roland JV2080 adding its own output-stage noise to the proceedings. Muting its channel until needed cured the problem.

SuperFreq

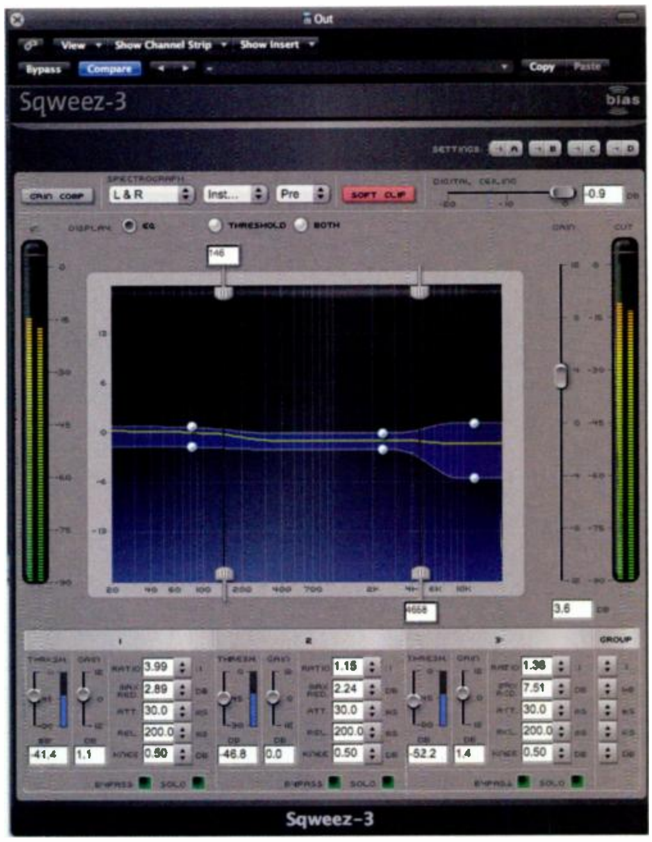
SuperFreq is a parametric EQ that comes in four-, six-, eight- and 10-band versions. There's a familiar curve display, draggable curve points and separate adjustment of gain, frequency and Q for each band, with the ability to set up Peaking, High Shelf, Low Shelf, High Cut and Low Cut Filter types, each with a cut/boost range of ±24dB. Bandwidth (Q) can be set from 0.1 to 30, and each frequency band can be tuned over the range 20Hz to 20kHz. Individual bands can be bypassed, and there are high-resolution level meters to go with the global controls and compare buttons found on all these plug-ins.

Conceptually this is a very conventional EQ plug-in, but I have to say that it sounds surprisingly analogue-like: the results are somehow musically satisfying, and usually require very little cut or boost to achieve. A little low-end boost brings up the bass end of a mix without turning things to mud, while at the other end of the spectrum, a dB or two of air EQ at 10kHz or above really hits the spot, without making the mix seem gritty or aggressive.

Sqweez

Sqweez-3 and Sqweez-5 are three- and five-band versions of the same multi-band compressor, and both provide automatic gain compensation, switchable soft-clipping and a user-settable limiter ceiling. Sqweez has an integrated spectrograph display, and each band can be soloed, allowing you to hear the effect of your compressor settings in each area of the frequency spectrum. There's a global gain control plus adjustable band levels, and each band has controls for Threshold, Ratio, Maximum Gain Reduction, Attack, Release and Knee shape.

The ability to set the maximum amount of gain reduction applied regardless of input level makes this compressor very powerful. In effect, it means that you can apply compression to mid-level signals without overly changing the dynamics of very loud or very quiet sounds. It is also possible to select negative compression ratios, to turn the bands into expanders where necessary for increasing dynamic range.



The Squeez multi-band compressor offers some unusual features, such as the ability to restrict the amount of gain reduction in each band.

I enjoyed using Sqweez, as it is straightforward and has an informative interface that shows in a very intuitive way how the gain changes. The graphic display has draggable crossover points, so you can decide where best to split the audio spectrum if you don't like the defaults. The ability to limit the amount of gain reduction taking place in any band is hugely useful, and I got some great-sounding results out of it in just a few minutes. For mastering, low ratios and low thresholds are the order of the day, and some very low values need to be typed in, as they fall between the steps you get if you use the on-screen controls, which is no problem. Higher ratios can be useful on individual tracks: a particularly useful application of multi-band compression, for example, is to even up the level and tone of a vocal where the singer hasn't been too careful about maintaining a sensibly constant distance from a cardioid-pattern microphone, and the changing distance also alters the amount of bass lift.

Summing Up

In all, the Master Perfection Suite is a nice bundle, with the stars for me being the SuperFreq EQ, the Sqweez multi-band compressor and the elaborate Reveal metering. SuperFreq, in particular, sounds very 'analogue' and smooth, which can't be said of all plug-in parametric equalisers. The pitch and fingerprint EQ plug-ins work more or less as well as the ones in Logic 8, and users of other DAWs that don't include such tools should find them very useful. While the expander/gate is very smooth and predictable, with a better-than-usual display, the lack of side-chain filtering could be a limitation for some users.

This particular plug-in bundle covers a lot of ground, and given that not everyone will find all the components equally useful, it's sensible of BIAS to make them available individually. The bundle price is very reasonable given the undeniable quality of the individual plug-ins, but if you already have some equivalent plug-ins that you're happy with, it's nice just to be able to buy the tools you need. Overall, the Master Perfection suite lives up to the very high standards that BIAS have set themselves in other software areas. SOS

information

- **£** Master Perfection Suite (boxed) £419.01 including VAT. GateEx $59; SuperFreq $79; all other individual plug-in downloads $149.
- **T** SCV London +44 (0)20 8418 0778.
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Classic Tracks

Blondie 'Hanging On The Telephone'

The partnership between Blondie and producer Mike Chapman created a perfect pop record — and catapulted the group from the underground to mainstream chart success.

Artist: Blondie
Track: Hanging On The Telephone
Label: Chrysalis
Released: 1978
Producer: Mike Chapman
Engineer: Peter Coleman
Studio: The Record Plant

Richard Buskin

If Blondie's *Parallel Lines* album was the New York erstwhile-punk band's finest hour (all 38 minutes of it) and a perfect encapsulation of top-drawer, high-tech 1978 pop-rock, then it also marked the career apex of its producer, Mike Chapman, a man who had already established himself with a form of music that has come to define its era.

In popular music terms, the 1970s was a decade that swung wildly from glam, reggae, progressive and AOR to metal, disco, corporate, punk, new wave, neo-mod and pure pop. And there, at both the start and end, was composer/producer Chapman, crafting glam rock — with writing partner Nicky Chinn — by way of acts like the Sweet, Suzi Quatro and Mud, and then classic power pop on his own with Blondie, Pat Benatar and the Knack.

'Co-Co', 'Poppa Joe', 'Little Willy', 'Wig-Wam Bam', 'Block Buster' and 'Ballroom Blitz' by the Sweet; Suzi Quatro's 'Can The Can' and 'Devil Gate Drive'; Mud's 'Dyna-Mite' and 'Tiger Feet' — all are synonymous with the visual excesses of glittery jackets, flared trousers and platform shoes as much as they are with the glam music that melded pop melodies with crunching, fuzzy guitars and heavy drums, bathed in a sound that was a throwback to Sun Records in the mid-'50s. Featuring catchy songs often performed by talented musicians, the 'Chinnichap' formula certainly worked during the halcyon period of 1973-74, when it accounted for 19 Top 40 UK singles, including five chart toppers. And when glam's sparkle had faded, Chapman then caught a second wave with his aforementioned solo productions. Not that the partnership with Chinn had ever really been about songwriting once the hits began rolling in.

From The Club To The Charts

"I first met Nicky in late 1969 when I was working as a waiter in the discotheque of a London club called Tramp," recalls Chapman, a native Australian who had relocated to the UK in June 1967, at the height of flower power. "That was the last real job I ever had. I was in a struggling band named Tangerine Peel and needed to do something to pay the rent. I'd been writing pop songs for a couple of years — all of which was second nature to me, having grown up listening to 'Peggy Sue', 'That'll Be The Day', 'Wake Up Little Susie', 'Blue Suede Shoes' and so on — and by 1969 I felt like I was on the right track. At the same time, Nicky was a regular customer at Tramp; a rich English kid with nothing better to do than go out every night and dance very badly. After he told me he'd heard I was in a band and that he, too, was a songwriter, I took my guitar to his apartment in Mayfair and we set about trying to write together.

"Over the next two weeks we knocked out four little pop songs, but I quickly discovered that Nicky's musical taste was completely different from mine, and that would cause us a lot of problems as the years went on. He was into James Taylor, Carole King and Joni Mitchell — all of the things that I couldn't stand. I appreciate them now, but at that point, never having been much of an album buyer, I was more into great pop songs like the Archies' 'Sugar, Sugar' and all of the Creedence Clearwater material; anything with a big old hook, guitars and a great beat. The minute I saw him, David Bowie also influenced me, as did Marc Bolan in about 1970, but Nicky Chinn didn't understand any of that. I had to explain to him what it was all about while he'd sit there with a blank look on his face and ask me, 'Have you heard the new Joni Mitchell album?'

"One night, after I'd quit the job at Tramp to concentrate on my songwriting, we were sitting in Nicky's apartment and he said, 'Why don't we call Mickie Most?' I'd been talking about how great Mickie was, and so we looked in the phone book and there he was. I said, 'I'm not calling him. You call him,' so Nicky called, Mickie answered the phone, and he said, 'Sure, come in tomorrow and play me your songs.' The next day we went to his office — a huge, square room on Oxford Street, with a desk in each corner facing toward the middle. Mickie was sitting at one desk, blasting out music; Peter Grant, who was managing Led Zeppelin, was at another, making deals and cursing people in all four corners of the world; and then there was Ronnie Madison, who was trying to do the accounts; and Dave Most, Mickie's brother, who was a promo man, screaming at radio guys. I'd never seen anything like it."

While all this was going on, Chapman

The Glam Rock Sound

After Richard Dodd recorded the very first sessions with the Sweet, Peter Coleman was Chapman's full-time engineer all the way through the 1970s, helping him to shape the aforementioned glam rock sound that had been influenced by Tony Visconti's productions of T-Rex, which themselves had been influenced by records of the late-'50s and early-'60s.

"For me, a track like 'Ride A White Swan' was pure magic," Chapman says. "Its groove epitomised what I was trying to accomplish, and then, when I heard 'Hot Love', it was like, 'Oh, my God, that's it.' So Visconti was a big influence on me in terms of the sonic approach, while the grooves all came from the mid-to-late-'50s. On top of that, it was about what could be done with the drums to make them sound a bit different from everybody else, as well as how freaky we could get with the guitars and vocals. There was that kind of slapback echo sound together with very '70s-sounding rock guitars, and it was the combination of those elements that became the blueprint for glam rock, which all came out at once with Gary Glitter. Mike Leander put all of the elements together, as did Slade to a certain extent.

"Still, if you listen to those records side by side, they're all different. They all have the same sort of vibe to them, but none of the drum sounds are alike, none of the guitar sounds are alike, none of the vocal sounds are alike. All of us producers were trying very hard to sound different from one another, even though we were following the same path, and it's pretty hard to do that. When I listen to the emo kids these days, I can't tell the difference between one band and another, and that's because producers aren't trying to be different anymore, they're just trying to do what everybody else is doing. Back in 1973 and 1974, the challenge was 'Let's have a hit, but for God's sake, let's make it sound different to Gary Glitter, T-Rex and Slade.' For my part, I had the Sweet, Suzi Quatro and Mud going all at once, and I had to make all three of them sound different, which they did."

Photo Roberta Bayley

Five sixths of Blondie, with Mike Chapman (centre, in sunglasses) and manager Peter Leeds (far right), in the control room of the Record Plant's Studio A during the recording of *Parallel Lines*.

attempted to play five songs that he and Chinn had written, only for Mickie Most to stop him after about eight bars of each of the first four numbers, muttering, 'No, no, no, no.' Feeling dejected, Chapman nevertheless managed to make it all the way through the fifth song, a number titled 'Tom Tom Turnaround', and at the end of his performance Most asserted, "That's a hit." Which it was, after he cut it with an Australian trio named New World.

Hit Factory

"I understood why Mickie liked that song," Chapman says. "It was different to the other four that Nicky and I had written together. Nicky didn't write melodies, he saw himself as a lyricist, and those songs contained a lot of his lyrics. On the other hand, 'Tom Tom Turnaround' marked the beginning of me coming up with titles that would prompt him to look at me and go, 'What's all that about?' You can imagine what he said when I told him I was going to write songs with titles like 'Can The Can', '48 Crash' and 'Ballroom Blitz'.

From 1970 to 1975, New World, the Sweet, Suzi Quatro and Mud all required three hits a year, and the responsibility for this basically fell to Mike Chapman — an annual total of 12 hits.

"If somebody asked me to do that now I'd run screaming," he says. "I don't know how on earth I managed to do it, but I pulled it off. I was so focused on what I needed to do next, whether it was the next hit for the Sweet, Suzi Quatro or Mud, or whether Mickie needed another song for New World, like 'Living Next Door To Alice' before I cut it with Smokie. I was constantly aware of the requirements and my head was just full of all these bizarre words. They'd pop into my head and suddenly I'd go, 'That's it, that's the song.'

"In the beginning, with the first half-dozen songs that Nicky and I wrote together, I'd compose the melodies because he didn't have any musical knowledge. I'd play guitar, he'd sit there with a pen and paper, and we'd come up with the words together. But then, as the songs became more and more bizarre, it was pretty much me sitting there and

Mud, Sweet & Suzi Quatro

It was through Nicky Chinn's own efforts that he and Mike Chapman became involved with the Sweet. This was after Chapman followed Chinn's advice to have a proper band demo their songs in the studio rather than record them himself on a small Revox multitrack tape machine. One of those songs was titled 'Funny Funny', and although not intended for the Sweet, it subsequently became the group's first hit after Mickie Most made a rare slip by allowing them to sign with RCA.

"Here was a Deep Purple-type rock band with a song called 'Funny Funny', but it was on the charts and that's all I cared about," Chapman remarks. "Then again, Phil Wainman was the producer, and it was difficult for me when we were cutting it in the studio. The song was in my head, I'd be trying to tell him what to do, and he ended up telling me to go sit in the back of the control room or leave the control room altogether. 'I'm the producer,' he'd say. 'Go away, you're getting on my nerves.' 'OK, sorry,' I'd reply. 'It's just that the song's in my head and I'd like to help you.'

"As the hits went on I slowly but surely inched my way closer to the console, and I started producing full-on in 1972 when Mickie Most called and asked me to work with Suzi Quatro. He had cut a few things with her but just couldn't find the right style, and to my surprise he called me at home one night and asked me to write a song for her and produce it. This was his big new artist that he'd found, and he handed me the opportunity on a plate. I was so excited. Within two days I came up with 'Can The Can', and a week or two later I was producing Suzi in the studio. Then, having decided we needed another band, Mud came along right after that, and so now I was producing Mud and Suzi Quatro, and also having a lot more to do with the Sweet's production as we got into tracks like 'Wig-Wam Bam' and 'Ballroom Blitz'."

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"So, pretty much what happened from that point was Nicky would go out and hustle the records, calling the record companies and chasing the promo guys. He was really, really good at that, and I couldn't have done it. His role in the partnership was never really to write songs with me or produce the records. It was to take care of the business side of things. That was his forté. And from 'Can The Can' onwards his contributions to the songs became less and less. He'd be out and about all day, I'd be sitting in his apartment with the guitar, and when he'd come back and ask, 'What have you got?' I'd generally have most of the song written. Then, once I had a first verse and chorus, he'd understand where I was going and he would sit down and try to help me with the words... There's a wonderful photo of us in the studio with Suzi, where I'm sitting at the console and he's in a corner reading the *Financial Times*."

New Beginnings

As his relationships soured with both Nicky Chinn, with whom he didn't see eye to eye, and the Sweet, who appreciated neither his pop sensibilities nor his autocratic work methods, Mike Chapman determined to "get the hell out of England," and in 1975 he relocated to Los Angeles. There, while still churning out the hits for Smokie and Suzi Quatro, he could focus on a US market that hadn't afforded him much success, and it was in 1977 that Terry Ellis asked for Chapman's feedback on Blondie, who he was thinking of signing. Chapman subsequently saw the band play on three successive nights at the Whiskey a Go Go on the Sunset Strip. After his rave review prompted the Chrysalis boss to purchase Blondie's contract from Private Stock Records, reissue their eponymous 1976 debut album and release the 1977 follow-up, *Plastic Letters*, Chapman was hired to produce the third LP.

By then, working with Mud, Suzi Quatro and Smokie, he had already produced albums containing others' material. *Parallel Lines*, on the other hand, was the first such project on which he contributed none of the songs. Instead, he was assigned by Terry Ellis to ensure that what Blondie brought to the summer 1978 sessions evolved into hit material, and ensure this he did. Following its release in September of that year, the album would hit number one in the UK chart, peak at number six in the US, and yield chart-topping singles in the form of 'Heart Of Glass' in the US and UK and 'Sunday Girl' in the UK, where 'Picture This' reached number 12 and 'Hanging On The Telephone' peaked at number five.

"I wasn't being used as a songwriter, but as a song manipulator and song construction consultant/technician," Chapman says. "There was a lot of stuff that needed to be put together, because as loose as the band were, their songs were even looser. Of course, being that I'd started out as someone who wasn't really into albums but into writing singles, I had done a complete turnaround, and I was loving every minute of it. You see,

Mike Chapman with Debbie Harry and Ronnie Spector, who was visiting the session, and whose presence apparently made Debbie very nervous when doing vocal takes for 'Fade Away And Radiate'.

by then the only writing responsibilities I had were to come up with a hit or two each year for both Suzi Quatro and Smokie, and those were easy gigs because they were nice people to work with. There was no suffering on those sessions. Blondie, on the other hand, was all about suffering."

Meet The Band

"The Blondies were tough in the studio, real tough. None of them liked each other, except Chris and Debbie, and there was so much animosity. They were really, really juvenile in their approach to life — a classic New York underground rock band — and they didn't give a f*ck about anything. They just wanted to have fun and didn't want to work too hard getting it."

For *Parallel Lines*, the group's line-up comprised lead singer Debbie Harry, Clem Burke on drums, guitarists Chris Stein and Frank Infante, English bassist Nigel Harrison and keyboard player Jimmy Destri. Yet, even though Chapman loved Blondie's first two albums and was enamored with the group members' offbeat sense of humour, he doesn't mince his words with regard to what he describes as "musically the worst band I ever worked with."

"The only great musician among them was Frankie Infante," he asserts. "He's an amazing guitarist. The rest of them were all over the bloody place. Jimmy Destri was a pretty good songwriter, but he wasn't a great keyboard player. What he did, he did well, and I didn't ever try to push him beyond that because I knew there wasn't anything beyond that. Chris Stein was always so stoned, and although Clem Burke had all the right ideas, he had no sense of timing. I mean, he had a lot of ability, but I always felt he was trying too hard, and that's what I used to tell him.

"If you're going to use the Keith Moon approach, you'd better be able to pull it off, because if you don't it's just going to be a shambles. Sure, there were some timing discrepancies on some of those Who records, but Keith Moon had the ability to do all that manic stuff and keep the groove solid at the same time. Clem hadn't learned how to do that yet — he would later on — and so although Nigel was a very competent bass player the rhythm section was totally out of whack.

"Nigel actually brought a lot to the table. He brought terrific songwriting, a sense of humour, and the fact that he was English added another dimension to the band. He got along great with Debbie and OK with Chris, whereas Chris never wanted Frankie in the band. The fact was, Frankie made Chris look like a terrible guitar player. Then again, I loved Chris, and I worked very, very hard with him for years and years because I felt he deserved my time. He, to me, was a wonderful, wonderful songwriter, a great songwriter, and he was always so concerned about his playing ability. I'd say to him, 'Why would you even worry about that when you're such a great songwriter? You can't be everything. Let Frankie play those solos.'

"He didn't like that idea, and so I'd send everybody else out of the studio and I would sit with Chris for hours and hours; just me recording and him playing to get all his parts right. I wanted him to get his parts right because I knew that was important to him. To this day, when he listens to those records, he knows the work that was put in and he's proud of it. And thank God we did that. I think a lot of other producers would have said, 'No, you're not playing that part right. I'll bring in a studio player.' But that was never an option with Blondie.

"Debbie is a great singer and a great vocal stylist, with a beautifully identifiable voice. However, she's also very moody. I love Debbie and I learned a lot from her about the psychology of recording vocals. Up until her I had been pretty barbaric in my approach to vocalists, like 'Get out there and sing!' Once I encountered Debbie, I learned how to soft-shoe it a little more. The vocals I got out of her I really had to fight for psychologically, and when I listen to them now I remember those sessions very clearly. They were tough times, with a lot of tears, a lot of disappearing into the bathroom for hours. She's a very emotional person and those songs meant a lot to her. When she was on — bang, it all happened really quickly. She'd never had to work hard in the recording studio prior to meeting me. She'd go in and do one pass and that was it. I'd have her out there, singing over and over again, until I felt that she had lost the plot, at which point I'd say, 'OK, that's it for today, let's try it again tomorrow.'

"Things like that didn't sit too well with her in the beginning, but it worked in the end. Musically, Blondie were hopelessly horrible when we first began rehearsing for *Parallel Lines*, and in terms of my attitude they didn't know what had hit them. I basically went in there like Adolf Hitler and said, 'You are going to make a great record, and that means you're going to start playing better.'"

Blonde Ambition

"On *Parallel Lines*, I was given the responsibility by Terry Ellis to put this ▶

Suffering the effects of Mike Chapman's incredibly high playback volume during mixing for *Parallel Lines*.

band at the top of the charts. He knew they could achieve that and I knew it, too, but I also knew that, given how they were when I began working with them, it might never happen. Terry said, 'Can you do it, Mike?' and I said, 'Yes, I can.' He said, 'OK, I'm going to leave you alone. You've got six months.' So I had to go in there and knock this band into shape."

In the event, the album took six weeks, not months, to record at the Record Plant in New York, before Mike Chapman and Peter Coleman opted to get away from the big city by taking care of the manual mix on an obscure Sphere console during about 10 days at a small studio named Forum in Covington, Kentucky. Indeed, since only five weeks were initially booked at the Record Plant, the final recording sessions were switched from that facility's Studio A to what Chapman now describes as "the slummy room at the top. It was a real bomb of a studio, but hey, a case of whatever we could get... There was an API console in the room that we started in, the main monitors were Westlakes, and we recorded to 24-track.

"Back then, I liked pretty much any speakers that were big and loud. For nearfields, the only thing we had in those days were Auratones, and we'd normally just use one of those to check out the sound in mono, since we still had AM radio playing number one records. Then again, having grown up listening to everything on '11', I'd turn things up as loud as they could go, thinking that if it felt good and sounded good at that level it must be right. That having been said, for the early records that I produced in England I was always working on Neve consoles, and they all had that little mono speaker. Well, we would check all of our mixes back on that tiny little mono speaker. It sounded dreadful, but if your mix was good coming out of that, and at 4000 watts, then you knew you'd got it right."

Not as informed about — or interested in — recording technology as he would become once computers entered the scene, Chapman would usually remedy a sonic problem by instructing engineer Peter Coleman to "just keep turning the knobs and I'll tell you when it's right."

"I used to act dumb by using a lot of that terminology in the studio," he now admits. "I did that for two reasons: I wasn't very technically proficient because I just hadn't paid enough attention over the years; and with most bands, especially Blondie, it was important for them to see me as somebody who was fighting for performance rather than trying to make them sound spectacularly hi-fi. I was there as one of the foot soldiers."

Evidently, this approach paid off. "He was a very good producer," Jimmy Destri later remarked. "He wasn't very technical, but he was very organic and he was a very good mixer on his own, too. I mean, he knew the console like nobody else I've ever seen. He would say things like, 'Jimmy, if you shut out the lights I'll be able to EQ by ear' without even looking at the console! He taught me a lot about making records, that's what Mike did. And he was another member of the band at that point, and he was just like in there with us. And from *Parallel Lines* and onwards, Mike was integral, he was really integral, as we couldn't go in the studio without him. As far as the recording process of those albums goes, we all learned a lot from Mike."

Not that his input was always appreciated. Tension was often a part of Blondie's make-up, and during the *Parallel Lines* sessions Nigel Harrison and Mike Chapman butted heads when the producer kept pushing the bass player to improve his performance.

"We almost came to blows," Chapman recalls. "He told me, 'Shut the f*ck up! What do you know? I'm trying my best,' to which I responded, 'Well, it's not good enough.' Still, no matter who I pissed off — and Jimmy was certainly among them — the Blondies all basically knew I would get it right. They sometimes didn't like the procedure, they didn't like the amount of time they had to spend doing it, but after we'd finished *Parallel Lines* they understood why I did what I did and they were all very proud of the record."

Heart Of Glass

In geometric terms, parallel lines conform to a pattern but they don't connect, and neither do the characters in most of the songs on the aforementioned album: Harrison and Harry's 'One Way Or Another'; Destri, Harry and Stein's 'Picture This'; Harry's 'Just Go Away'; Stein's 'Sunday Girl' and 'Fade Away And Radiate'; Harry and Stein's 'Pretty Baby' and 'Heart Of Glass'; even non-band-member Jack Lee's 'Hanging On The Telephone'.

These were supplemented by Infante's 'I Know But I Don't Know', Destri's '11:59', Lee's 'Will Anything Happen' and an energetic cover of Buddy Holly and the Crickets' 'I'm Gonna Love You Too'. And as directed by Mike Chapman and adorned with Deborah Harry's slut-next-door appeal, this added up to a platinum-selling album. Yet only some of the songs were completely written when the participants first entered the studio.

"Debbie and Chris had a great set of ears," says Chapman. "When they said they wanted to record a song, I never said, 'No', even if it was an outside song, and in this case there were three of those numbers right from the start. 'Heart Of Glass', on the other hand, was called 'Once I Had A Love' and they had it in a lot of different versions, but it wasn't right in any form."

Originally recorded by the band in 1975 with a relatively slow tempo and blues/reggae vibe, 'Once I Had A Love (aka The Disco Song)' made its way onto a digitally remastered 2001 reissue of *Parallel Lines* courtesy of a 1978 Chapman-produced demo.

"After they'd played me the covers, as well as some of their sketchy song ideas, I decided the first thing we should work on was 'Once I Had A Love'," Chapman now recalls. "I thought that track was the one that probably needed the most attention, because even though it was complete, it was wrong, and I knew that if we could get it right it might be a big hit. So there we were on the first day of rehearsals, in some little hole-in-the-wall on the Lower East Side, and all of the band members were being very, very cautious about having a new producer. This was not their idea, they would have gone back to Richard Gottehrer. And although they knew who I was and what I'd accomplished, they didn't quite understand what was going to happen. Neither did I.

"In discussing what to do with 'Once I Had A Love' I tried to include everybody, and after we played it a few times I said, 'Let's get rid of the reggae.' We then tried to do it as straight rock, but that didn't work, and I could see Debbie was getting a bit frustrated. So, I asked her, 'Debbie, what kind of music that's happening right now really turns you on?' She said, 'Donna Summer.' I said, 'OK, then how about us treating this song like it was meant for Donna Summer?' Thay all looked at me as if to say, 'What?' I said, 'Well, it's disco, right?' 'Yeah, it's disco,' they mumbled, but when Debbie then said, 'I like disco,' the others basically went along with it.

"Anyway, we fooled around with the song, and after a couple of hours of very intensive work we had it sounding pretty much the way it sounds today. We had a little Roland drum machine, and I said, 'Why don't we just put a groove together in this and play along to it?'

"At the same time, we also changed the title. I said, 'You can't call it 'Once I Had A Love'. The hook line in there is 'heart of glass'. Let's call it 'Heart Of Glass'.' So that's how the song evolved, and after leaving the studio that day all of us were on the street, getting cabs or whatever, and Debbie walked alongside me and said, 'Mike, I really like what you did with 'Heart Of Glass'.' 'Thanks, Deb.' 'You're welcome.' We had broken the ice. Still, recording 'Heart Of Glass' was tortuous. In those days, we didn't have MIDI, so we used the Roland drum machine and, because I added a time change in the middle, I had to actually sing through the song with that thing going at the same time, and then press the button to stop and start it again so that we were on the right beat."

Group Dynamics

"You see, 'Heart Of Glass' was the only track that we put together piece by piece. Everything else was played together as a band, and since we didn't use click tracks for that whole album I set up a mic in the middle of the room. Debbie didn't want to sing scratch vocals, and I didn't want her to, either, because it would have blasted her voice. So while she and Peter Coleman sat in the control room and laughed at me, I would be in the live area with the rest of the band, keeping time and singing scratch vocals for all of the songs. I was isolated and wore a set of headphones, and I'd say, 'OK, here we go. Count it off, Clem, and watch me for your time.' Then I'd launch straight into it, getting very nasally while screaming and jumping up and down, and when I'd look into the control room Debbie would be in fits of laughter because I looked so stupid. We didn't have video cameras in those days — I wish we had. It was entertainment for her.

"I had all the energy that the band needed. You know, 'Just take the energy from me.' That's why I liked to stand in the middle of the room with them and conduct and sing and scream and jump up and down. I didn't care if it ended up leaking onto the drum mics so long as the performance was right and the groove felt good. Never mind if I was completely out of tune. I'm sure that the vibe I gave off was part of the reason they were able to put that into the track.

"I basically served as the drum machine on most of the songs, whereas on 'Heart Of Glass' we used the Roland. It took us ages to get that part right. Then, when it came to the real drums, we had to record them one piece at a time, which none of us had ever done before. They were all looking at me like, 'Wow, this is cool. We're experimenting.' I said, 'Let's just have fun with this,' and by the end of that first day we had all of Clem's drums down. We put the kick drum down first, then a hi-hat after that, followed by a snare... He didn't want to do it this way at all, and he was very, very moody, but Debbie and Chris were running the show and they said, 'Just do it.' He hated it, and he probably still does, but at the end of that first day we had a great drum track and we all knew it.

"After that we put down sequenced parts with various different keyboards, trying to incorporate that Donna Summer vibe, and then we added lots of Chris's guitars with echoes on them. At that point I realised everybody was actually having fun with this track. Having made hit records for the past seven years, I had a handle on what I was trying to accomplish, and I knew that with each piece that we added we were getting closer and closer to our goal — which was to have this incredible track that didn't sound like anything else. It was a combination of all sorts of different things, and although Debbie's voice and the song's structure meant it was Blondie, it was Blondie as they had never sounded before. When Debbie sang it, she really did become Donna Summer, and I thought it was good that the track wasn't like anything else that was on the album."

One Way Or Another

At the Record Plant, the recording setup for *Parallel Lines* was a traditional one. The drums were miked with a Neumann U47 on the kick and KM84s on the toms, snare and hi-hat, along with a couple of U87s overhead.

"After the basic track was cut, we'd go through the whole thing and tighten up the kick drum with the snare, by way of pencil erasing," Chapman says. "That meant using a pencil to hold the tape away from the head and erasing up to the kick drum. If a bass part was ahead of the kick, you could erase it so that it sounded like it was on top of the kick. That's very easy to do these days, but back then it was quite a procedure just to get the bottom end sounding nice and tight."

While the bass combined a DI signal with that from Nigel Harrison's amp, and the DI/amp method was also applied to recording Jimmy Destri's Farfisa synth, Frank Infante's Les Paul was recorded with a combination of Shure SM57 and AKG 414 mics on his Marshall cabinet.

"Chris had all kinds of weird and wonderful guitars," Chapman remembers, "and he also had some weird amps, although he liked Fenders, too. He didn't care so long as it sounded good in the control room."


Blondie guitarist Frank Infante and bass player Nigel Harrison supporting Mike Chapman.


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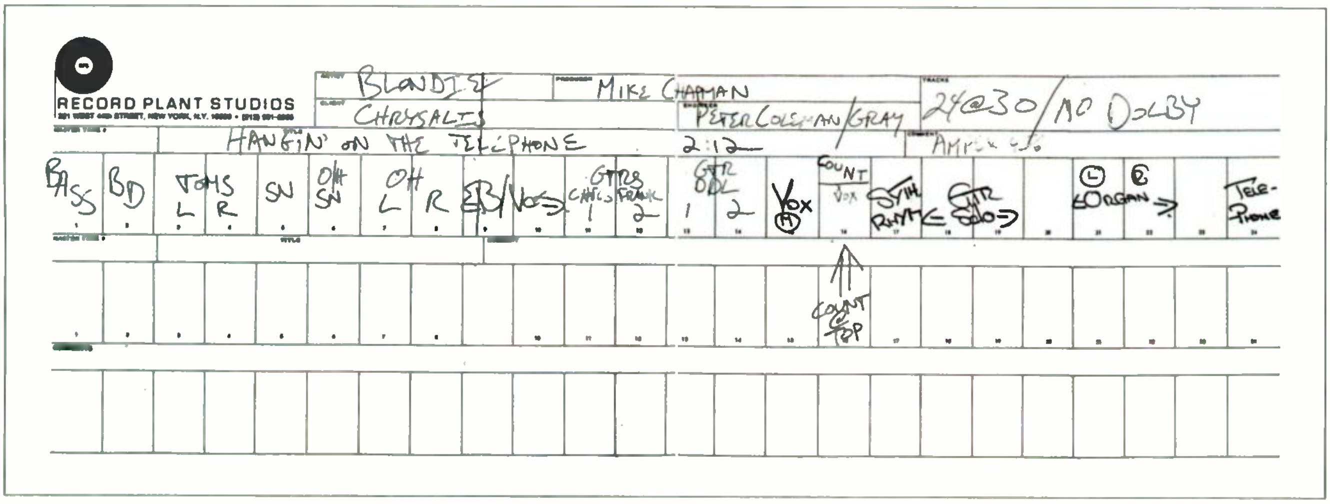
RECORDING 'HANGING ON THE TELEPHONE'

Once the basic track was recorded, this would be followed by the lead vocal, backing vocals and then overdubs, and while there'd be plenty of the latter on subsequent albums, in the case of *Parallel Lines* it was rare for a song to utilise all 24 tracks. Not that this was always obvious in advance. Many of the songs, such as 'Sunday Girl', 'Picture This' and 'One Way Or Another', were still only half-written when the rehearsals took place, the last-mentioned simply comprising a riff with no melody or lyrics.

"Debbie started writing that song during the rehearsals and it was finished just prior to going into the studio," Chapman explains. "Then again, I don't think the lyrics to 'Sunday Girl' were written until we got to the studio, and the same probably applied to 'Picture This'. I also remember 'Fade Away And Radiate' being very sketchy at the rehearsal and Chris repeatedly saying, 'I want to get Robert Fripp, I want to get Robert Fripp'. I was thinking, 'Oh heavens, no! Who knows what he'll do?' As things turned out, having his guitar part was a good idea.

"'Pretty Baby' was in fairly good shape, but, as with many of the songs, Debbie didn't finish writing the lyrics until we were in the studio. In fact, a lot of the time she was still scratching out lyrics when I was asking her, 'Are you ready to sing?' 'Yeah, just a minute...' Some classic songs were quickly knocked out like that by our Debbie."



The track sheet for 'Hanging On The Telephone'. Note track 24...

Hanging On The Telephone

One number for which she didn't have to put pen to paper was *Parallel Lines*' dynamic opening track, 'Hanging On The Telephone', which was also the opener on a 1976 EP by guitarist Jack Lee's short-lived LA power pop trio, the Nerves. Blondie had shared a bill with the Nerves on one of their first visits to the West Coast, and they had already worked on the song by the time they introduced it to Mike Chapman.

"That track was magic from the beginning," he says. "Unlike some of the others, it was an easy one to cut because it was more like Blondie's normal, frantic sort of style, and I also vibed it up a lot. Initially, they didn't know quite how much to put into it, but I told them, 'Look, this is more like the stuff on your first two records. Let's give it that sort of punk/new wave attitude.' I knew that the energy level on that track would make or break it. If we didn't have that energy we'd miss the point, because the musical structure of the song is very tense — it sits you on the end of your chair, and we had to have a track that did the same thing.

"They were all very much into giving it that full-on energy, and of course this was Clem's favourite way of playing. If he really liked something, that in itself added extra energy. So, I think we did four takes and I then took the best one to work on and fix things. If there was a guitar mistake or a bass mistake, we'd punch in and out. In those days, I didn't cut the tape a lot like I'd do later on."

While Burke's sharp drumming and Nigel Harrison's pumping bass are punctuated by Frank Infante's electrifying, punk-edged guitar lines, 'Hanging On The Telephone' is nevertheless powered right from the start by Deborah Harry's energetic, in-your-face vocals as she spits out the song's staccato-style opening lines with machine-gun rapidity: "I'm in the phone booth, it's the one across the hall. If you don't answer, I'll just ring it off the wall. I know he's there, but I just had to call..."

"Debbie always got it right away whenever I tried to describe what to do, but a lot of the phrasing was totally down to her," Chapman states. "She has a strange way of delivering certain phrases, and I found myself accepting things from her that I never would have accepted from anyone else. I would have had other people change it, whereas with her I'd think, 'No, no, no, I've got to leave it like that,' or else it just wouldn't be her. For instance, in 'Hanging On The Telephone', the lines 'I heard your mother now she's going out the door. Did she go to work or just go to the store?' — I remember listening to those and thinking, 'This is the dumbest lyric I've ever heard.' However, it was so dumb, it was beautiful, it was brilliant, and when Debbie then sang it in her inimitable way it suddenly sounded even funnier. It just sounded like the weirdest, most bizarre thing I'd ever heard."

"I can't remember all of the specific phrasing issues, but I know there were many times with different songs where Debbie would phrase something in a very strange way and I'd think, 'Well, if I change that and make it normal, I'm going to take some of the character out of her voice.' It was always very important to me with the Blondies in general to present them the way they were. This wasn't a band that you messed around with or tried to reconfigure or reconstruct. Either it was going to work or it wasn't, and 'Hanging On The Telephone' was one of those cases of 'Just get out there and play it full-on!'

"I used to say to them, 'Think of being onstage. Imagine you're playing this to an audience, because we're trying to record something that you're going to have to listen to for the rest of your lives. So if this is not a high-energy performance, you're going to say, "How come we now do it better live than on the record?"' So many bands end up saying things like that: 'How come it always sounds better live?' Well, that's never going to happen with me as a producer. And in the case of 'Hanging On The Telephone', that's probably the best track on the album in terms of energy, although 'One Way Or Another' has a similar edge."

This Is Blondie

Once Harry's lead vocal had been recorded and comp'd, she and Chapman then contributed the backing vocals, with him harmonising an octave under her on the chorus title line, as well as on the 'whoa-oh' chants as she insists, "Hang up and run to me," and the song races breathlessly towards its manic finale.

"That 'whoa-oh' backing was something that I came up with because I felt that it just sort of added even more energy to the end of the song," Chapman recalls. "Then, after we had the track down, I said, 'You know, we should put a telephone ring on the front of this.' The Blondies all thought that was stupid and too gimmicky, but I said, 'C'mon, guys! Gimmicky? This is Blondie. Let's give it a try!' I told Peter Coleman to call anyone he knew in London in order to record a British phone ring, and then once we stuck that on the front of the song they all went, 'Oh, yeah, that does sound pretty cool.' It certainly heightens the impact of the opening: the ring, then a pause and — wallop! — in it comes.

"That's the magic of *Parallel Lines*. Every track is perfect from top to bottom, and it's a beautiful album because it works in every respect. It's hard to find a flaw in it, and there aren't many records during your career that you can say that about." SOS

Modern Times: Mike Chapman Today

"It's like the old days for me," says Mike Chapman with regard to producing LA 'blast pop' band the Automatic Music Explosion, among numerous recent new-artist projects. "Unlike alternative groups like the White Stripes and the Yeah Yeah Yeahs, who I think are wonderful, today's emo-style bands are so contrived, all doing the same thing as each other. There's nothing special about them, whereas AME is like stepping back into the past with a fresh approach. It's pure pop music, they're great musicians, they put on a great live show and the whole album rocks like hell."

That's quite an endorsement from a man who, in addition to following *Parallel Lines* with three more Blondie albums — *Eat To The Beat* (1979), *Autoamerican* (1980) and *The Hunter* (1982) — helmed projects during the '80s and '90s by Deborah Harry, the Divinyls, Lita Ford, Rod Stewart, Altered Images and Bow Wow Wow. After splitting with songwriting partner Nicky Chinn, Chapman also enjoyed compositional success with Tina Turner's 'Simply The Best' and Pat Benatar's 'Love Is A Battlefield' (both co-written with Holly Knight). And now, based on the East Coast, with a resumé that includes 140 Top Ten hits and estimated worldwide sales of 350 million records, he has "fallen in love with the whole package" that comprises AME: singer/songwriter/guitarist Matt Starr, lead guitarist Chris Price, bass player Jeff Covey, drummer The Max and singer/tambourine player Jodie Schell.

"Jodie's got an awesome presence onstage and an amazing, animal-like voice that can also sound cool, controlled and extremely sexy," he remarks. "When I saw Matt Starr basically running the show with his boundless energy, I knew that all I had to do was make a great record. We all know, in this day and age, that the music business is floundering. Nobody knows what they're doing, nobody knows where it's going, and everybody's so focused on so many things that nobody's focused on anything. With this band, and a number of other bands that I'm contemplating working with right now, all I need to do is focus on what I've always focused on: great songs and making a great pop record. And if I can pull that off, then no matter what the business is doing there will be a place for it.

"I believe that the old fashioned way of making pop records is still the best way, and so we cut these tracks on two-inch tape and dumped everything into the 48-track Otari Radar setup at my house where, in a Westlake-style room that I designed, there's also a 48-channel SSL K-Series console, two Otari MTR90 multitrack machines and a full-blown HD Pro Tools setup — I don't use Pro Tools, but since everybody else does, I have that to transfer stuff into my Radar. While the main speakers are custom-designed Radians, I still use [*Yamaha*] NS10s to monitor, and I recorded AME all-live just the same way as I've always done it, in the room together, and then comp'd each song from the best two or three takes.

"They know these songs back to front, and so the whole thing took a couple of weeks. No click tracks, just honest, straight-ahead rock & roll performances. That's what the business needs and that's what I'm coming back for. I work with a lot of young artists these days — the new artists are all I care about — and they're not letting anything stop them. The spirit is alive out there. They're still playing and they're still writing, and if the business could just figure things out, there are still people like me who can make great records with these kids. I'm telling you, there's a dozen bands that I've found in the last 12 months that I could put in the charts..."

Mike Chapman in his home studio in Conneticut with current project, the Automatic Music Explosion.

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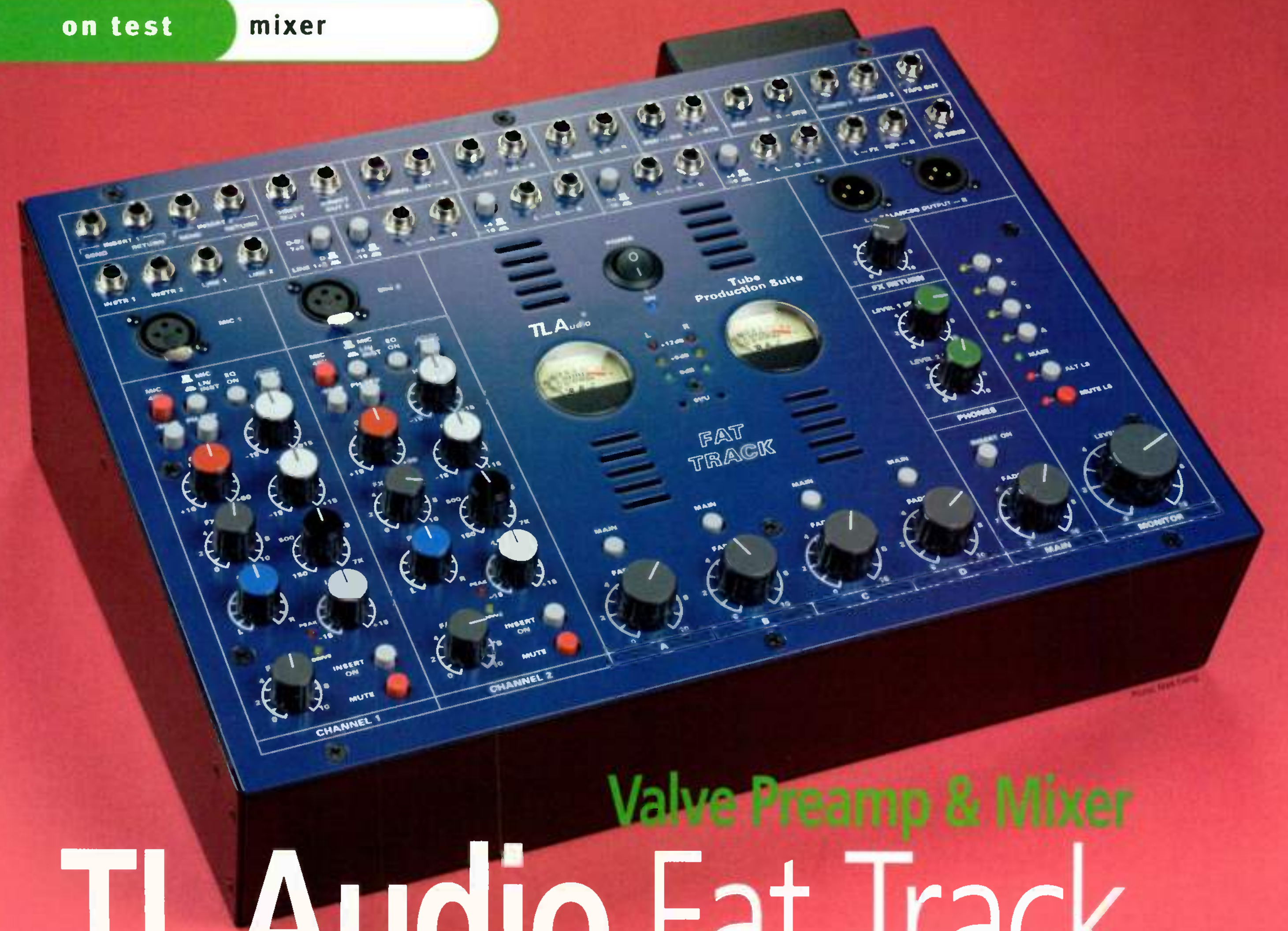

Valve Preamp & Mixer

TL Audio Fat Track

Part preamp, part summing mixer, all tube: can the Fat Track provide all the 'analogue warmth' your studio needs?

Mike Senior

Despite the promise of 'all in the box' recording offered by computer-based studios, many computer musicians still find a hardware mixer useful, providing input conditioning, monitoring facilities, and that elusive and intangible 'analogue warmth'. However, owners of small-scale studios who build up productions one track at a time will find many of the facilities on a traditional mixer a bit redundant, so a number of manufacturers now provide equipment to service these needs more efficiently — recording channels, monitor controllers and summing mixers, for example.

TL Audio's take on this concept is the Fat Track, which provides a comprehensive set of such tools for the computer user, combining them with the hybrid valve/solid-state circuitry for which the company have built an excellent reputation. In the chunky desktop box you get two full-featured recording channels, four stereo returns for source selection or summing purposes, headphone and loudspeaker monitor control, and optional ADAT connectivity for multi-channel digital interfacing.

Dual Recording Channels

The contrast between the pleasingly retro glow of the dual VU meters and the piercing hi-tech blue of the power LED will inevitably first draw the eye of the casual observer. Those with more than a passing interest, however, will soon find themselves peering closely at the left-hand side of the control panel, where the unit's input conditioning controls reside. Each of the two channels has the same facilities, comprising a mic/line/instrument preamp, three-band semi-parametric EQ, and a single aux send.

All the Fat Track's analogue I/O (almost all of it balanced) is arrayed in two rows of top-panel jack sockets, and although you might prefer your studio cable salad less proudly on display, this at least makes the socketry easily accessible. The only XLRs present are for the unit's main outputs and mic inputs. The latter have switchable phantom power and can boost mic signals by up to 60dB, although a 30dB pad switch means that you can easily accommodate super-hot sources too. Separate sockets for

> **SOUND ON SOUND**
>
> **TL Audio Fat Track £1174**
>
> **pros**
> - The sound of the valves and EQ is marvellous.
> - Forward-looking 'one-stop-shop' feature set for high-quality tracking, summing, mixing, mastering, and monitoring with your computer-based DAW.
> - Optional multi-channel ADAT interfacing.
>
> **cons**
> - The cue-monitoring options are frustratingly limited.
> - There are no talkback facilities.
> - There's no way to bypass the unit's valve stages if you want to record something more neutrally.
>
> **summary**
> The Fat Track's sound is good and will doubtless sweep some people off their feet all on its own, but more practically minded producers may find that this unit omits some operational features they can't live without.

Alternatives

I really can't think of anything that could be considered a direct alternative to the Fat Track, although there may well be other suitable products, depending on which features of the Fat Track you require.

line and high-impedance instrument inputs can be selected as the channel input via another switch, and both can access up to +/-20dB gain.

Your choice of input signal is fed, with switchable polarity inversion, to the first of the two 12AX7A (ECC83) dual-triode valve stages, which you'll see blushing coyly through front-panel ventilation slots. Two LEDs (Drive and Peak) provide an indication of how hard you're pushing the tube, and if you fancy thrashing it by cranking the Gain knob, you can safely rein in the level before recording with the Fader control that directly precedes the channel's Direct Out.

A 90Hz 12dB/octave high-pass filter follows the valve in the signal path, so that you can get rid of any low-frequency rumble (if not its valve-distortion harmonics) at source, and the audio heads from there to the same euphonic EQ circuit design used in the company's M1 Tubetracker. The fixed-frequency high and low shelves provide +/-15dB gain at 12kHz and 80Hz respectively, while the mid-band peaking filter can effect similar gain changes across a usefully wide 150Hz-7kHz range. Given that TL Audio suggest using the Fat Track's EQ for stereo bus processing, I'd have liked to see some kind of stereo link switch here. As it is you'll just have to line up the two processors as best you can by eye and ear.

A switchable insert point is provided between the EQ and fader, using separate balanced connections for send and return, and the former is half-normalled, so you could potentially use it as a pre-fader alternative to the direct out. Although there is an EQ bypass switch, I would personally have liked the option to move the insert point to between the high-pass filter and the EQ — the most logical insertion would be a compressor, and pre-compression EQ is trickier for most people to set up, in my experience. While we're on the subject, if the signal chain had been 'preamp, filter, insert, tube, EQ' you could have filtered out subsonics before they hit the tube, and you would then also have had the option of a non-tube-flavoured recording via the insert output. As it is, you'll have tube sound on everything, whether you want it or not — and on this point I share something of the concern Hugh Robjohns expressed in his *SOS* January 2007 review of the M1 Tubetracker.

After the feed to the direct out, each recording channel can be routed through a pan control to the Fat Track's stereo mix bus, whereupon it appears at three separate outputs: the main balanced XLRs, a pair of unbalanced jacks, and an unbalanced stereo TRS tape-out jack. The single mono aux bus is fed post-fader and post-mute via a separate FX send control, and there is also a dedicated stereo FX return input feeding the main mix bus directly through the FX return control.

Summing & Monitoring

The Fat Track's four remaining stereo inputs (A, B, C and D) offer nothing more in the way of control than a +4dBu/-10dBV sensitivity switch by the input sockets and a larger, grey rotary fader, and these channels can be sent to the main mix, if required, using the Main switch. The remainder of the controls govern the four sets of monitoring outputs, which comprise two headphone sockets and two pairs of speaker outputs. The Artist headphone output simply receives the main mix through a dedicated level control.

The mix feeding the rest of the outputs is configured using the A, B, C and D buttons, which can be pushed down in any combination to monitor a post-fader mix of the respective stereo returns — and if all the buttons are up then the main mix bus is monitored by default. The Producer headphone output again has a separate level control, while the large main Monitor Level knob affects both the Main L/S and Alt L/S outputs. A final pair of buttons enables you to decide which of the two sets of loudspeakers is active and mutes the monitoring completely.

On a product at this price point I was rather disappointed not to see a mono-sum switch, and I suspect that many people will find the absence of talkback facilities even more tiresome. I can only assume that TL Audio's market research has suggested that their target customers are mostly recording and monitoring in the same room, so don't need talkback.

There are a number of different ways you might actually use the Fat Track in a computer studio. At the writing stage, you could monitor the outputs of your computer alongside those of a few hardware synths and sound modules, recording the synths into the computer alongside any acoustic sources or DI'd instruments as tracking progresses.

At mixdown, you could use all 10 inputs for summing the mix in the analogue domain through the unit's second dual-triode, which is strapped over the main mix bus. Hugh was peeved at the lack of mix-bus insert points on the M1, so it's a pleasure to report that a switchable pre-fade insert point is provided here for your favourite esoteric mix squisher. Alternatively, if you happen to think summing's a load of old cobblers, you could use the four returns to reference your mix against three other stereo sources — in which case you might pass your mix through the recording channels on its way to your master recorder before routing the recorder's playback output to the appropriate stereo return.

Session Impressions

This is by no means a budget product, so it's good to feel that you're getting your money's worth even before you switch the thing on! The industrial metal construction hefts like a crate of beer, despite having a footprint no larger than an open copy of *SOS*, and the frame is surprisingly deep, standing a good six inches off the desk at its highest point. Punch a switch and it gives a satisfying click; grab a control and it travels beautifully smoothly and with a reassuring resistance.

Plug the Fat Track in, make it sweat, and you'll find that the sound amply lives up to the promise of the first impressions, delivering technical excellence with all the trappings of tube loveliness: smoother high end, rounder transients, that fraction more stereo interest, and a low-end thickness that really brings bass instruments alive. The preamps were quiet and clean when I wanted them to be, and warm and fuzzy when

Each of the two mic/line channels includes a switchable insert point, which is post-EQ but pre-fader, as well as a post-fader direct output.

I didn't, allowing each of the half-dozen mics I tried to shine in their own unique ways. The EQ is tracking-processing as it should be, delivering broad tonal changes with musicality. While some musicians might feel a semi-parametric design to be somewhat inflexible, to my mind they're missing the point of a tracking EQ — the pickier settings are best left to the mix engineer. In short, the sonics here are the main reason to buy the Fat Track, and the unusual combination of features helps improve the price/performance ratio considerably — you could pay a good chunk of the Fat Track's price for a decent two-channel valve DI or multi-channel valve summing box alone, for example.

With all that taken into account, though, any potential purchaser needs carefully to think through whether the Fat Track will actually meet their operational needs effectively. Besides the aspects of the design I've already touched on above, my main concern is to do with setting up monitoring while overdubbing, and stems from the fact that deselecting any of the returns from the main mix also defeats their feed to the monitor matrix. To see why this is a problem, let's imagine you're recording a guest vocalist. For zero-latency monitoring you'd have to switch off software monitoring in the DAW, to avoid phasing between the direct and latency-delayed vocal signals, but then if you switched the Producer headphones to monitor anything other than the main mix (in other words, exactly what is appearing at the Artist headphones output) you'd not hear the vocal you were recording.

If you used software monitoring instead, disconnecting the recording channels from the Fat Track's mix bus and monitoring the vocal coming back from the computer, the engineer could at least monitor without any reverb arriving at the FX Return inputs, by monitoring the stereo returns directly through the monitor matrix. However, there'll still be no straightforward way to create an independent cue mix for the singer, which seems to me a pretty basic requirement.

If the designers had simply fed the monitor matrix from *before* each return channel's Main switch, none of these problems would occur, because the engineer could create whatever mix he or she liked and listen to it through a stereo return disconnected from the main mix bus. As it is, though, this kind of proper cue monitoring is, for the Fat Track, currently the stuff of dreams.

Fat Is Beautiful

Despite my various operational quibbles, there can be no denying that this is a classy-sounding piece of kit that delivers a lot of different DAW support functions. What's more, it does so in a format which, while not exactly cheap, is comparatively cost-effective when considered against the alternative of putting together this feature set at this quality level using a combination of other equipment. Can you name another standard mixer design at this kind of price point that offers premium valve processing and instrument inputs? Which recording channels include built-in bus mixing? Which monitor controllers incorporate two fully-featured valve recording channels? The Fat Track really seems to be in a market of one.

However, great-sounding and innovative though this unit undoubtedly is, customers with this much cash to splash might justifiably feel entitled to expect proper talkback and cue-mixing facilities from any ostensibly one-box solution. Although I'm fairly confident that its formidable price/sound ratio will convince many musicians immediately to part with a stack of their hard-earned, I think that the less impulsive potential purchaser might sensibly bide their time a little to see if TL Audio (or, indeed, their competition), can come up with a future product combining this level of valve sonics with a more well-rounded feature set. SOS

D08 Multi-channel ADAT Interface

The D08 interface option can be installed into the Fat Track's rear panel and offers four channels of A-D conversion and eight channels of D-A conversion. The converters are 24-bit/96kHz capable, and there is Word Clock I/O too. Via ADAT lightpipe connections, you can stream audio post-fader from each of the recording channels and from the main mix outputs to a suitably equipped soundcard for recording, and can feed eight channels of audio from the computer directly to any of the 10 mixer channels. No card was shipped with the review unit, but its operation appears to be fairly well thought-out. That the price of the D08 (in the UK) adds roughly 50 percent to the overall system cost might raise some eyebrows, particularly of those who must also budget for an ADAT interface for their computer, but it's worth pointing out that it would rather defeat the point of a box like the Fat Track to compromise its sound with sub-standard A-D and D-A conversion, particularly if you're summing or mastering your entire mix through it.

information

£ Fat Track £1174; D08 ADAT card £586. Prices include VAT.
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Electronic Musician 2007 EDITORS CHOICE

REMIX TECHNOLOGY AWARD

Mix Rescue

The up-and-coming Two Spot Gobi are great on stage. Can we give their live recording the same spirit?

This month's Mix-rescuees, Two Spot Gobi, playing live at the Komedia in Brighton.

Photo: Alex Thomas

Mike Senior

When the band Two Spot Gobi recently played their hearts out for the crowd at Brighton's Komedia venue, *SOS* reader Bradley Steenkamp was on hand with his laptop-based recording system to capture the moment for posterity. Taking direct outs from the front-of-house desk, he recorded everything directly into Apple Logic 8 via the preamps of his Focusrite Saffire Pro interface and Behringer ADA8000 ADAT expander. After the gig he mixed down the project at home, using Wave Arts and PSP plug-ins, before posting the results in the busy My Sound Files area of the *SOS* Forum to get some feedback. In particular, he mentioned in his post that he'd found spill to be a real problem, and that he'd done quite a bit of work to try to reduce the levels of leakage between mics, using Logic's audio editing and automation.

I downloaded his MP3 and gave it a couple of listens, and while it was clear that he'd already produced a fair representation of what the band were about, with a sensible balance and reasonable clarity, to my ears it sounded as if the different instruments were rather disconnected from each other and occasionally rather thin-sounding — in particular the cello and trumpet parts that help make Two Spot Gobi's sound so unusual. My suspicion was that it might be possible to take a very different approach regarding the spill, leaving it in the mix and using it creatively to give a more cohesive sound, so I emailed Bradley and he agreed to send over the original multitrack files once he'd talked over the idea with the band and their management.

Initial Auditioning & Balancing

Because of the logistics of the recording setup, this song covered just 12 tracks. Electric bass and cello were both DI'd, while all the other instruments (drums, electric guitar, trumpet, lead vocal and backing vocal) were picked up by a selection of Shure and AKG mics. The drums were multi-miked, with separate tracks for kick, snare, overhead, hi-hat, rack tom and floor tom, Because of track limitations, only one overhead mic had been recorded (the left one), Bradley's reasoning being that the crash cymbal on the other side was already coming through pretty well on the hi-hat mic.

Listening to the tracks individually, one problem immediately presented itself: the front-of-house engineer had been fairly enthusiastic with his use of gating. Although this wasn't too problematic on the snare and tom tracks, the threshold of the kick drum mic's gate had been set very high, which meant that it was chopping off the front of each hit. Because of the very fast attack time of the gate, this was giving the kick an aggressive, clicky sound — something I'd originally put down to Bradley's mixdown processing choices. As it was, it became apparent pretty quickly that there was no decent result to be had from the track by processing means, so I resolved to replace the kick sound with a triggered sample once the rest of the track was in a good enough shape that I could judge what sound to choose.

With the kick track muted for now, I set about simply building up a balance of all the mic signals, in order to see how the spill components on the different mics would interact — unlike with an overdubbed multitrack production, no single mic was going to provide the entirety of any given

Some emulated tube distortion from Silverspike's Ruby Tube plug-in, high-pass filtered at 176Hz, helped to bring the bass part's mid-range further forward and saved having to EQ the bass track itself.

instrument sound here, so the only way to assess things was with all the mics up and running. While I was creating this balance, I panned the overhead, hi-hat and tom mics to give a bit of width, as well as splitting the guitar and trumpet either side of the lead and backing vocals. I also experimented with how polarity inversion would affect the sound, and discovered that, while most of the phase switches already appeared to be in the most suitable positions, the snare snapped into much better focus when the hi-hat signal was inverted.

As the rough balance came together, I began to feel that the overall sound was a bit woolly, and tracked this down in particular to spill on the lead and backing vocal parts, so high-pass filtered them at 75 and 325Hz respectively. While I was at it, I also high-pass filtered the tom tracks, to compensate for some excess 'boom' (which I imagine was due to the proximity-effect bass boost of the close mics). Additionally, the floor tom had a pronounced ring at 105Hz, which I notched out with a 12dB cut from a narrow peaking filter.

Bucking the overall trend, though, was the trumpet, which was rasping away rather unpleasantly. I combined some gentle 20kHz low-pass filtering and a 7dB peaking cut at 5.3kHz to smooth this out, shifting the focus more towards the more palatable lower registers of the sound.

Snare & Guitar Tweaks

Tickling just the strongest peaks on the snare-drum track with GVST's GMax brickwall limiter quickly returned a handful of over-enthusiastic hits to their allotted place in the mix. By contrast, adding a bit more snap to the sound proved more of a challenge, because the high shelving boost that I wanted to apply also worked on the hi-hat spill (which in this case had succeeded in getting past the FOH engineer's gate most of the time). This meant that by the time the snare was bright enough the hat was starting to sand-blast my ear canals.

I could have more heavily gated the snare track to try to get around this, but I was reluctant to, as the spill on that mic seemed to be adding something useful to the mix. Instead, therefore, I turned to the Platinumears IQ4 dynamic equaliser to boost the brightness only during the snare peaks. I set one of its four available bands to a 2.4kHz high shelf and applied the boost I felt that the snare needed to cut through the track better — a hefty 7dB. Switching this band into downward expander mode meant that the Threshold and Ratio controls could then be adjusted to pull the high frequencies down between snare hits. A fast Attack setting defeated the expansion smartly when a snare peak was detected, while the Release control was set by ear.

As is often the case with such powerful tools, dynamic EQ can be a bit fiddly to set up, but the reward in this case was a very useful extra presence to the snare, without lots of harshness in the hi-hat, so it was clearly worth the effort. That said, the hi-hat tone didn't get away scot-free, so I applied 3dB cut to a fairly narrow region around 8.3kHz on both the overhead and hi-hat mics (where the harshness was at its worst), to help preserve the status quo. As a final touch, I also added a tiny 1dB peak at 230Hz on the snare track, to increase its girth.

The next aspect of the mix that caught my attention was the guitar part, which wasn't quite delivering, lacking warmth while at the same time containing unruly high-mid-range peaks that made it difficult to balance reliably. To deal with this, I turned to GVST's GMulti multi-band compressor, splitting the signal into three bands at 430Hz and 3.5kHz. A low 1.3:1 ratio in the low band, but with fast time constants, increased the sustain of the low mid-range and reduced some cabinet 'thump', and I then boosted this band by 6dB using the make-up gain control, to achieve the tonality I was after. A slightly higher 1.5:1 ratio in the high band was just enough to squeeze the upper spectrum into a more mixable shape.

Lead & Backing Vocals

The lead vocal had been recorded with a Shure SM58, and sounded like it had already been compressed by the FOH engineer. While this processing kept the vocal levels fairly consistent most of the time, the ratio didn't quite seem high enough to control the singer's full dynamic range, and some syllables still poked out of the mix a little far. However, I'm never going to complain about someone undercompressing while recording, and it was no problem to catch the remaining peaks with GVST's GMax limiter.

This still left the sound of the SM58, which lacks the clarity of typical vocal condenser mics, so I reached for the high shelving EQ to dial in a bit of boost and realised that the track's spill conspired against this simple approach: the singer's mic was picking up not only a lot of spill from the drums, but also

The cello DI was made to sound a little more realistic with some firm EQ and a super-short reverb impulse response. The latter was too short even to provide any kind of ambience, merely adding a hint of natural resonance back into the signal.

quite a bit of audience noise, both of which didn't respond nicely to extra brightness. Firing up another instance of IQ4, I pulled a similar stunt as on the snare track, configuring the dynamics of a 5.2kHz high shelf so that the boost was much less active during gaps between the singer's phrases.

As with any high-frequency vocal boost, there's the potential for increased sibilance, and so it was here. Digital Fishphones' Spitfish plug-in is my first-call de-esser at the moment, but it introduces a processing delay that isn't compensated for by the Cubase SX 2 mix system I was using, so after I'd addressed the sibilance problem by focusing the processing on the 9kHz region, I listened to the track again, exercising the plug-in's bypass switch to check for any undue tonal shifts resulting from the change in phase relationships. Luckily the difference in this instance was vanishingly small, so I turned my attention to the backing vocal track.

Parallel compression (a compressor set up as a send effect) was used on most of the live parts, to bring up the ambience in the original recording and bind the parts together. However, the dynamics processing also brought up the high end a little too much, so Mike cut 3dB from the return using high shelving EQ, to restore a more acceptable tonal balance.

The main problem here was that the singer was moving on and off the mic quite a lot, so his level needed serious control. However, the spill on this mic was not only loud, but also pretty unpleasant — all harsh trumpet, abrasive hi-hat and boxy bass — so I knew that compressing this track to sort out the vocal levels was going to compromise the rest of the sound wherever he wasn't singing. In this instance, then, I figured it would be best to minimise the spill levels before stamping on the signal firmly with MDA's simple Limiter plug-in. Rather than faffing around with a gate plug-in for such a simple task, though, I just deleted all the spill sections using Cubase's audio editing tools.

Again, some de-essing was important here, but I used Cubase's less well-specified onboard De-esser for this, to avoid phase problems with Spitfish's latency. A little sculpting with fairly narrow 4dB peaking cuts at 300Hz and 5kHz completed the picture, but I still held the fader level as low as I could to keep the remaining spill as far out of the way as possible.

DI Processing & Kick Replacement

At this point the mic signals were beginning to work together pretty well, so I began to work on fitting the DI bass and cello parts in — the lack of spill on these signals meant that they weren't going to inter-react significantly with the miked signals, which made them much easier to deal with. The bass part was a little too dynamic to give a consistently solid low end, so I compressed at a 2:1 ratio with Buzzroom's Gran Comp, taking off around 6dB on signal peaks. Mixing it in with the other parts, I didn't feel that it really cut through in the mid-range as much as I'd have liked, so I set up a send effect to Silverspike's Ruby Tube tube-saturation plug-in, driving it hard to introduce a few extra harmonics and high-pass filtering it below 175Hz, before mixing in just enough to increase the line's audibility. This changed the timbre enough that no further EQ was required.

The cello DI was a pretty unlovable raw sound which, even after 5dB of 400Hz peaking cut and 4dB of 13kHz shelving boost had tamed the worst of the boxiness, simply needed some extra resonance to make the instrument appear more real. After a bit of experimentation, the best tool I found was a truncated impulse response (only 0.11s long!) in the SIR plug-in, which managed to breathe a bit of life back into things.

The mix was now at a stage where I felt I could make a decent judgement regarding kick-drum samples, so I used Koen Tanghe's KTDrumTrigger plug-in on the gated kick-drum track to generate MIDI notes and then loaded a few samples into Linplug's little RMF sampler until I found one that seemed to fit the bill. A bit of time spent selecting the right sound meant that I didn't need any other processing here, although I did adjust some of the MIDI note velocity values in order to match the level of the triggered kick to the drummer's original performance dynamics.

Increasing Ambience & Cohesion

All the tracks were now up and running, and although they were beginning to cohere because of the spill contributions, everything still sounded a bit dry, and there was patently work to be done in developing a suitable ambience. This is typically the point in mixing an overdubbed studio multitrack where delays and reverbs start playing a part, but as there was so much spill to play with here, I first tried putting more emphasis on that before adding artificial processing.

To bring up the ambience, I set up another instance of Gran Comp as a send effect, feeding it with a mix of all the parts I wanted to connect (snare, overhead, hi-hat, guitar, and trumpet) and then thrashing it with a 3.5:1 ratio at a very low threshold, so that it was hitting 20dB gain reduction at points. The attack and release times were also set pretty fast (5ms and 23ms respectively) and I switched in the limiter for good measure, to produce a comprehensively liquidised output, effectively killing all the peaks in order to zero in on the ambience. Although you wouldn't have wanted to listen to this signal on its own, once it was balanced back into the mix at a low level it simply faded up the overall ambience level and immediately bound everything together much more tightly.

Of course, this parallel processing inevitably changed the balance as well, but it was a fairly simple task to set that matter straight. Another side-effect was that the high end was being emphasised unduly (partly as a result of the particular compressor Character settings I had chosen), so I reined in the high end of the compressed return with a 3dB shelving cut

The guitar part needed warming up, but also had some rogue high-frequency transients which were causing problems with the balance. The solution to both of these problems came in the form of GVST's GMulti multi-band compressor, which boosted the low end and knocked the HF peaks back into place.

at 7.5kHz too. Despite this cut, the hi-hat was sounding a bit harsh again, so I decided to de-ess the hi-hat track a little to take more of the edge off the high-frequency peaks. Funnily enough, although the cello DI track had no spill on it, it turned out that sending some of that signal to the parallel compressor as well helped it sit better in the track, so I spent a minute finding the most suitable level for this.

Additional Send Effects

Having taken the inherent room sound as far as I felt I could, I was still hankering for a slightly more flattering overall reverb treatment to round things out. Pulling up another instance of the SIR convolution plug-in, I loaded in one of my favourite 1.4s impulses (which is great for adding a sense of spaciousness and ambience without an obvious reverb tail) and high-pass filtered the channel, as I normally do, to keep the low frequencies clear. The high-pass filter setting for this is something you have to determine by ear, according to the particular track you're working on, and I often find myself adjusting it at various stages as the overall mix tonality develops. ▶

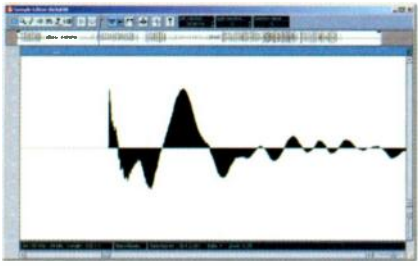
Rescued This Month...

Bradley Steenkamp works for First Choice airways as a First Officer, but spends most of his spare time working on music-related projects, including writing music for various media applications and recording in both live and studio environments. His recording of Two Spot Gobi's Komedia live show was for the company Studio Dynamic, to support a short web interview.

Two Spot Gobi are an alternative-pop band from Brighton, whose fresh and soulful sound has already gained them a huge following. Formed in 2005, the band successfully mix cello and trumpet into their own unique and commercial sound. They were recently featured on MTV and have just recorded their first single, 'Sunshine Lady', due for release on June 9th this year.

www.bradleysteenkamp.com
www.studiodynamic.net
www.twospot.co.uk
www.myspace.com/twospotgobi

Two Spot Gobi take in some fresh sea air on a sunny day in Brighton.



Here you can see the waveform of one of the kick-drum hits on the original recording. The sharp spike at the start was caused by aggressive gating during the recording and rendered the track pretty much unusable for mixing purposes.

Most of the parts went to this reverb, either directly or via a send from the parallel compressor that I'd already set up. The aim was simply to warm up and widen the sound overall, and although I made sure to keep the levels low enough that you could never hear the reverb as an effect as such, you could still hear a significant difference when the effect return was muted.

In addition to this main reverb, I also set up a simple stereo tempo-sync'd delay effect, sending to it from the trumpet and cello tracks to make them a bit more sweeping. I used a stereo effect so that I could pan the delay send to match the direct signal, such that the delay repeats matched the position of the instruments in question — I find this helps to avoid too many delay repeats building up in the centre of the stereo image where they can easily veil more important things. As both of these instruments seemed to benefit from a bit more mid-range, I restricted the frequency spectrum of the delay return by bracketing it with filters at 100Hz and 3.2kHz, and then also sent from the delay channel to the main reverb to distance them a little more.

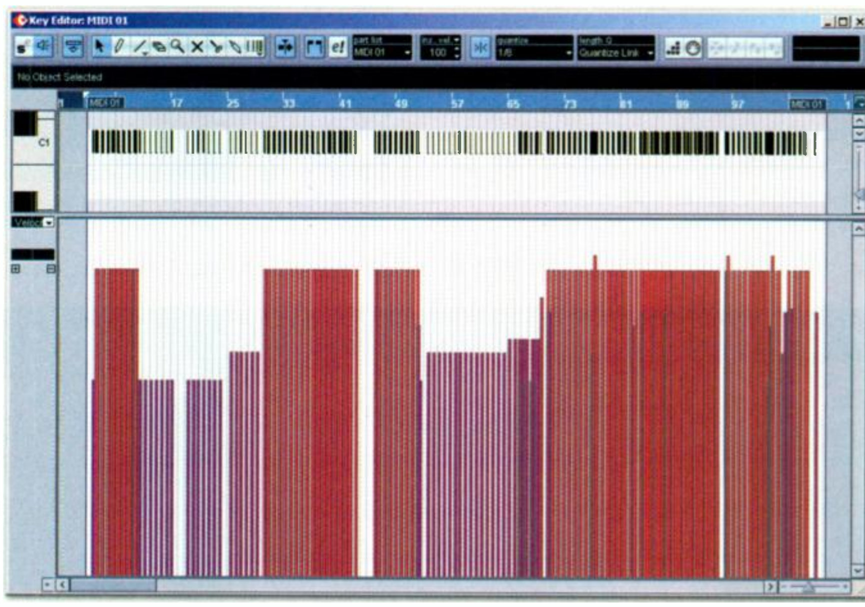
I was reluctant to add too much in the way of reverb to either the vocal or bass parts, for fear of clouding them and reducing their impact, so for these I added slightly different effects instead. For the vocals, I chose another ridiculously short, bright impulse response from SIR, which widened and blended the vocal in equal measures, and then supplemented that with a hint of 'venue wall' using a single de-essed 60ms slapback, brightened with a high-pass filter turning over at 450Hz. Again, I fed some of this return back to the main reverb. The bass benefited from some of the short vocal impulse as well, but I added in a fair amount of stereo chorusing too, because the bass sound seemed too tightly focused in the centre of the stereo field.

Remix & Re-remix

When I was satisfied that I had a pretty solid mix balance together, I emailed Bradley to get his feedback. Although he liked the new sound overall, he felt that I'd left the venue sounding smaller than it actually was — a fair point, as I'd subconsciously been envisaging a pub gig, and Komedia's is no pub! With this in mind, I revisited the mix and added in a richer-sounding recital-room reverb, chaining it with a tempo-sync'ed stereo delay for some additional sustain — as is sometimes the case, I found an impulse response that had just the right tonality but simply wasn't quite long enough. This new reverb was added to the guitar, trumpet, cello, drums, and lead vocals to increase the apparent size, and with hindsight I'd agree with Bradley that the final mix is all the better for it.

While I was at it, I also tweaked a few other things which still bugged me a little: the guitar still had a slightly nasal sound, which I adjusted with a small EQ cut at 1.7kHz, and the lead vocals were disappearing behind the backing parts on the odd occasion, requiring a little level automation to keep them audible throughout. In addition, I noticed that the levels of hi-hat spill on the lead vocal mic were upsetting the drum balance for a couple of sections, so I pulled down the vocal fader for these sections.

Bradley had suggested that I should contact the band directly too, so I sent them the remix to see what they felt about it — and initially received a somewhat lukewarm response! Although they liked the way the individual sounds had filled out and blended together in general, they mentioned that the trumpet was getting in the way of the vocals



This screenshot from Cubase SX 2's Key Editor shows the MIDI notes which triggered the replacement kick-drum sample in the remix. Notice that the note velocities (and hence the sample's volume) have been edited to mimic the changes in the drummer's playing during the song's different sections.

Remix Reactions

Bradley Steenkamp: "I posted a link to my mix on the *SOS* forums for people to comment on and got very positive feedback. Most people (including me!) seemed to agree that the level of the trumpet was a little low, but in addition I found crowd and instrument spill to be an irritation. (Thanks to Bryan, Cailen and Geoff at Studio Dynamic and also Steve Richer for all the feedback!) I think it's very interesting to find out how different people tackle a mixing project, so when Mike got in touch with me about remixing the song for Mix Rescue, it seemed like a great opportunity to learn something from a master at work!

"I am very impressed with the finished mix. It is far more intimate and the levels of the trumpet, cello and guitar are better balanced. Mike has embraced the spill and used it very effectively as part of the live sound. This is something I have avoided so far but will keep a more open mind about when mixing in the future, as it seems to add to the feel of the song. I also much prefer the sound of the bass guitar and kick drum."

Two Spot Gobi: "Overall we are very happy with the remix. The main improvement for us is that the instruments sound a lot more natural and less EQ'd, especially the cello, where you've managed to widen the spectrum of sound so that we can really hear the bass frequencies that usually suffer as a result of it being DI'd for the live setup. The frequencies of the trumpet have also been fully captured, opening out the sound a lot more and adding to the live feel of the song. The lead vocal doesn't get lost in the mix, which is really important for our sound, and it gels well with the backing vocals. We've found that the bass also often gets lost in live recordings, but that comes through nicely too. Thank you!"

at times, and that the overall tonality was bass-light and tending towards harshness.

Calling up the mix again and referencing it against the band's soon-to-be-released single, which they'd attached to their email reply, it soon became clear that Bradley's view of the way they should sound (on which I'd based my mix) was different from the band's, which meant that I'd been aiming for the wrong target! Fortunately, rectifying the situation to the band's satisfaction turned out to be fairly straightforward: just some level automation for the trumpet, a bit more bass and kick in the balance, and a dose of corrective buss EQ (gently lifting the low end below 200Hz, pulling down an octave-wide region around 1.5kHz, and adding some air at the top end).

Thrills & Spills

I hope that I've been able to demonstrate how the acoustic spill between different mics in a live ensemble recording can be your friend, providing an 'ambient glue' (to use producer Tony Platt's term) that holds the production together. The other advantage of going easy on the gating is that more of the information about each instrument is captured, and that can really help fill out the sound — if you listen to some of the audio files on the *SOS* site, you'll hear how some parts sound quite different just through their own mic than they sound in the context of the mix, simply because of the spill contributions from all the other mics. Admittedly, you sometimes have to think laterally to get the control you need over individual sounds despite the leakage, but that can actually be easier in the long run than having to use mountains of processing first to remove the spill, and then to artificially re-establish an organic-sounding connection between the different instruments. SOS

Platinumears' IQ4 dynamic EQ plug-in came in handy on a couple of occasions where the wanted signal needed brightening, but not the spill: the instance pictured here shows the settings used on the snare, but it also came in handy for the lead vocals.

Hear The Changes On-line!

You can listen to the original recordings and Mike's remix, along with some of the treatments given to the different elements of the mix, on the *SOS* web site at www.soundonsound.com/sos/jun08/articles/mixrescueaudio.htm

Olympus LS10

Portable Stereo Recorder

The latest entry into the world of portable stereo recorders comes from a company better known for their cameras and accessories.

Sam Inglis

The first the world knew of the LS10 was a high-profile launch at the NAMM show in January, where everyone I talked to came away from the Olympus stand saying something along the lines of "That looks like a classy piece of kit." It's an impression that is only reinforced by looking at the review unit, which simply oozes quality.

Olympus are a huge name in photography, and also make a range of dictation machines and other recorders aimed at those with a need to record speech, but the new LS10 is their first foray into Sound On Sound territory. It's a portable, solid-state, stereo recorder with built-in microphones, which presents an affordable alternative to the revered Sony PCM-D1 and competes more directly with the likes of the M-Audio Microtrack II, Edirol R09 and Zoom H2 and H4. (If you're interested in comparing the specs of portable recorders, take a look at the charts on the Wingfield Audio web site: www.wingfieldaudio.com.)

On The Outside

I was going to say that the LS10 is about the size and shape of a mobile phone, but then I remembered that most *SOS* readers have probably bought a new mobile phone within the last eight years. Even so, it sits comfortably in the hand, and would fit fairly easily in most pockets. A soft padded case is included. The metal casing is light but feels sturdy, and the controls are all easily accessible, yet Olympus have given serious thought to ensuring that they won't get activated accidentally — the thumbwheels for playback volume and record level are protected by unobtrusive bulges in the casework, while the power switch has to be slid and held for a couple of seconds to turn the unit off. The backlit LCD is clear, and it is also much larger than, for example, that of the Edirol R09. Other nice touches include a mounting for a standard camera tripod, and neat little foam windshields that clip to the heads of the microphones.

The microphones themselves are fixed in an X/Y configuration, although some electronic jiggery-pokery described in the 'Virtual Microphones' box can be used to vary the apparent width of the stereo field. A mini-jack headphone socket doubles as a line-level output and is complemented by a small pair of speakers, permitting you to check how your mixes translate to being played through a teenager's mobile phone on the top deck of the number 73 bus. The LS10 is clearly designed to be used in conjunction with a computer, as there is no dedicated line out or S/PDIF digital output. I didn't find that I missed them, but others might.

A remote control kit is an optional extra, as, unfortunately, is the 5 Volt power supply. Since the LS10 is equipped with a USB connector for transferring files to and from a computer, I was rather hoping it might be equipped with internal batteries that could be charged, iPod-style, over USB, but unfortunately this isn't the case. However, two AA batteries are supplied, and the manual quotes battery life as ranging from eight hours to 35 hours, depending on what kind of batteries you're using and what you're doing with the LS10 — recording is more battery-intensive than playback. These figures compare extremely well to most of the LS10's competitors.

Simple Yet Effective

There seems to be some law of product design whereby those manufacturers who are thoughtful enough to create

massively detailed manuals are also clever enough to create intuitive products that don't require massively detailed manuals, and so it proves here. The LS10's extensive printed documentation is actually far more intimidating than the product itself, which is about as easy to use as you could possibly hope. This is not something you can say for all its rivals.

The LS10 is equipped with a fairly generous 2GB of built-in, non-volatile memory, and has an additional slot for Secure Digital (SD) cards; the manual doesn't say what sizes are supported, but only lists recording times for 8GB and below. Uncompressed PCM data can be recorded at up to 24-bit, 96kHz, or you can record at three different bit-rates in either MP3 or WMA format. The internal memory and any SD cards used are formatted in such a way as to contain five folders for recorded audio, plus a Music folder. There's no obvious way to change the default structure, but it seems perfectly sensible to me, providing enough options to keep unrelated recordings separate whilst keeping operation very simple. When you hook the LS10 up to a Mac or PC it appears on the desktop just like any other removable storage device, and I had no problems opening up these folders and dragging things in and out of them. Thanks to the 'high speed' USB2 interfacing, even large files transfer quickly. Simple file-management tasks such as erasing and moving files are handled straightforwardly, and if you power down during playback, the LS10 will pick up seamlessly when you switch it on again.

The Music folder allows you to treat the LS10 like a conventional MP3 player, should you wish to. You can drag and drop albums of MP3 or WMA files from your computer into this folder, or synchronise it with Windows Media Player, and all names are displayed correctly.

The folders and the menus are brought up on the LCD by hitting the List button, and navigated using a conventional arrangement of cursor keys around a central Play button, and there are dedicated buttons for Stop and Erase. There's also an A-B Repeat button, which allows you to set up simple looped playback within a recorded file, should you wish to do so. The Menu button does precisely what you'd expect a button labelled Menu to do, while a neat addition is a single Function button that can be assigned as a shortcut to various different menu items; so if, for example, you find yourself often needing to switch between recording WAV and MP3 files, you can set this button to take you directly to the relevant submenu.

SOUND ON SOUND

Olympus LS10 £269

pros
- Very intuitive and easy to use, yet offers lots of functionality.
- Good sound quality from the built-in mics.
- Excellent physical design and build quality.
- Long battery life.

cons
- Doesn't record timestamp information, markers or other metadata.
- Can't be recharged over USB, and mains power adaptor is a cost option.
- No dedicated line-level or digital output.

summary
Stylish, well-featured and above all intuitive to use, the Olympus LS10 is a most impressive entry into the world of serious audio recording.

The top level of the menu structure is displayed as a vertical list of symbols down the left-hand side of the LCD, while the main area displays a submenu corresponding to whichever of these is selected. Selecting one of these items brings up a further submenu, which will usually allow you to select between three or four well-chosen settings. In case I've made that sound more complex than it is, let me say again that the LS10 is impressively easy to use. Although there are lots of menu items in total, the structure is very well thought-out and intuitively arranged, and everything is easily accessible with only a few clicks.

Making Tracks

Recording on the LS10 is simplicity itself. You hit the Rec button once, it flashes, and the display shows a large level meter to augment the unit's red LED Peak light. Adjust the Rec Level dial and Mic Sensitivity switch to suit (or switch it to Auto level), press Rec again and you're in business. It's possible to connect external mics or other input devices via Line and Mic mini-jack sockets, and 'plug-in power' for mics is available (as opposed to phantom power, which is not). I didn't feel the need to attach an external mic in any case, as the quality of the inbuilt units is very good indeed. I'm not sure there are many circumstances in which I'd bother recording at 24-bit/96kHz with this sort of device, but the mics and other circuitry are quiet and clean enough to capture even fairly distant, ambient sounds without adding noticeable self-noise.

The battery compartment is found on the LS10's rear panel, where there's also a camera tripod thread and some small playback speakers. The mic and line inputs, a high-pass filter and mic-sensitivity and recording-level controls are found on the right-hand side.

Virtual Microphones

Perhaps by way of compensation for the fact that its stereo mics are fixed in position, the LS10 employs a technology called DiMagic Virtual Microphone to "record by focusing on sound from any direction". This can only be used when recording 16-bit, 44.1kHz WAV files, and is activated by selecting one of five options from the Zoom Mic menu (can you tell that Olympus make cameras?). At one end of the spectrum there's Wide, which makes the apparent width of what you're recording much broader. I found this very effective for some ambient sounds such as passing traffic, where you want exaggerated movement within the stereo field, but I'd be hesitant about using it for music: there's an obvious hole in the centre of the field, and recordings made with the Wide setting are not at all mono-compatible. At the other end of the spectrum, the Zoom setting records mono, but somehow matrixes the two microphones to produce a more directional pickup pattern. DVM settings can't be changed or undone after the fact, so it's probably best to leave them switched off unless you're very confident they can deliver the results you're after. The LS10 also offers basic reverb and 'Euphony' settings that can be applied on playback, but these don't get recorded.

The high-quality built-in mics are fixed in an XY stereo pattern, but a neat 'virtual microphone' system offers more options than you might expect.

The fixed X/Y configuration is not as flexible as some of the LS10's rivals, although the DVM technology (see box above) adds useful options. For ambient recordings, I found that it delivered a convincing and reasonably wide stereo field, although it seemed to me that the high end was attenuated a little when I converted my field recordings to mono. The supplied windshields are pretty effective, but you do need to be a little careful how you hold the LS10 to prevent handling noise being picked up. A 200Hz high-pass filter is switched manually using a slider control, while there's also a digital limiter that is activated by a menu setting.

Each of the LS10's folders can contain up to 200 recordings, which are given the names 'LS100001', 'LS100002' and so on. The LS10 does have a clock built in, and when you transfer WAV files to your Mac or PC, the 'Date Modified' field in the Finder or Windows Explorer shows the time and date of recording, but the files are basic rather than Broadcast WAVs, and don't appear to contain timestamp information or other metadata. I was slightly surprised by this, although it's unlikely to present a big problem for most music applications. A more serious omission, as far as I'm concerned, is that it's not possible to embed markers during recording, as you can with rivals like the Zoom H2 and M-Audio Microtrack II. This facility would be very useful when making long recordings, such as of a band's live set.

Conclusions

Although it's competitively priced, the Olympus LS10 is not the most affordable product in its class: that honour goes to the Zoom H2. With its fixed mics and lack of support for markers or Broadcast WAV features, you could also argue that other recorders outgun it slightly in the 'bells and whistles' stakes. But as far as I'm concerned, unless you really need timestamping, markers, or an S/PDIF output, the LS10's plus points massively outweigh any negatives.

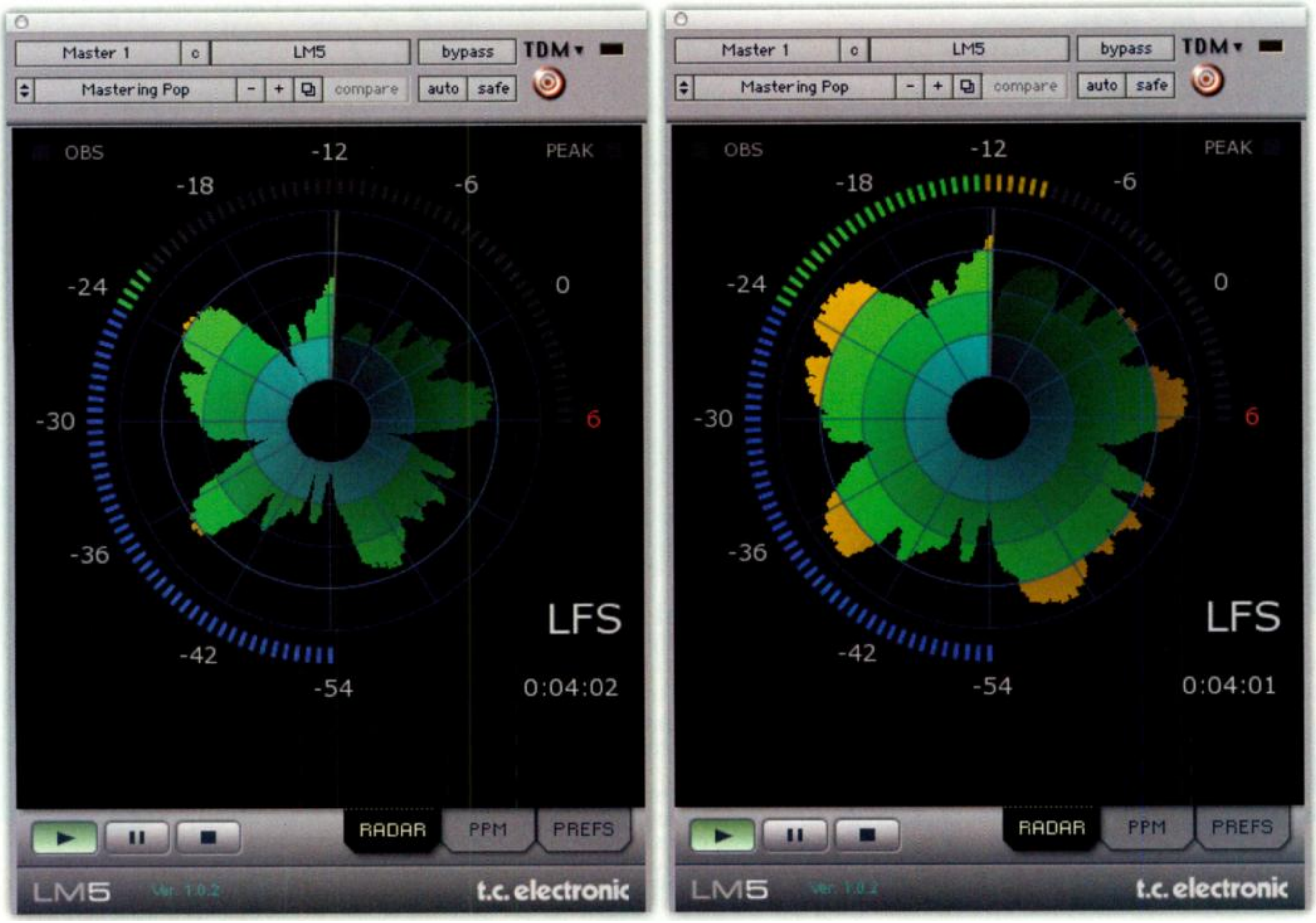
The most important of these positives are its sound quality, simplicity and ease of use. Unlike some of its rivals, this is a product you can pick up and master within a few minutes, thanks to an extremely well thought-out user interface and control layout. For my money, it also trumps the competition in terms of physical design and construction: if you're prey to gadget lust, be prepared to whip out the credit card when you try one of these out! All in all, a very neat piece of kit, and I hope Olympus have more ideas in store for our market. SOS

information

- £ £269 including VAT.
- T Olympus UK +44 (0)1923 831100.
- E info@olympus.uk.com
- W www.olympus.co.uk

TC Electronic LM5

Loudness Metering Plug-in For Pro Tools



The same programme material before (left) and after mastering. Peak levels are the same, but the mastered version is louder overall, as well as being more consistent in terms of loudness.

TC Electronic have created a Pro Tools HD plug-in that can measure the loudness of programme material and its suitability for different playback systems.

Mike Thornton

I am sure that we have all complained at some point about how loud the adverts are on TV compared with the drama or documentary they are placed in. Most of you will have realised that the adverts have been processed with various compression techniques to improve their perceived loudness, with the desired effect of having that ad 'stand out' from the rest. All broadcast content, both programmes and adverts, has the same maximum peak signal level, but that doesn't stop some of it being subjectively louder than the rest.

However, creating a meter that will measure loudness has always been difficult, as loudness is a subjective assessment in which sound pressure level, frequency content and duration are all important factors. This has made it difficult to quantify, and a meaningful generic loudness measurement is possible only through large-scale subjective tests on a number of subjects. In 2004, McGill University, in conjunction with TC Electronic, undertook such tests, while the International Telecommunication Union (ITU) have been investigating 'Audio Metering Characteristics suitable for use in Digital Sound Production' for the last six years or so. As a result of several rounds of trials, listening tests and review, a draft standard called ITU-R BS.1770 was drawn up, and we are starting to see a number of loudness meters coming onto the market supporting that standard. One of these meters is a Pro Tools TDM plug-in: the LM5 Radar Meter from TC Electronic.

Why Radar?

In order to be able to judge loudness consistency over time, any meter is going to need some form of histogram as part of its display, as well as an instantaneous loudness indicator and a conventional peak meter. TC Electronics have resolved these requirements with a graphical interface that works like an aircraft radar system, with a rotating cursor that leaves a trace behind to show the loudness history. The use of a radar-type display is an inspired piece of design, and the colour coding means that a quick glance at the display will show not only the instantaneous loudness and historical loudness but also the consistency of loudness through the track or programme. Consistency is easy to judge: you look at the height of the 'hills' and 'valleys' in the display.

The two screens, left, show the same mixed track. The one on the left is the track as I mixed it, and clearly has a wide dynamic range but is relatively quiet. The second screen shows the mastered version, and you will be able to see two major differences. The first is that the whole track is louder: this is significant, especially when you realise that I had normalised the original mix so both files have the same maximum peak level of -0.01dBfs. The second difference is that the distance between the 'hills' and the 'valleys' is much less, so the second version can be described as more 'consistent'.

Consistency in volume has come to matter a lot in the digital world where it is technically possible to deliver wide dynamic range content, but there is arguably little point in doing so if the available dynamic range of the consumer's listening environment is restricted. TC's research has shown that consumers have a distinct 'dynamic range tolerance' (DRT) which is specific to their listening environment, as the graph above shows. The DRT is defined as a Preferred Average window, with a certain peak level headroom above it. The average

SOUND ON SOUND

TC Electronic LM5 1174 Euros

pros

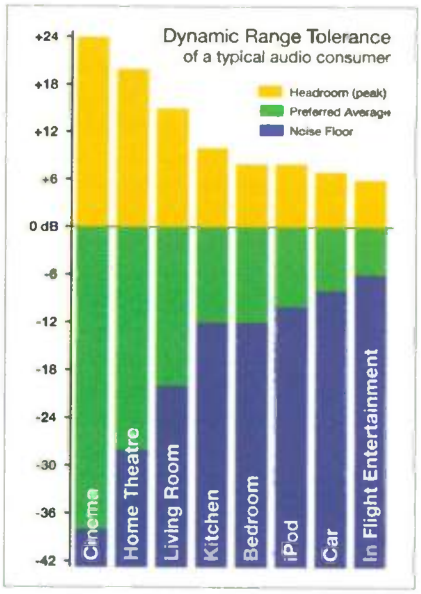
- At last, a reliable measure of loudness in a plug-in.
- Well-designed interface and display make it easy to take in lots of information.
- Allows you to tailor the dynamic range of your material to its intended playback systems.

cons

- Only available for Pro Tools HD.

summary

A very well-designed plug-in that is useful in a wide range of circumstances and projects, both for post-production and music mixing.



TC's graph showing listeners' tolerance of dynamic range in a variety of playback environments.

TC ELECTRONIC LM5

sound pressure level, which obviously is different from one listening condition to another, has to be kept within certain boundaries in order to maintain speech intelligibility, and to avoid music or effects from becoming annoyingly loud or soft.

As audio engineers we instinctively target a certain DRT profile when mixing, but because level normalisation in broadcast and music production is based on peak level measures, material with low dynamic-range signatures ends up the loudest. Audio production can therefore be trapped in a downward spiral, going for loudness at the cost of ever-decreasing dynamic range.

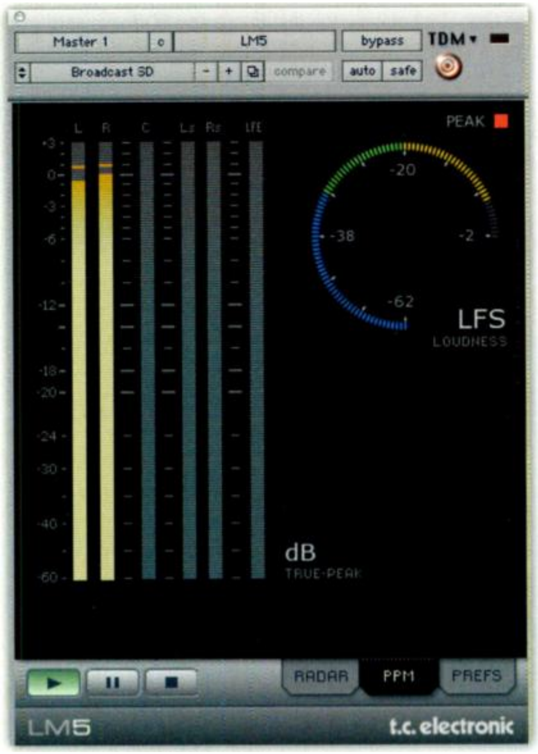
In line with this research, you can use the Preferences page to set up LM5 to suit a variety of playback systems, and it colour-codes its loudness information to help identify a target level range (green), levels that are below the noise floor of the listening system (blue), and loud events (yellow). A selection of factory presets covers a number of common applications, such as pop mastering, as well as EBU and NAB settings, and you can also create your own presets. The scale of the radar meter can be set to either 'Loudness Units, LU' or 'Loudness Full Scale, LFS', while the '0 LU Equals' parameter sets the loudness required to obtain a 12 o'clock reading on the outer ring. This is the same as the border between green and yellow on the Radar page, while 'Low Level Below' determines where the display will change between green and blue, indicating levels at risk of being lost below the end user's listening environment noise floor.

How It Works

The outer ring of the LM5 Radar display is the Current Loudness indicator, which shows an instantaneous reading of loudness at the present time, with the 12 o'clock position representing 0LU, and the trace is colour-coded to show where in the DRT the loudness is sitting.

The radar display inside the ring shows a loudness 'landscape' that is designed to let you judge whether loudness emphasis is put where it is required. In film, are dialogue segments balanced against action scenes, or music segments? In music, does the chorus of a song have a push against the verse? In TV is the audience too loud in a game show? All are questions that are designed to be answered by the LM5 radar display.

The duration of one radar revolution can be set between one minute and 24 hours, with reference either to the real-time clock of the computer or Programme Time, which starts when you press the Play button on the Radar or PPM pages. You can set the loudness scale to 3, 4, 6, 8, 10 or 12



LM5 also offers 'true peak' metering, which can predict cases where the reconstructed audio signal will clip the D-A converter, even though no individual sample hits 0dBfs.

dB between each concentric circle. The 0LU point is always marked as the border between green and yellow at the bold concentric circle. Transport controls in the bottom left-hand corner are used to run, pause, stop and reset the radar. The OBS indicator lights to show certain inter-channel anomalies, and if it lights TC recommend the user checks the PPM page to see what might be causing the anomaly. The Peak indicator shows that at least one channel is exceeding its true-peak maximum.

PPM Page

The PPM page is a 'true peak' level meter, which can be used to visually monitor the balance between channels and their headroom. It shows conventional bar-graph signal PPM meters on the left side, with a round Current Loudness display, identical to the outer ring of the Radar page, on the right-hand side. The peak meters display true-peak level as specified in ITU-R BS.1770. True-peak meters are designed to give a better indication of headroom and show the risk of distortion in downstream equipment such as sample-rate converters, data-reduction systems and consumer electronics.

What the true-peak meter is displaying is what will happen when the samples are converted back into the analogue domain: it is possible that although no one individual sample goes above 0dBfs, a pattern of samples could produce an analogue waveform that will exceed the 0dBfs point and so potentially cause distortion, especially in consumer equipment. TC Electronic recommend that for data-reduced delivery via compressed formats like MP3, -3dBFS should be regarded as maximum level. For more information, the TC Electronic web site features a Tech Library where you can download some very informative articles.

In Use

At the time of writing this review I had two TV documentaries to mix, and I found LM5 invaluable. At first, while editing and setting levels, I had the Radar set to one minute per revolution so I would get relatively quick feedback as to the relative loudness of each clip. When it was time for the final mix, I changed the Radar speed to one revolution per four minutes, and found it ideal to keep a check on the loudness consistency throughout the programme.

In both cases, it was very useful indeed, but it's important not to allow your decisions to be completely controlled by LM5 — or any meter, for that matter. These devices should be considered as tools to help produce a better sound. Don't forget the best arbiters of what sounds right are the two devices we have bolted to the sides of our heads!

Overall, this plug-in is an excellent piece of technical and interface design and is long overdue. The radar-style meter is inspired and makes the plug-in so easy to use, with the colour-coding of both the outer loudness ring and the histogram radar section meaning that you don't need to clog the display with loads of numbers. To know that the loudness rating is based on large-scale trials by both TC and McGill University is very reassuring. It would be great not only for use in the broadcast audio sector but in mixing music, to help show which parts of a song are louder. LM5 isn't only useful for producing loud mixes: it also helps you to produce mixes with a wide dynamic range safely and reliably. Finally, for mastering, it would be invaluable in attaining a consistent loudness thoughout an album project. SOS

information

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- **W** www.tcelectronic.com

Daptone Studios

Gabriel Roth: Recording For Daptone Records

Dan Daley

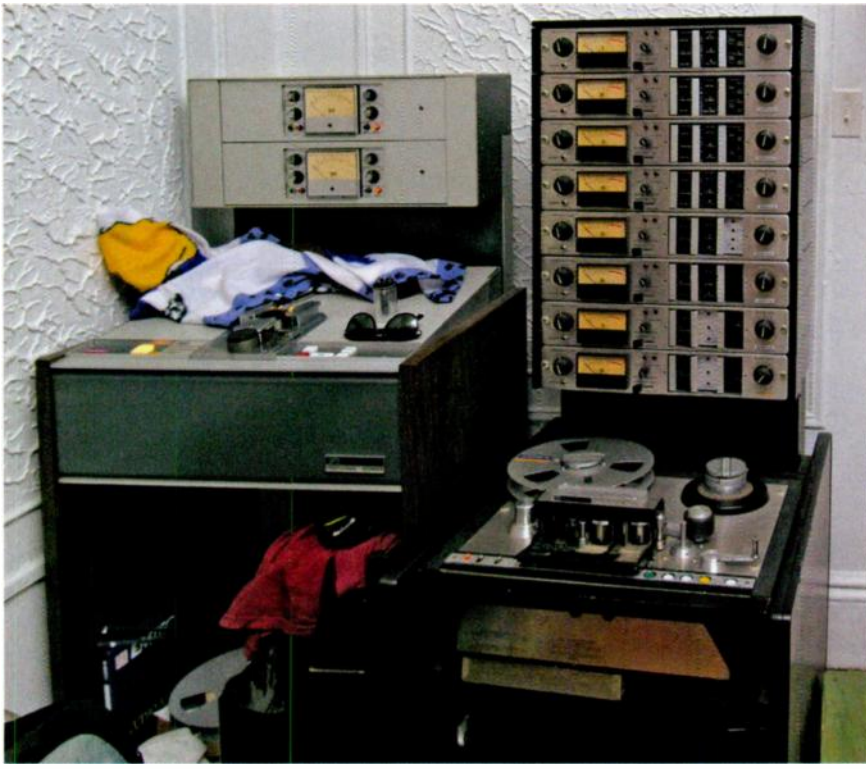
The 'vintage' sound of artists like Sharon Jones & The Dap-Kings and Amy Winehouse has little to do with super-expensive valve gear. Instead, it's the minimalist approach of house engineer Gabriel Roth that sets the tone.

Gabriel Roth is no ideologue. He says, flat out, "Show me a computer that sounds as good as a tape machine and I'll use it." Gabriel is the audio architect of the evocative Motown/Stax-infused tracks behind Amy Winehouse, as well as those for the extended musical family for whom Daptone Studios and Daptone Records are the hub, including Sharon Jones & The Dap-Kings, the Daktaris, the Soul Providers and the Sugarman Three, the last eponymously named for Gabriel's partner in the label and studio.

When you hear Winehouse's 'Rehab' or Jones' 'Nobody's Baby' you experience a shift in time and place — you're suddenly in Detroit in 1965 or Memphis in 1962. In reality, you're in 21st-century Brooklyn, in a ramshackle house in the Bushwick neighbourhood, one of the last holdouts against the decade-long gentrification of that borough. Daptone Studios occupies a floor in the rambling residence that Gabriel and the Daptone collective have remade into a haven for that sound. However, Roth bridles a bit at the term 'retro'.

Gabriel Roth at the Trident desk in Daptone Studios.

The Daptone building in the Bushwick neighbourhood of Brooklyn.



Increasingly, Daptone recordings are made to eight-track rather than 16-track.

"We're not doing a purposely 'retro' thing," he says. "It's not about an ideology. Some people say that there hasn't been a good record made since 1973, and I pretty much agree with them most of the time. But I'm not listening to the old soul records and taking them apart clinically. It's more like a kind of informal schooling. You listen — listen for where the horns bite and the crackle of distortion on a vocal. You don't want to imitate it; you want to let it influence you. One of the things you learn is that sometimes mistakes are what make a track sound great. Music should not be perfect or correct. When we play and when we record, we're looking to find what makes us feel good. We're steeped in those old records, but we're not consciously trying to remake a record from 1962."

You certainly cannot call up Auto-Tune to fix any clams on a Daptone track. For starters, it's a tape-only proposition, and Roth seems to be moving joyfully backwards in time: the Tascam 16-track deck in the control room increasingly gives way to an Otari MX5050 eight-track half-inch machine. From the Trident Series 65 24-input desk, mixes go to either a nicely restored 3M M23 quarter-inch two-track deck or a somewhat scruffy Otari MX5050 quarter-inch machine. Both of these are also called into service to provide authentic tape-slap and echo effects, augmenting a pair of Orban spring reverbs and a Stocktronics plate reverb.

"There's an old-school feel to the studio, but it's by no means some kind of vintage museum," Roth states. In fact, the gear list is as much serendipitous as it is calculated: there's a newish Rode NT1A large-diaphragm condenser mic, a classic RCA DX77, and an assortment of cheap Radio Shack microphones cohabitating in the mic closet, while an Ampeg Gemini guitar amp made its way to the studio when it literally bumped into Roth as it was being tossed out into a dumpster near a building renovation. Nothing is here because it's cool; everything that is here is here because it sounds good for something.

Collective Studio Building

The current Daptone Studios (aka House of Soul) is the third incarnation, roughly gauged. Gabriel Roth was partner in an earlier record company, Desco Records, which had a small studio on Manhattan's Lower East Side and another later on West 41st Street. When he formed Daptone Records with new partner Neal Sugarman, they put together a studio in a sublet basement space in Williamsburg, Brooklyn. That disappeared when the primary tenant was evicted. They moved the recording gear to Sugarman's apartment.

The first couple of Dapco releases sold reasonably well and they were informed that they had royalties of $30,000 coming to them from the distributor. "Knowing that was coming, we signed a long lease on this house in Bushwick, maxed out our credit cards and borrowed money from other sources and started building the studio," says Roth. Naturally, this being the music business, they never saw the $30,000. (The distributor's rep collected the money and promptly declared bankruptcy.) "So we ended up building the whole place ourselves," says Roth.

Members of the collective applied whatever skills other than music they possessed: Daptone collective member Charles Bradley knew plumbing; Sharon Jones and Roth actually ran electric mains and did the grounding; Roth's father helped them float the isolation booth's floor using old tyres found out in the streets (along with New York's legendary giant subway rats, another seemingly apocryphal legend is actually true: on any given day in New York, you can furnish an entire apartment from stuff people leave on the street); and Roth's mother sewed acoustical curtains at home in Riverside, California. "And the Budos Band tuned out to be very good at kicking down walls," he notes. "It was a long cold winter that year, but we got it built."

Don't Let It Become A Formula

Those who peruse these pages in search of granular descriptions of complex recording techniques won't find much to chew on at Daptone. On the other hand, you do learn that you can make quite a lot out of a very little. With the caveat from Roth that he doesn't rely on formulas for capturing any instrument, his basic approach is 'less is more'.

Drums usually get one or two microphones, and that's about double the number everything else gets. The RCA DX77 or the Shure 55 often goes on the floor next to the bass drum in such a way that it picks up the snare, as well. "From the drummer's point of view, if you looked down between the snare and the kick drum, you'd see it about a foot or two away from the snare," he explains. "The second microphone is often in the same spot as the first but adds different frequencies. Sometimes the ▶

DAPTONE STUDIOS

Gabe Roth's favoured single-mic placement for recording drums, with a Shure 55 between kick and snare; above, from the drummer's point of view.

one and only mic is over or behind the drummer's head. Sometimes the only mic is a Radio Shack dynamic."

In one scenario from a Winehouse session, he placed both a DX77 and a Shure 55 close together on the floor, then boosted the high-mids on the DX77 and cut the low-mids, around 400Hz, on the 55, resulting in less definition but way more chunk. "The trick isn't how many microphones you use, it's where you place them," the former audio student at New York University instructs. "One of the big problems with modern engineering is everyone telling everyone else how they put this microphone here and another there, and you have to use this condenser for overheads and this large-diaphragm condenser for the kick, and so on. Once it becomes a formula, people stop using their ears." This minimalist approach explains one of the classic characteristics of the great soul records of yore, which is how the tom fills seem to have a perfectly smooth decrease in volume as they increase in density: the fills move in a direction away from the lone microphone. (Listen to the fills on Winehouse's 'Rehab'.)

Roth admits that on the rare occasions when non-Daptone players encounter how he works the studio (it's not for hire), he receives some arched eyebrows. But he recalls a session with famed drummer Bernard Purdie in which he used only a Radio Shack dynamic mic placed overhead and the Akai 'Dictaphone' mic (as Roth refers to it; according to Bob Paquette, owner of the Microphone Museum in Milawaukee, Wisconsin, it's probably an Akai MC50, made in the '60s to pair with home tape recorders) on the kick. The microphones were EQ'ed and compressed together ("When the kick hits the compressor, it needs to step on the cymbals"), and both microphone channels were sent to a single track. "He said he hadn't heard that sound in a long time," Roth recalls after the playback in the tiny Daptone basement control room. "He liked it so much he said he'd come back sometime and do another track for free."

Roth is similarly minimal with the horn sections: a single microphone, often the workhorse Shure 315 or 55 (the "Elvis" microphone), is used to collect the three Daptone Horns, though sometimes he'll go to the '80s-era Radio Shack microphones, placed on a stand about four feet in front of the section. "What you want to do with horns is let them mix themselves," he says. "Give them enough room for the sounds to blend before they hit the microphone. The sound you want really is coming from the musicians, and when guys have played together for a while it's not a strain to get a good sound." On the album of instrumentals by the Budos Band, one of the Daptone Records artists, a Radio Shack condenser microphone was added on the baritone sax. "We wanted to pan the bari on one side and the trumpets on the other," says Roth.

Guitars, as you might expect, also get a single microphone, often a Shure 57, placed very close up on the speaker. In fact, while working on a gospel album recently, a shortage of tracks compelled Roth to place a single ribbon microphone in between the bass and guitar amps, which he then proceeded to record to a single track on the eight-track deck. This works out to 0.5 microphones per instrument. "No tricks — just good guitars played by really good guitar players through really good amps," he says.

Back in the control room, Roth employs other techniques to get the classic soul sound. "I roll some low end off the tracks, around 80Hz to 100Hz, before they get to tape," he explains. "The lowest frequencies

The Sting, Dap Style

If you want to viscerally comprehend the allure of the kind of sound that Daptone creates (or recreates, if you prefer), consider this tale related by Gabriel Roth. "When we started Desco Records, we did vinyl releases only, and we recorded records with the great soul music sounds. But it was definitely a very small niche market. So we made up the story of an old king-fu movie from the '70s called *The Revenge Of Mr. Mopoji*. A total fake film, but we had a plot and we even gave it a kind of history, with production in Hong Kong. We put the 'soundtrack' out as a reissue and took it around to record stores. These stores would never have touched a funk or soul record by a new band, but when they saw a 'reissue' they scooped it up. We heard people saying, 'Oh, yeah, my cousin had that movie on VHS.'

"Next, we recorded [*instrumental band*] the Daktaris, which was a pretty rough recording. So we said it was recorded in Nigeria and people assumed it was an authentic African group. We never told anyone anything else about it; they just created their own assumptions. We had an ethnomusicologist in LA tell us that he had other Daktari records! It's kind of disconcerting, seeing how much bias people look at things through."

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The main outboard rack at Daptone Studios houses some desirable modern gear such as the Tube-Tech compressor and EQ, two Purple Audio MC77 compressors, an Anthony Demaria Labs tube compressor and an API Lunchbox, along with some quirky vintage units such as Orban spring reverbs and an Altec compressor.

have the most energy and they [*saturate*] the tape before the rest of the sound has a chance to get the benefit of the nice tape distortion. This lets the distortion take place closer to the upper range of the sound and keeps it tight. I'll put the low end back in during the mix if it's necessary."

Panning & Singing

A Daptone record encourages one to play with the left-right channel control on a car radio. Roth's panning is radical and static — when it's not simply mono. Much of this is due to the need to bounce tracks at times, but he says it's also stylistic. "Lately we haven't been doing what you might call 'subtle' panning," he concedes. "But you'll hear the same thing on Motown records. It's cool to have everything so right there."

On Sharon Jones & The Dap-Kings' *100 Days, 100 Nights* album, the panning is Beatlesque: all the drums hard left and the bass hard right. "It's kind of like when you're in a restaurant and they have music playing through speakers in the ceiling but where you're sitting you can only hear one side of the stereo," as Roth describes it. But there's a method to this particular madness. "It gives a lot of space down the middle for vocals," he adds.

Good time to discuss vocals, then, and Jones' record typifies the adventurous spirit of the studio. "I'll try anything on vocals — the RCA, the [*Sennheiser*] 421, a Shure 58," Roth says. As often as not, the pilot vocal from the tracking session winds up as the keeper, and on some tracks Jones did her vocal while sitting at the upright piano. She wasn't as 'on' the microphone as usual, but the performance overshadowed that.

When capturing the most inspired performance is the goal, it figures that there will be some clean-up work after the fact. On a track that Jones sang through a Rode NT1, Roth noticed sibilance problems. "I bought a cheap de-esser and rolled off some of the high end, and that mellowed it out," he says. In another instance, the snare drum leaked significantly into the vocal track, particularly noticeable in the 200Hz to 400Hz range. Roth rolled those frequencies back, but that thinned out the lead vocal. Roth's habit of always doing instrumental mixes of songs resolved the matter. "By bouncing the original instrumental mixes from the quarter-inch two-track to the one-inch eight-track and then flying the vocals from the one-inch 16-track over them on the eight-track, I was able to remix [*and re-EQ*] just the vocals without having to remix the instruments," he explains.

While he leans towards spring reverbs on vocals and much else, Roth likes tape slap as well, requiring the kind of calculations not often seen in the digital age. "We have two inches between the record and the repro head and at 15ips it's one second divided by seven, so you have a delay of about 140 milliseconds," he reckons.

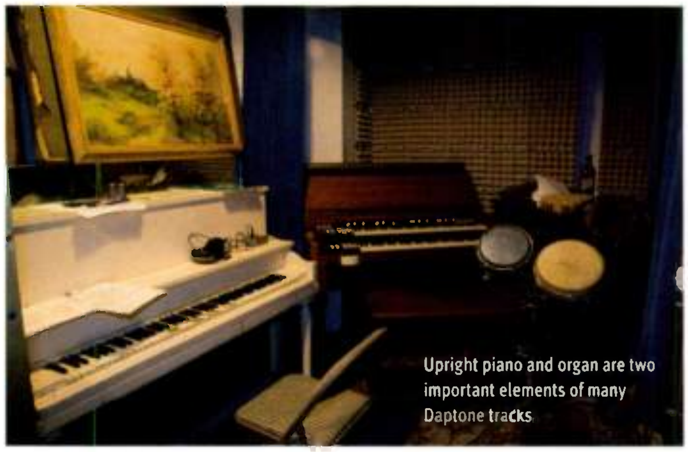
The Mix — Not Much To Fix

Roth's mixes suggest what it was like to fly aeroplanes before automatic pilots were invented. "The needles don't really tell you anything useful," he says earnestly. "I put all the machines into repro while tracking and listen off the playback head, which is the most accurate way to know what you're really listening to. Tape distortion is something that can be heard but not accurately seen by a VU meter, because different transients and frequencies saturate the tape and affect the needles differently. For example, a bass guitar can pin the needle for an entire song and sound fine. On the contrary, sometimes a tambourine will be hot and crunchy and barely move the needle at all. You have to be careful of trusting anything but your ears. I like to think I listen with the ears of a fan of the music. I'm not trying to inflict or avoid distortion — I'm listening for what makes the music sound good."

What comes out of Daptone is fun. And serious. It's not some Disney-like attempt at recreating what it must have been like at Muscle Shoals Sound or American Studios in Memphis in 1965. For some music engineers, the gold standard is Steely Dan; for Roth and his compatriots at Daptone, it's Irma Thomas records. Roth and the Daptones complete three to five basic tracks per day, half that if they're also doing vocals. He is not being disingenuous when he says "I'm not sure if that's a long time or not."

"It's funny," he continues, "People work so hard to get their drums to sound like they were recorded with one microphone. We just put one microphone out there. I'd rather spend two weeks looking for the right place to put the one microphone than on setting up two dozen mics and trying to balance them. We like working on eight-track and 16-track decks — it forces you to commit to decisions about sounds and arrangements on the spot."

Roth can go on and on, making specific technical references one minute and providing the philosophical rationalisation for them the next. I believe him when he says that the Daptones are not wilfully making retro records for the sake of it. Daptone are not some super-cool karaoke cover factory churning out the soundtracks to '70s blaxplotation flicks, or the Red Sauce web site churning out note-for-note recreations of classic tracks, but a place where people who really love a certain kind of music use the tools of the time to continue to make new editions of that music. The songs are new, the artists are new; the circumstances are similar to those who made this kind of music before them. They are not Civil War re-enactors going out on the weekend to show the families sitting on the side of the hill what it was like to watch the battle of Gettysburg go down; instead, the Daptones are the Amish, getting up every morning (but not too early) to go forth and do what their spiritual predecessors did, with the same kinds of tools and the same kind of passion.



Upright piano and organ are two important elements of many Daptone tracks

It's not about recreating history. It's not about homage. It's about pulling the lessons out of pieces of black vinyl and figuring out how your ancestors did it. And the evidence at Daptone suggests that the ancestors would definitely approve. SOS

Kick & Snare

Advanced Recording Techniques

Mike Senior

We pass on the hard-won wisdom of fifty top producers in the essential *Sound On Sound* guide to recording kick and snare drums, the backbone of modern music.

When it comes to describing in interview how they record drums, top producers seem to spend more time discussing techniques for snare drum and bass drum than anything else. The reason I know this is that I've recently been combing *SOS's* interview archive and Howard Massey's book of interviews *Behind The Glass*, looking for nuggets of wisdom on this very subject. In the process, I've compared the snare- and kick-recording tricks of fifty high-profile producers, who are collectively responsible for hundreds of millions of record sales. My aim in this article is to distil that information for you, to help you on the way to your perfect drum sound.

Clearly, different miking methods suit different records, and even the studio greats disagree about which techniques 'sound best', so ultimately it's up to you to decide which works for you. However, drum-miking is long on options, whereas most studio musicians are short on session time and don't have the luxury of comparing lots of setups while the band are breathing down their neck! So I've created audio files to demonstrate dozens of the most commonly used mics and placements for both snare and kick, using the same drum kit, the same drummer, and the same room. This means that you can audition and evaluate them all side by side at your leisure, and decide on the most useful contenders before your next session.

I'll be dropping a lot of producers' names in this article, some of which you may not initially be familiar with, even though you've probably heard a lot of their records! To avoid an avalanche of parentheses, I've listed all of

them in separate boxes throughout this article, together with a handful of their most relevant credits for reference.

Fix It In The Live Room

As with recording any instrument, the choice of drum and the manner of its tuning and preparation can make a huge difference to the sound you capture, so this should always be the place to start. Even if you don't play drums yourself, it makes sense to discuss the sound with the drummer, encouraging him or her to make any adjustments that might improve the sound. When discussing his recordings for The Darkness, Roy Thomas Baker also stresses how the type of music dictates the target sound: "You're going for a sound that's appropriate for the song, not necessarily what's a good sound, and what's appropriate can vary greatly... That's why we had three drum kits and a multitude of different snare drums and tom-toms."

Nile Rodgers takes a similar view, adjusting the kit to suit the song: "Even if the band uses one drum kit for the whole record, I want it tuned right for each song. We'll change the heads or tune it differently, all that kind of stuff. Sometimes we change the beaters... It all depends on how those frequencies are responding to the key of the music, to the pulse of the music. Every record is different, every song is different, every tape is different."

That said, Alan Winstanley found that sticking with a particular model of snare drum that suited his sound constituted a useful production shortcut: "I remember renting a Ludwig Black Beauty snare drum from a record company for [*Madness' The Rise And Fall*], and [*their drummer*] actually ended up buying one because he liked the sound of it so much. At the same time, it became a staple part of... my drum sound for quite a few years — 'If you haven't got a Ludwig Black Beauty, we're going to rent one in'."

Snare Miking: Over, Under, Or Both?

Important as it is to get the sound right at source, the main focus of this article is on how to translate that sound into recorded form. In this context, the predominant studio tool for manipulating snare and kick sounds while recording is the close mic. In the case of the snare drum, although the drum overheads will usually include a great deal of the snare sound already, a carefully selected and placed additional close mic has the potential to significantly remould the sound. For this reason, most engineers have highly developed personal preferences here.

Miking the snare drum's batter head is an almost universal first choice, either with one mic or two different mics together, but opinions are divided as to whether the drum should be also be miked from below. John Astley, Joe Barresi, Dave Eringa, Chris Thomas and Alan Winstanley are amongst those who regularly mic both sides of the drum, but Steve Churchyard is more circumspect ("I'll put something underneath, but invariably I won't use it"), Alan Parsons is less of a fan ("I try and steer away from using two mics on the snare. I prefer a good over-the-top snare sound"), and John Leckie avoids it most of the time (I very rarely use an underneath mic — it's very dangerous"). By way of contrast, though, Bruce Botnick regularly used the under-snare position on its own, combining this with just overhead and kick mics to provide a complete drum sound.

Even top and bottom snare mics together aren't enough for some people, though. When I interviewed the Feeling recently, they had also set up a large-diaphragm condenser at the side of the snare drum. They explained

To give some idea of the kinds of tonal change that fine positioning can bring about when using a snare-drum close mic, we set up six different Shure SM57s as shown here, generating the 'SnareTopPos' audio files.

why they used it: "Whenever you get really close to a drum head you get all sorts of strange frequencies you don't hear from a distance, so it sounds really odd. The side mic gets a much more overall view, and captures the top and bottom sounds together, but without the real upfront sound... It's been much more useful having that side mic than even a bottom mic a lot of the time... You get the 'crack' from the top mic, fizziness from the bottom mic, and then this side mic seems to find that 'beef' area."

Dealing With Phase

When Tony Visconti uses an under-snare mic he gates it to keep it carefully under control: "If I'm not getting brightness, if it's a very dull snare, or something's wrong that day, or it's just not cracking, then I will put an under-mic in, but I will always gate it — I don't like it rattling around all the time... I want to clean that bugger up before it gets to tape." He also touches on the importance of the phase relationship between the two snare close-mics: "Invariably [*the under-snare mic*] is out of phase with the top one. I have never ever had the good luck of having both mics in phase naturally, so if you do that trick you must check it. Even though it's being gated, you'll hear a big difference if you just play with the phase button, and you'll find that the low end will disappear if it's out of phase."

When two different over-snare mics are used simultaneously, phase cancellation can also easily make a mess of the instrument's high frequencies if the mic capsules aren't carefully aligned, but phase is even an issue if ▶

Five classic snare mics were compared over the top of the drum for the 'SnareMics' audio files (left, clockwise from top left): a Neumann KM84, a Shure SM57, an AKG C451EB, a Neumann KM86 and an AKG C414B-ULS. Above, under the snare were a further SM57 and C451EB.

you opt for just a single snare mic, given its interaction with the overheads and any other close mics. "I mix a lot of other people's stuff," says Steve Churchyard, "and by far the most common problem with drum sounds — particularly when they're multi-miked — is phase. Typically the overheads are out of phase with everything else, because they're hearing things later than the closer mics. So typically I'll [*invert the polarity*] and usually all the sounds kind of come forward. When you've got things out of phase, it sounds like the snare's kind of sucked into the kit and the bottom end is gone; it's a strange thing."

"Anything and everything should always be checked for phase," concurs Thom Panunzio. "I check the bass drum phase with the overheads, I check the snare with the overheads, I check the toms with the bass drum, I check the toms with the snare... All you've got to do is just hit the [*phase invert*] button on the console and see if it sounds better in or out of phase... It takes no time — if it took an hour, it would be well worth the trouble it saves you later."

The Perfect Mic Position?

Most engineers recognise that the exact position and angle of the snare mic are very important considerations. Ian Grimble: "Another crucial factor is mic positioning... You have to get the sound right at source and use the right mic in the right position. This results in a much cleaner sound. So I tend to fiddle around with mics a lot, and often spend more time [*in the live room*] trying out mic positions than in the control room." Jon Kelly also recalls Geoff Emerick taking "immense care positioning the mics" when recording drums.

Despite this, however, all the producers I researched remained staunchly noncommittal as to the exact mic positions they use. If you're the kind of person who lines their hat with Baco Foil, then you might suspect an industry-wide conspiracy to closely guard trade secrets, but Craig Leon provides a more down-to-earth explanation: "How you mic something and how you EQ it — in fact, everything that you do — is actually driven by the way the instrument is being played... so there isn't a standard setup that works; no one thing works all the time." Steve Albini elaborates a little: "It's hard to describe where I place [*the mics*] and it varies a lot. If the drummer plays very lightly, then there's a lot of attack and not a lot of tone, and I want the microphone to look at the contact point of the snare drum. If the drummer is playing very hard and he's exciting the whole drum, I usually have to back the microphone off a little bit so that it's not overloading."

As defensible as these viewpoints might be, it's not much help for those short on drum-recording experience. After all, if you don't really know what differences you can achieve with changes in mic positioning, then following Albini's advice and trying to make adjustments in response to the drummer's playing style will be a bit of a stab in the dark. So to help out I've created a set of audio files to demonstrate the sounds of close mics in a variety of positions. By comparing the audio files, you can get a feel for the sonic options available and thereby speed up the process of finding a great sound in the heat of your next session. These audio files, like all the others in this article, are available to download from the *SOS* web site at www.soundonsound.com/sos/jun08/articles/kickandsnareaudio.htm:

- **SnareTopPos1inchMidway**
- **SnareTopPos3inchMidway**
- **SnareTopPos8inchMidway**

These recordings were made with Shure SM57s aimed at a point halfway between the centre and edge of the head, at distances above the batter head of one inch, three inches, and eight inches respectively. A very

Who's Who: Selected Discography

Here is a list of the producers mentioned in this article. I've put the source of the interview in parentheses after each name in case you fancy reading more about them, and have also appended a selection of the most influential recordings they've been involved with for reference.

Robbie Adams (*SOS* March 1994 & July 1997)
U2: *Achtung Baby, Zooropa*; Pattie Griffin: *Impossible Dream*; Smashing Pumpkins: *Adore*; Lara Fabian: *A Wonderful Life, One*; Donnie Osmond: *What I Meant To Say*.

Steve Albini (*SOS* September 2005)
The Pixies: *Surfer Rosa*; Nirvana: *In Utero*; Bush: *Razorblade Suitcase*; PJ Harvey: *Rid Of Me*; Jimmy Page & Robert Plant: *Walking Into Clarksdale*.

John Astley (*SOS* May 2005)
Eric Clapton: *Crossroads, Just One Night*; The Who: *Who Are You*.

Joe Barresi (*SOS* July 2005)
Queens Of The Stone Age: *Queens Of The Stone Age, Lullabies To Paralyze*; Tool: *10000 Days*; The Melvins: *Stoner Witch*; Hole: *Celebrity Skin*; Limp Bizkit: *Chocolate Starfish & The Hotdog Flavoured Water*; The Lost Prophets: *Start Something*; Skunk Anansie: *Stoosh*.

Bruce Botnick (*SOS* December 2003)
The Beach Boys: *Pet Sounds*; Love: *Forever Changes*; The Doors: *The Doors, Strange Days, Waiting For The Sun, The Soft Parade, LA Woman, Morrison Hotel*; Tim Buckley: *Happy Sad*.

Billy Bush (*SOS* June 2002)
Garbage: *Beautiful Garbage, Bleed Like Me, Version 2.0*; Pat Monahan: *Last Of Seven*; Against Me!: *New Wave*.

close placement will tend to emphasise certain frequencies in the drum unduly, whereas moving the mic further back gives a more natural sound, albeit with increased levels of spill from other instruments.

- **SnareTopPos3inchCentre**
- **SnareTopPos3inchMidway**
- **SnareTopPos3inchRim**

For these recordings I used Shure SM57s three inches above the batter head, aimed respectively at the centre of the drum head, halfway between the centre and edge, and at the edge.

- **SnareTopPosC414Shell**

I also set up an AKG C414B ULS large-diaphragm condenser to mimic the setup I'd seen the Feeling using — about six inches to the side of the drum shell.

- **SnareTopPosOH**
- **SnareTopPosKick**

While listening to the individual mic positions solo is interesting in its own right, it's difficult to really evaluate their effectiveness in practice without hearing them mixed in with the overhead mics (as they would normally be during mixdown). For this reason, I recorded a stereo overhead mic pair (Shure KSM141s in an X/Y configuration) alongside every set of audio examples, so that you can compare sounds in context, and there was also an AKG D112 mic set up just inside the hole in the kick-drum's resonant head for similar reasons. To mix and match all these files, though, you'll have to import the examples into your MIDI + Audio sequencer and line them up so that they all start at exactly the same time.

Snare Mics To The Stars

If you read my article on recording electric guitars back in *SOS* August 2007 (www.soundonsound.com/sos/aug07/articles/guitaramprecording.htm), you'll know that Shure's SM57 can lay a strong claim to being the king of guitar mics. However, this rugged little mic is also the most commonly-cited choice for snare drum: a list of users includes such audio luminaries as Steve Churchyard, Bob Clearmountain, Mike Hedges, John Leckie, Elliot Scheiner, Stephen Street, Bill Szymczyk, Chris Thomas and Tony Visconti, to name only a few — and at least half a dozen of them are happy to use SM57s for both the drum heads.

Given the spatial restrictions when placing snare close mics, the SM57's low-end 200Hz roll-off is doubtless part of its appeal, combating proximity-effect bass boost when it's used up close, as well as minimising kick-drum spill. The other response characteristics (a 300-500Hz dip and a generous 2-12kHz presence peak) serve to reduce muddiness and add more 'snap'. However, the mic's rapid slump in sensitivity above 12kHz means that some engineers using this mic always reach for EQ boost: Steve Churchyard mentions the 10kHz region, while John Leckie goes for around 8kHz and Jim Scott 5kHz. The SM57's fairly tight cardioid polar pattern, designed with on-stage feedback reduction in mind, also has to be a factor, helping to reduce spill from other drums, and particularly from the neighbouring hi-hat.

Other than the SM57, two other mics, both small-diaphragm condensers, also stand out as leaders of the pack: AKG's C451EB and Neumann's KM84. What has come to be known colloquially in the trade as 'the 451' was actually part of a modular system of mic preamp bodies and interchangeable capsules with different polar patterns. However, it's probably fairly safe to assume that it is the mic's CK1 cardioid capsule (the one originally shipped as standard with the preamp body) that is being referred to, given that no mention ever seems to be made of the

Who's Who: Selected Discography

Steve Churchyard (*SOS* September 2005, *Behind The Glass*)
The Pretenders: *Learning To Crawl*; Counting Crows: *Recovering The Satellites*; Celine Dion: *Falling Into You*; Ricky Martin: *Vuelve*, *Almas Del Silencio*, *Sound Loaded*; Shakira: *Laundry Service*; The Stranglers: *La Folie*, *Feline*; Big Country: *Wonderland*; Bryan Ferry: *Boys & Girls*; INXS: *Listen Like Thieves*.

Bob Clearmountain (*SOS* July 2006)
The Pretenders: *Get Close*; Bryan Adams: *Into The Fire*, *Reckless*, *Cuts Like A Knife*; Paul McCartney: *Tripping The Live Fantastic*; mixing for Bruce Springsteen (*Born In The USA*), The Rolling Stones (*Tattoo You*), Bon Jovi (*These Days*, *Crush*), Roxy Music (*Avalon*), David Bowie (*Let's Dance*), INXS (*Kick*), and The Corrs (*Talk On Corners*).

Jack Douglas (*Behind The Glass*)
Aerosmith: *Get Your Wings*, *Toys In The Attic*, *Rocks*, *Draw The Line*, *Rock In A Hard Place*, *Honkin' On Bobo*; John Lennon: *Double Fantasy*; Cheap Trick: *Cheap Trick*, *At The Budokan*, *Standing On The Edge*.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type of capsule used. Indeed, AKG's modern reissue of this mic, the C451B, only offers a CK1-style cardioid polar pattern. Tonally, the C451EB is a bright-sounding mic, with a good 4dB frequency boost in the 10-15kHz area as well as a very gentle low-end roll-off from 200Hz, and it's no surprise to find Joe Barresi, Bob Clearmountain, Dave Eringa, Butch Vig, and Toby Wright all using it alongside an SM57, thereby compensating for the latter mic's HF roll-off. Three of them actually strap it to the SM57 with gaffer tape to make the setup easier to manage. Other producers (such as John Leckie, Ian Little, and Al Schmitt) are happy to use this mic on its own as well — although John Leckie usually uses an SM57, he adds that "sometimes I substitute the AKG C451 with the pad in if the drummer is playing with brushes or is playing lightly; then it's better to use a condenser mic... it gives you more of a 'sizzle'."

Neumann's cardioid KM84 gives a much more neutral tonality, with a frequency response that's essentially flat between 100Hz and 15kHz, and only 2dB down at 50Hz and 20kHz, so it's eminently capable of capturing pretty much any instrument all on its own. Alan Parsons used it on Pink Floyd's *Dark Side Of The Moon*, for example ("I could never get a sound I was happy with using any mic other than a KM84."), directly influencing Peter Henderson to do the same for Supertramp's *Breakfast In America* sessions. John Fry chose one for his work with Big Star, and John Astley put KM84s above and below the snare when in the studio with Glyn Johns and the Who — the extended frequency response of a condenser makes it well suited to capturing the complex under-snare signal. The KM84's neutrality and extended bandwidth also provide a good contrast to the SM57's more characterful sound, and combining the two mics is a technique for which Jim Scott and Ian Grimble have both expressed a partiality.

There are, of course, many other mics that are the personal preferences of individual producers, but none of them are as commonly singled out as the SM57, C451EB and KM84. Nevertheless, a few others are worthy of mention given that different producers independently agree on their merits. For example, Joe Barresi shares his preferences for Neumann's dual-small-diaphragm KM86 and AKG's dual-large-diaphragm C414B ULS with Chris Thomas and Gil Norton respectively. Barresi also joins Toby Wright in selecting Sennheiser's wide-bandwidth, supercardioid dynamic MD441 as a suitable under-snare contender. Another shared preference arises between Bruce Botnick and Steve Albini, who have both made use of Sony's 1950s-vintage C37 single-diaphragm condenser (a precursor to the current, breathtakingly pricey C800G).

To illustrate the sonic differences between some of these mics (in terms of both timbre and spill rejection), and to show the impact on the sound of using different drums, I lined up five over-snare mics (Shure SM57, AKG C414B ULS, AKG C451EB, Neumann KM84 and Neumann KM86) and two under-snare mics (Shure SM57 and AKG C451EB), and recorded them with the stereo overheads and kick close-mic to create three sets of audio files, one set for each of the following drums: an Orange County 14x5-inch maple snare (for the 'Snare1Mics' set of files); a Ludwig Black Beauty 14x5-inch hammered-brass snare ('Snare2Mics'), as singled out by Alan Winstanley; and a deeper Gretsch 14x6.5-inch mahogany snare ('Snare3Mics'). I was careful to try to phase-match the positions of the SM57, C451EB and KM84, so you should be able to experiment with typical combinations of these mic signals without encountering significant cancellation problems.

Classic Kick Mics

When it comes to choosing which mics to use for kick drum, all sorts of different models crop up in interviews, but some mics are cited

The three different drums used for our 'SnareMics' audio files (left to right): a Ludwig Black Beauty 14x5-inch hammered-brass snare, an Orange County 14x5-inch maple snare, and a Gretsch 14x6.5-inch mahogany snare.

Who's Who: Selected Discography

Geoff Emerick (*Behind The Glass*)
The Beatles: *Revolver, Sgt. Peppers Lonely Hearts Clud Band, Magical Mystery Tour, Abbey Road*; Paul McCartney: *Band On The Run, Flaming Pie*; Elvis Costello: *Imperial Bedroom, All This Useless Beauty*; Badfinger: *No Dice*; Robin Trower: *Bridge Of Sighs*.

Dave Eringa (*SOS* April 1999)
Manic Street Preachers: *Everything Must Go, This Is My Truth Tell Me Yours, Send Away The Tigers*; Idlewild: *100 Broken Windows, The Remote Part*.

Chris Fogel (*SOS* March 1997)
Alanis Morissette: *Jagged Little Pill, Supposed Former Infatuation Junkie; Under Rug Swept*.

John Fry (*SOS* April 2006)
Numerous Stax label releases by Isaac Hayes, The Staple Singers, and Booker T & The MGs; Big Star: *#1 Record, Radio City*.

Ian Grimble (*SOS* June 1998)
Manic Street Preachers: *Everything Must Go, This Is My Truth Tell Me Yours*; Texas: *White On Blonde*; The Beautiful South: *Welcome To The Beautiful South*; Siouxsie & The Banshees: *Through The Looking Glass*.

Mike Hedges (*SOS* June 1998, *Behind The Glass*)
Manic Street Preachers: *Everything Must Go, This Is My Truth Tell Me Yours*; Texas: *White On Blonde*; Travis: *The Man Who*; The Beautiful South: *Welcome To The Beautiful South*; The Cure: *Three Imaginary Boys, Seventeen Seconds, Faith*.

Peter Henderson (*SOS* July 2005)
Supertramp: *Even In The Quietest Moments, Breakfast In America, Famous Last Words*; Frank Zappa: *Sheik Yerbouti*; Paul McCartney: *Tripping The Live Fantastic, Flowers In The Dirt*; Jeff Beck: *Wired*; Rush: *Grace Under Pressure*.

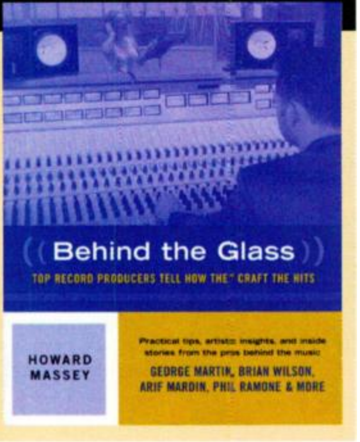
Tore Johansson (*SOS* March 1999 & June 2004)
Franz Ferdinand: *Franz Ferdinand*; The Cardigans: *First Band On The Moon, Gran Turismo, Long Gone Before Daylight, Super Extra Gravity*.

Jon Kelly (*SOS* June 2004)
Kate Bush: *Lionheart, Never For Ever, The Whole Story*; Chris Rea: *Road To Hell, Auberge*; The Damned: *Phantasmagoria*; The Beautiful South: *Gaze, Miaow, Painting It Red*; Heather Nova: *Siren*; Paul McCartney: *Press To Play*.

Howard Massey: *Behind The Glass*

This great book of interviews is, in my opinion, one of only a handful of truly essential record-production books, and is packed with down-to-earth recording advice as well as discussions of the art of production. In addition to the interviews I've referred to in this article, the book also features such greats as Glen Ballard, Arif Mardin, Brian Wilson, Phil Ramone, Mitchell Froom and George Martin, and one of the strengths of Massey's approach is that he often asks them similar questions, which makes for interesting comparisons. There are also two interesting panel discussions where several of the featured producers discuss their trade head to head.

£ *Behind The Glass* by Howard Massey (ISBN 0879306149), £16.95 including VAT.



again and again. Probably the most commonly mentioned are AKG's D12 large-diaphragm dynamic mic and its more recent D112 successor, used by producers such as Steve Albini, Roy Thomas Baker, Steve Churchyard, Bob Clearmountain, Elliott Scheiner, Steven Street, Chris Thomas, Butch Vig... the list goes on and on. A tight cardioid directional characteristic and the ability to cope with just about any SPL feature high on the list of reasons why they've become so popular, but it also has to do with the way their tonality is specifically tailored to flatter recordings of bass instruments. In particular, a special resonant chamber incorporated in both designs introduces considerable added weight at around 100Hz, while the D112 also has a pronounced presence boost at 3kHz that really emphasises beater definition, perhaps one reason why Alan Parsons has referred to it as "the punchiest drum mic I've found". A significant minority, including Tony Platt, Geoff Emerick and Chris Kimsey, turn instead to AKG's cosmetically similar D20 and D25, which use similar large-diaphragm capsules voiced with a flatter bass response. Both models are technically identical, but the D25 is built into a rather groovy-looking elasticated shockmount.

Another large-diaphragm dynamic mic that is a favourite of Joe Barresi, John Leckie, Peter Henderson and Toby Wright is Sennheiser's MD421. On account of its 100Hz low-frequency roll-off and hefty 5-8kHz sensitivity peak, it is in practice often used in combination with another, bassier mic (such as the D12/D112), although Henderson has used it on its own too. Shure's ubiquitous SM57 is name-checked for kick-drum recording by Ian Little and Tore Johansson, but given its even higher 200Hz bass roll-off, Little's tactic of combining it with AKG's C414B ULS, notable for its extended LF response, comes as no surprise. Both Steve Churchyard and John Leckie mention, though, that such combination techniques can easily cause disastrous phase cancellation, just as with snare multi-miking, so be prepared to finesse mic positions and experiment with phase-inversion for the most solid sound.

This picture shows the six classic kick-drum mics we compared in the 'KickMics' audio examples. In the middle is the Electrovoice RE20, above it is the Neumann U47 FET, and below it the AKG D12. To the right is Sennheiser's MD421, while a Shure SM57 and an AKG D112 are to the left.

Electrovoice's RE20 is another popular choice, with a frequency response that's unusually flat for a dynamic mic, and good resistance to proximity effect. Glen Kolotkin, Jim Scott and Chris Tsangarides have all been happy to use this mic on its own on their recordings, but Don Smith and Tony Visconti have both combined it with the more tonally sculpted D12 — in Visconti's case using the RE20 to focus on the beater definition while laying the D12 on a pillow in the drum to catch the low end.

Although there are a number of popular preferences when it comes to dynamic mics for kick, in the field of condenser mics one model reigns supreme: the large-diaphragm Neumann U47 FET. Joe Barresi, Steve Churchyard, Eddie Kramer and Jon Kelly are just some of the producers using this mic, and ▶

Chris Kimsey (*SOS* April 2004)
The Rolling Stones: *Sticky Fingers, Some Girls, Emotional Rescue, Undercover, Steel Wheels*; Peter Frampton: *Frampton Comes Alive!*; The Chieftains: *Long Black Veil*; Killing Joke: *Laugh? I Nearly Bought One, For Beginners*; The Cult: *Dreamtime*; Marillion: *Real To Reel/Brief Encounter*.

Glenn Kolotkin (*SOS* June 2000)
Santana: *Supernatural, Caravanserai, Borboletta, Moonflower*; The Rolling Stones: *Their Satanic Majesties Request*; Jimi Hendrix: *Electric Ladyland*; Janis Joplin: *Pearl*.

Eddie Kramer (*SOS* November 2005, *Behind The Glass*)
Jimi Hendrix: *Are You Experienced?, Axis Bold As Love, Electric Ladyland, Band Of Gypsys, The Cry Of Love*; Led Zeppelin: *Led Zeppelin II*; Kiss: *Alive!, Alive II*; Peter Frampton: *Frampton Comes Alive!*.

John Leckie (*SOS* May 1997, *Behind The Glass*)
Pink Floyd: *Meddle*; Radiohead: *The Bends*; Muse: *Showbiz, Origin Of Symmetry*; The Stone Roses: *The Stone Roses*; The Verve: *A Storm In Heaven*.

Craig Leon (*Behind The Glass*)
The Ramones: *The Ramones, Ramones Mania*; Blondie: *Blondie, No Exit, The Curse Of Blondie*; The Fall: *Extricate, Shift-work*; Rodney Crowell: *But What Will The Neighbours Think*.

Ian Little (*SOS* July 2004)
Roxy Music: *Avalon*; Duran Duran: *First, Seven & The Ragged Tiger*.

Steve Marcantonio (*SOS* October 2002)
Rascal Flatts: *Still Feels Good*; Faith Hill: *Fireflies*; Keith Urban: *Keith Urban*; Brooks & Dunn: *Hillbilly Deluxe*; Kenny Chesney: *The Road & The Radio*; Tim McGraw: *Not A Moment Too Soon*; George Strait: *Carrying Your Love With Me, Lead On*.

Ken Nelson (*SOS* October 2000)
Badly Drawn Boy: *The Hour Of Bewilderbeast*; Coldplay: *Parachutes, Rush Of Blood To The Head*, ▶

Here you can see the recording setup for the 'KickHeadOffMid' audio files, demonstrating the sounds of different mics and mic positions inside the kick drum. The five AKG D112s were at equal distances from the batter head, but in different lateral positions. (This same multi-D112 rig was used for all the other 'KickHeadOn' and 'KickHeadOff' files as well.) Above them, at the same distance from the batter head, were (left to right) an AKG D12, a Sennheiser MD421, and an AKG C414B-ULS.

many more mention the U47 without specifying the FET or valve models — although given that none of the interviewees mentioned the U47 valve specifically, I'd suggest that the FET design is fairly universally preferred. It's not that other mics aren't mentioned for use on kick drum (the AKG C414B ULS, Blue Mouse and Sennheiser MKH20 all popped up in my research), it's just that the U47 turns up 20 times as often!

So what's so special about this mic? Well it probably goes without saying that the low-end response remains substantially flat down beyond 30Hz, but this is also combined with a little presence boost in the 2-5kHz range and a larger sensitivity peak up at 10kHz for clarity. The mic will handle a maximum SPL of 147dB for 0.5 percent distortion, another good reason to go for it in this application over the valve version, which manages just 120dB. However, another important, and often overlooked, factor of its design is the well-controlled supercardioid polar pattern, a pattern that typically picks up less room ambience than the more common cardioid, helping to keep the kick sound clear and focused even when miking from outside the drum.

Enough of talking about all these mics: how do they actually sound? To give you an idea, I lined up six of them (AKG D12, AKG D112, Electrovoice RE20, Shure SM57, Sennheiser MD421 and Neumann U47 FET) right next to each other, 18 inches outside the front of the kick drum, and recorded them (along with usual stereo overheads setup and a Shure SM57 snare close-mic) to create the 'KickMics' set of audio files. Check them out and decide for yourself whether they live up to the hype!

Mic Positioning Inside The Bass Drum

It probably goes without saying that the kick drum will benefit from the same care and attention as the snare when it comes to preparing it for recording — tuning and damping can often solve problems much more easily than mic technique. You can also try using different heads or beaters, although, on the evidence of my survey, producers seem less inclined than with snares to try out different drums. One factor that needs extra thought with the kick drum is what to do about the resonant head: leave it on, take it off, or use one with a mic-access hole in it? There appears to be no real consensus amongst the producer interviews as to which of these approaches yields the best results, despite the obvious sonic differences between them.

Taking the resonant head off completely or using a head with a mic-access hole means that you can easily mic up the batter head to get a prominent beater 'slap', while the kick drum's shell helps keep spill from other instruments to a minimum. The problem with miking inside the drum, though, is that the resonant modes within the shell cause the sound to vary much more for small changes in mic position than you might normally expect. "It's amazing how much the sound changes," remarks Steve Churchyard, "if you're completely on-axis to the kick-drum pedal beater rather than more off-axis and to the side of the kick drum." Taking the resonant head off completely makes these internal resonant modes a little less severe, but it also lets in more spill and gives a different character to the sound. "I used to take the front head off, with a cushion inside" says John Leckie, "but now I prefer to leave the front head on, but with a hole cut in it; it sounds a bit more contained."

It may be because small placement tweaks make such a big difference that few producers give any exact indication of where they place mics inside the kick. However, there is one discernible trend in the way the sound changes that's worth bearing in mind: you'll get more beater presence when you're miking close to the batter head and more tone as you move the mic further away. You can also adjust the timbre and level of beater sound by

Who's Who: Selected Discography

X&Y; Paolo Nutini: ***These Streets***; King Of Convenience: ***Quiet Is The New Loud***; Gomez: ***Liquid Skin, Bring It On.***

Gil Norton (*SOS* December 2005)
The Pixies: *Trompe Le Monde, Bossanova, Doolittle*; Foo Fighters: *The Colour And The Shape*; Throwing Muses: *Throwing Muses*; Jimmy Eat World: *Futures*; James: 'Sit Down'; Echo & The Bunnymen: *Ocean Rain.*

Thom Panunzio (*Behind The Glass*)
U2: *Rattle & Hum*; Deep Purple: *The Battle Rages On*; Black Sabbath: *Reunion*; Ozzy Osbourne: *Live At Budokan*; Bruce Springsteen: *Tracks, 18 Tracks.*

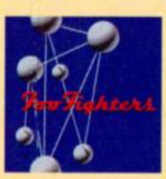
Alan Parsons (*Behind The Glass*)
The Beatles: *Abbey Road*; Pink Floyd: *Dark Side Of The Moon*; Al Stewart: *Year Of The Cat*; The Hollies: 'He Ain't Heavy, He's My Brother', 'The Air That I Breathe'.

Tony Platt (*SOS* April 2001)
Bob Marley: *Catch A Fire, Burnin'*; Toots & The Maytals: *Funky Kingston*; Aswad: *Aswad*; AC/DC: *Highway To Hell, Back In Black*; Foreigner: *4*; Boomtown Rats: *The Fine Art Of Surfacing*; Anathema: *Eternity.*

Bill Price (*SOS* September 2004)
The Sex Pistols: *Never Mind The Bollocks*; The Clash: *The Clash, Give 'Em Enough Rope, London Calling, Sandinista!*; The Pretenders: *Pretenders, Pretenders II*; Elton John: *Too Low For Zero*; Pete Townshend: *Empty Glass*; The Jesus & Mary Chain: *Darklands*; The Libertines: *The Libertines.*

Nile Rodgers (*Behind The Glass*)
Any record by Chic; Sister Sledge: *We Are Family*; Diana Ross: *Diana*; David Bowie: *Let's Dance, Black Tie White Noise*; Madonna: *Like A Virgin.*

Elliot Scheiner (*SOS* February 1996)
Steely Dan: *Aja, Gaucho, Two Against Nature*; Donald



adjusting the angle of your mic. These principles not only allow you to balance the different elements of the kick sound to taste with careful mic positioning, but can also be used to separate the 'beater' and 'tone' characteristics to different tracks for more control at mixdown, much as Tony Visconti does with his 'D12 plus RE20' technique (see above). In a different interview, Tony Visconti also suggests an alternative placement for the D12, right in the hole of the resonant head, and this is a setup that offers similar practical advantages.

In the absence of much exact placement information from the pros, I've recorded some audio examples to try to demonstrate the ways the sound varies as you change mic positions inside the drum, both with the resonant head removed and using a resonant head with a mic-access hole cut into it. To do this, I set up five D112s in a line next to each other, to cover five different lateral positions inside the drum: mic one was a little to the left of centre (from the audience's perspective), mic two was roughly in the centre facing the beater, and mics three to five worked their way out towards the shell. I recorded this rig at three distances from the batter head: as close as possible ('KickHeadOnClose' and 'KickHeadOffClose' sets); about eight inches back ('KickHeadOnMid' and 'KickHeadOffMid' sets); and as far back towards the position of the resonant head as I could ('KickHeadOnFar' and 'KickHeadOffFar'). Listening to these files not only illustrates how dramatic the tonal variation can be even for mics at the same distance from the batter head, but also shows how the sound changes in a general way as you move the mic forwards or backwards. The usual stereo overheads and snare-drum close-mic were recorded alongside.

Outside The Bass Drum

With an unperforated resonant head in place you tend to get the most resonant sound, and if you mic at the usual frontal position you won't get any real beater definition (because the mic can't see the kick pedal) and you'll pick up quite a bit of spill from the rest of the kit. While the resonance can be reduced if required, by virtue of extra damping inside or outside the drum, John Leckie points out that you can also deal with this simply by miking from a greater distance, about three feet away. In this position he sometimes uses an RCA 44DX ribbon microphone, a type of mic that is very rarely associated with kick-drum applications on account of the mechanical fragility of ribbon transducers.

To increase beater definition, you can simultaneously mic up both drum heads: ▶

Recording The Audio Examples

The audio files (which we've placed on the *SOS* web site at www.soundonsound.com/sos/jun08/articles/kickandsnareaudio.htm) were recorded over two days in the Colin Hill Recital Room at Hills Road Sixth Form College, Cambridge, making use of their Music Technology department's adjoining control room: thanks to the Director of Music, Jonathan Sanders, for making this all possible. Many thanks are also due to Ant Fox, our long-suffering drummer, and to Tom Adams, Matt Coulson and Geoff Smith, who all helped out tremendously with the session logistics. Likewise, I'm grateful to FX Rentals (+44 (0)20 8746 2121) for supplying many of the highly desirable mics we tested. If you fancy using any of them on your own sessions, you can rent them for just a single day if you like. For rates and discounts, see www.fxgroup.net.

Robbie Adams favoured a Sennheiser MD421 on the batter head when working on U2's *Zooropa*, while Steve Albini suggests a small dynamic or condenser mic, such as Shure's cardioid electret SM98. Billy Bush and Eddie Kramer also mention covering the mics and the front of the drum with blankets to cut down on the spill if necessary, a very common studio trick. So even if you can't get inside the drum, there's still a lot you can do to catch a tighter, punchier sound when you need it.

Rigging up my five D112s about four inches away from the drum (which had a full resonant head fitted), I recorded some more audio files ('KickHeadOnOutside'), with the usual overheads and snare close-mic, so that you can compare this sound to those produced by the various internal mic positions. In addition, I set up an MD421 and a KM84 over the top of the kick drum pointing down at the batter head's beater-contact point, along the lines suggested by Robbie Adams and Steve Albini.

External miking positions aren't just for when the inside of the drum is inaccessible, though; combining the signals from internal and external mics is actually a pretty common pro technique. The concept behind this is very similar to that of Tony Visconti's internal multi-miking techniques — the mics provide contrasting timbres, which can be balanced to get the required composite sound without heavy processing. "I use a three-microphone technique on bass drum," says Eddie Kramer of his variant on this approach: "A Shure SM52 and SM91 inside the bass drum and a U47 FET outside the bass drum. By playing games with the various qualities of each, I get the sound I'm after. I'm a great believer in the sound quality of each microphone, and you don't have to use a lot of radical EQ to get great sounds if you choose microphones for their particular qualities and put them in the right place."

The recording of the 'KickHeadOnOutside' audio files placed five AKG D112s at four inches from the kick drum's resonant head, and combined them with a Sennheiser MD421 and a Neumann KM84 miking the batter-head side. You can also see the positions of the two externally positioned U47 FETs recorded alongside all the 'KickHeadOn' and 'KickHeadOff' files, which were there to demonstrate the potential of a number of common dual-miking techniques.

Steve Churchyard and Steve Marcantonio independently describe almost identical techniques, pairing a D12 or D112 inside the drum with a U47 FET outside. Marcantonio makes it clear that he positions the internal mic right next to the beater skin, and the resultant hard, clicky sound balances well with the U47 which, remarks Churchyard, "adds a lot more fullness and roundness to the sound." Billy Bush went for a fairly similar technique in his work with Butch Vig's band Garbage, but selecting an Audio Technica ATM25 for the inside mic, and occasionally substituting a Blue Mouse for the U47.

Joe Barresi, on the other hand, prefers the MD421 or Shure Beta 52 inside, and although he lists the U47 as one option for the external mic, the alternatives he offers (a Beyerdynamic M160 ribbon mic or the old studio trick of using a Yamaha NS10 speaker cone as a mic) imply that he's looking to it more for a unique character than for general-purpose warmth. Another different

Who's Who: Selected Discography

Fagan: *Nightfly*; Billy Joel: *Songs In The Attic*; Fleetwood Mac: *The Dance*; Roy Orbison, *Black And White Night*; John Fogerty: *Premonition*; Van Morrison: *Moondance*.

Al Schmitt (*SOS* October 2005, *Behind The Glass*)
George Benson: *Breezin'*; Steely Dan: *Aja, FM (No Static At All)*; Toto: *Toto IV*; Natalie Cole: *Unforgettable*; Diana Krall: *When I Look In Your Eyes, The Look Of Love*; Ray Charles: *Genius Loves Company*; Jefferson Airplane: *After Bathing At Baxter's, Crown Of Creation, Volunteers*.

Jim Scott (*SOS* December 1999)
Red Hot Chili Peppers: *Californication, By The Way*; Dixie Chicks: *Taking The Long Way*; Sting: *Dream Of The Blue Turtles*; Johnny Cash: *American Recordings, Unearthed*; Foo Fighters: *One By One*; Lucinda Williams: *Car Wheels On A Gravel Road*.

Don Smith (*SOS* December 1994)
The Rolling Stones: *Voodoo Lounge*; Ry Cooder: *Chavez Ravine, My Name Is Buddy*; Stevie Nicks: *Rock A Little, Trouble In Shangri-La*; The Tragically Hip: *Up To Here, Road Apples*; Tom Petty: *Long After Dark, Southern Accents, Full Moon Fever, The Last DJ*; Roy Orbison: *Mystery Girl*.

Al Stone (*SOS* December 1999)
Jamiroquai: *Return Of The Space Cowboy, Travelling Without Moving, Synkronized*; Daniel Bedingfield: *Gotta Get Through This*; Stereo MCs: *Connected*;

Bjork: *Debut, Post*; Turin Brakes: *The Optimist*; Lamb: *Fear Of Fours*; Eagle Eye Cherry: *Sub Rosa*.

Stephen Street (*SOS* July 1994 & January 2005)
The Smiths: *Meat Is Murder, The Queen Is Dead, Strangeways Here We Come*; Morrissey: *Viva Hate, Bona Drag*; Blur: *Leisure, Modern Life Is Rubbish, Parklife, The Great Escape, Blur*; The Cranberries: *Everybody Else Is Doing It, So Why Can't We?, No Need To Argue, Wake Up & Smell The Coffee*; Kaiser Chiefs: *Employment, Yours Truly Angry Mob*.

Bill Szymczyk (*SOS* November 2004)
The Eagles: *On The Border, The Long Run, One Of These Nights, Hotel California*; The Who: *Face Dances*; BB King: *Live & Well, Completely Well*; J Geils Band: *The Morning After*; Joe Walsh: *Barnstorm, The Smoker You Drink The Player You Get*.

approach comes courtesy of Ian Little, who turns the typical 'dynamic inside, condenser outside' idea on its head, putting a C414B ULS inside and an SM57 outside. And there's no need to abandon this kind of dual-mic technique just because you can't get a mic inside the drum, either. Ian Grimble used a combination of a D12 close to the drum and a Sennheiser MKH20 small-diaphragm condenser further away when recording the Manic Street Preachers *This Is My Truth Tell Me Yours* with Mike Hedges.

How Far Is Far Enough?

A final question to answer is: how far from the drum should an outside mic be placed? Steve Churchyard and Steve Marcantonio are pretty much on the same page again with "just outside" and "four to 10 inches away from the hole" respectively, whereas Robbie Adams went out to about 18 inches for U2's *Zooropa.* Joe Barresi and Ian Grimble constitute the 'three feet away' camp, presumably roughly in line with Al Stone's unusual three-mic setup for Jamiroquai's *Travelling Without Moving*: "*Travelling Without Moving* was a three-mic job, not even close-miked — one mic a few feet in front of the bass drum, one over the drummer's shoulder covering the hi-hat and snare, and another to the side to cover the cymbals."

To give you the option to experiment with the sounds of these kinds of multi-mic techniques I took the opportunity to add in a few extra mics while recording the 'KickHeadOn' and 'KickHeadOff' sets of audio files. Alongside all these file sets, you'll also find files for two different U47 FETs, one placed 12 inches from the drum, and the other at three feet. This means that you can freely substitute different internal mic positions to see how they combine with either of the external mic positions. Inspired by the dual-mic approaches of Steve Churchyard, Joe Barresi and Ian Little, I also slung up additional D12, MD421 and C414B-ULS alongside the D112s while recording the 'KickHeadOffMid' files, to give an impression of the different flavours imparted by those specific mics when they're used inside the kick.

In the light of the range of opinions as regards the best miking distance, I've also created a separate set of audio files ('KickDistance') to illustrate what kinds of sounds are on offer here. Fitting a resonant head with a hole in it to the kick drum, I re-used the same five D112s in a line, roughly one in front of the other: mic one was inside the kick drum, about 12 inches from the batter head; mic two was directly in the hole in the resonant head; and mics three, four, and five were 10 inches, 18 inches, and three feet away from the resonant head's hole respectively. The usual overheads and snare close mic were recorded alongside.

One of the things you'll notice, of course, in these files is that there's quite a lot of spill from the other instruments in the kit at larger miking distances. This means that they aren't necessarily very representative of those techniques where the kick mics have been baffled with blankets to reduce spill, so I also re-recorded the same set of microphones for the 'KickDistanceBlankets' files, using a few spare mic stands to support a closed tunnel of blankets and foam suspended around all the mics and gaffered to the drum itself.

In addition to this 'blanket tunnel', I also set up a different tunnel technique which Butch Vig described when talking about recording Nirvana's influential album *Nevermind.* "In the case of Dave Grohl's kit, I used an AKG D12 and a FET 47 on the kit, ▶

The tone of the kick drum changes dramatically as you move away from the batter head, so we recorded five AKG D112s at different distances to demonstrate this, generating the 'KickDistance' file sets. The amount of spill from other instruments also increases with distance, and many producers baffle the mic stands and kick drum with blankets to reduce this, so we created the 'KickDistanceBlankets' files to show how much of a difference this can make. In these photos you can see the final baffled setup and how the mics were positioned inside.

and then we built a drum tunnel consisting of old drum shells attached to the bass drum and extended out about six feet. That way you can move the mic back three to four feet, and the U47 was a little farther away from where the front head would have been. By having the drum tunnel, you isolate the room, so you don't get all the cymbal bleed."

Finding four extra kick-drum shells is by no means the easiest thing in the world to do, but once I'd tracked them down I removed the blanket tunnel I'd used for the 'KickDistanceBlankets' files and slotted the shells in place instead, gaffering them together before covering them in all the blankets again for good measure. In the light of Butch Vig's stated mic choices, I then added in a D12 next to mic number five (three feet from the drum) and put up a U47 FET right at the end of the tunnel.

Was it all really worth the trouble? Well, the resonant qualities of the tubular structure certainly impart a unique character to the sound, and what particularly impressed me was the way in which a U47 six feet from the drum could be made to sound as punchy as a mic sitting right in the hole in the resonant head. But don't trust my ears — take a listen to the 'KickDistanceTunnel' files and decide for yourself!

For the 'KickDistanceTunnel' audio files, we set up a technique described by the producer Butch Vig, where the kick-drum mics are protected from spill using a six-foot-long tunnel constructed from kick-drum shells. The mics inside the tunnel are the same as those used for the other 'KickDistance' files, but are also joined by an AKG D12 three feet from the drum, and a Neumann U47 FET at the end of the tunnel. For the recordings, we also covered the whole assembly with blankets to damp the drum shells and further reduce spill.

The Role Of The Room

In addition to the various close-miking techniques I've looked at so far, room ambience is regularly used specifically to support the kick and snare sounds, giving them an extra impression of power and size. However, you'd be forgiven for wondering how it's possible to do this, given that ambient mic positions will pick up a mix of the whole kit, rather than just kick or snare.

Even if you've somehow managed to avoid Phil Collins' 'In The Air Tonight' you'll doubtless have heard one of the most common tactics professional engineers use, namely gating the ambient mic tracks and then keying the gate side-chains from the kick and/or snare mics. Before you write off this

Who's Who: Selected Discography

Chris Thomas (*SOS* September 2004 & September 2005)
Pink Floyd: *Dark Side Of The Moon*; Dave Gilmour: *On An Island*; Razorlight: *Razorlight*; U2: *How To Dismantle An Atomic Bomb*; Pulp: *Different Class, This Is Hardcore*; INXS: *Listen Like Thieves, Kick, X*; The Pretenders: *Pretenders, Pretenders II, Learning To Crawl*; The Sex Pistols: *Never Mind The Bollocks*; Roxy Music: *For Your Pleasure, Stranded, Siren*.

Chris Tsangarides (*SOS* July 2001)
Thin Lizzy: *Renegades, Thunder & Lightening*; Ozzy Osbourne: *Blizzard Of Oz*; Judas Priest: *Painkiller*; Black Sabbath: *The Eternal Idol*; Gary Moore: *Back On The Streets, The Power Of The Blues, Back To The Blues*.

Butch Vig (*SOS* March 1997 & June 2002)
Nirvana: *Nevermind*; Smashing Pumpkins: *Siamese Dream, Gish*; Garbage: *Garbage, Version 2.0, Beautifulgarbage, Bleed Like Me*; Sonic Youth: *Experimental Jet Set Trash & No Star, Dirty*.

Tony Visconti (*SOS* October 2003 & October 2004, *Behind The Glass*)
T-Rex: *Electric Warrior, The Slider*; David Bowie: *Diamond Dogs, Young Americans, Heroes, Low, Scary Monsters, Heathen, Reality*; Iggy Pop: *The Idiot*; The Moody Blues: *The Other Side of Life, Sur La Mer*; Thin Lizzy: *Bad Reputation, Live And Dangerous, Black Rose*.

Alan Winstanley (*SOS* July 1998)
Madness: *One Step Beyond, Absolutely, Seven, The Rise & Fall, Keep Moving, Mad Not Mad*; Elvis Costello: *Punch The Clock, Goodbye Cruel World*; Bush: *Sixteen Stone*; Hothouse Flowers: *People, Home*; Morrissey: *Kill Uncle, Bona Drag*.

Toby Wright (*SOS* December 2005)
Slayer: *Divine Intervention*; Alice In Chains: *Jar Of Flies, Alice In Chains, Unplugged*; Korn: *Follow The Leader*; Metallica: *And Justice For All*; Motley Crue: *Girls Girls Girls*.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Ambience mics are often used by producers to bolster the sound of both snare and kick close-mics, so we recorded some extra ambience mics for the 'Amb' files, to let you experiment with the possibilities they afford. A pair of Earthworks TC20 omni mics are suspended high above the kit on the tall stand; a Sennheiser MKH416 rifle mic is also above the kit, angled to emphasise snare ambience; and a pair of Crown PZM mics are on the wall behind the kit. The PA used for the 'AmbPA' files was mounted on the wall opposite the PZMs.

technique as something only fit for '80s throwbacks, though, bear in mind that it has been used more subtly on a much wider range of records than you might initially imagine. You can hear something of this nature on the snare in AC/DCs 'Back In Black', for example, and Bill Price has divulged that keyed ambience was used all over the Sex Pistols' *Never Mind The Bollocks*.

"Various strategic ambience mics that suited the task were placed around the room," Price recalls. "A couple of old BBC ribbons were literally at floor level behind the drums to try to pick up the ambience of the bass drum and the bottom skins of the tom-toms, and a couple of Neumann KM84s were slung above the kit to pick up the ambience of the cymbals... [*The drum sound*] involved me keying different ambience mics off the drums as they were being hit, using the old-fashioned Kepexes. These were the earliest American gates available, and using them was pretty much an integral part of the sound. [*The producer, Chris Thomas's*] suggestion that we could shorten the ambience with gates, providing more without it sounding too distant, all made total sense to me."

Which close-mic channels you feed to the gate's side-chain and how the gating parameters are set up both make a big difference to the final sound. So rather than trying to demonstrate some 'ready-made' gated-ambience drum sound (for which you might as well just put on one of the records listed above) I've just recorded a basic kit setup alongside a few different ambience mics to create the 'Amb' set of audio files. This allows you to import the files into your MIDI + Audio sequencer and try out different gating strategies for yourself. The kit was miked with the usual stereo overheads: an SM57 and a KM84 over the snare; an SM57 under the snare; a D112 inside the kick drum, poked through the hole in the resonant head; and a U47 FET placed 18 inches outside the hole in the resonant head. Based on Bill Price's mic positions (and a very similar setup used by Ian Little), I miked in stereo both from above and behind the kit. A spaced pair of Earthworks TC20 mics occupied the high positions, their omni polar patterns emphasising the room ambience, and a pair of Crown PZM boundary mics were placed about a meter up the wall behind the drummer. Boundary mics (usually referred to as PZMs or Pressure Zone Microphones, although both terms are trademarks of one manufacturer), are a common choice for ambience miking, counting Joe Barresi, Dave Eringa, Ken Nelson, Ian Little and Tony Visconti amongst their fans.

Adjusting The Ambience Mix

An alternative strategy for supporting the kick and snare sounds, in particular, using ambience is to adjust the mix picked up by the ambient mics. Alan Winstanley had a trick for doing this by using a highly directional rifle mic, "positioned high and pointing down on the kit as an extra room mic, trying to pinpoint the snare more than the other stuff." I rigged up a Sennheiser MKH416 rifle mic in this way, alongside all the other mics, while recording the 'Amb' files, so you can hear what it sounds like. On a practical note, though, I found it took a little while to find a position for this mic where the snare level rose above the cymbal spill, so if you do decide to give it a go, make sure to work with the placement for the best results.

A much more common technique for increasing the snare and/or kick levels in the ambient mics, however, is to set up a PA system in the room and feed it with a mix of the close mic signals. Chris Kimsey adopted this approach for the snare on the Rolling Stones' 'Start Me Up', for example ("The PA was aiming at the drums, so the snare would actually come back through the overhead mic and create this quite unique sound."), while Steve Churchyard used tinny little Yamaha powered speakers behind the drummer when recording the Pretenders. He also sent the odd delay effect through there too.

Chris Fogel has applied a similar approach with kick drum. "I run everything that's at the bottom end, like kick and low tom, into a powered 18-inch subwoofer situated right behind the drummer... Basically it extends the bottom end on the kick drum, and when the drummer hits the kick, you really feel it... In fact, in some cases I didn't have to EQ the drums at all." Jack Douglas takes this idea and turns it up to 11: "Another trick I'm trying these days is to spread out a bunch of powered subwoofers — maybe six of them — in the room. Then I'll put a contact mic on the bass drum and use that as a trigger, so whenever the drummer hits his bass drum, everything from about 80 cycles down to 20 suddenly shakes the room. The subs fill the room with this low wave, which everything picks up — room mics, guitar mics, everything. That'd be pretty tough to do in a project studio, though..."

For my final set of audio files, 'AmbPA', I fed a mix of the kick and snare close-mics in the 'Amb' setup to a small PA system which happened already to be installed in the hall where we were recording. Although the PA we had was not the most powerful of systems, the way the extra snare and kick ambience changes on all the mics between the 'Amb' and 'AmbPA' sets demonstrates the potential of this idea.

101 Ways To Skin A Drum

It should be pretty obvious by now that there's no standard way to mike up a snare or bass drum, with different engineers gravitating towards certain techniques based not only on how they sound, but on how much control they provide at the mixdown stage. If you've listened through to all the audio files which accompany this article (or, better still, if you've imported them into your sequencer and A/B'ed them in context), you should be much better equipped to choose the right method to deliver the goods, whatever sound you're after. SOS

About The Author

Ex-*SOS* Reviews Editor and regular contributor Mike Senior has worked professionally with artists such as The Charlatans, Nigel Kennedy, Therapy, and Wet Wet Wet. He now runs Cambridge Music Technology, delivering courses and training based on the studio techniques of the world's most famous producers.
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FXpansion BFD2

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BFD is back, and with over 50GB of samples the 'B' is even bigger than before. So if you want the best in acoustic drum samples, should you be making space on your hard drive for BFD2?

The main Kit page of BFD2 with a 10-piece kit loaded.

John Walden

BFD caused quite a stir when it was first released, and reviews were extremely positive (including Paul White's, which appeared in the February 2005 issue of *SOS*). With a 9GB core library, plus the additional 22GB in the XFL expansion pack, it certainly qualified as 'big' and it was also definitely 'drums' — FXpansion let users make up their own minds about the 'F', with 'friendly', 'fabulous' or something more expletive being the obvious candidates. BFD wasn't just a sample library, however; the dedicated front-end provided a playback engine that could be used within all the major DAW hosts and it also included a library of over 1000 grooves — essentially MIDI drum patterns — that could be triggered from within BFD or dragged and dropped into the host sequencer for arranging and editing. Mixing facilities for the various close and room mics were also provided, allowing the user control over the degree of room ambience.

In the new release BFD2 boasts a 55GB core library, covering 10 full acoustic kits, multiple mic positions, performance articulations and up to 96 velocity layers. The user interface has been completely redesigned and, as well as comprehensive mixing capabilities, the new version also includes over 5000 drum grooves and a Cubase-like drum-pattern editor. As before, BFD2 works as a VST, RTAS and AU plug-in, and as a stand-alone application.

The world of sampled and virtual drumming has not been short of new products since the original BFD was launched (see the 'Drummer Auditions' box for some obvious competitors), so does BFD still make the grade?

Ten Drum Kits And A Studio

While the overhaul of the user interface is the first obvious change for BFD2, the sample data is also entirely new. The PDF manual gives the full details but all the BFD2 sounds were recorded in Studio 1 at AIR Studios in London, using a combination of vintage and high-quality modern equipment. The FXpansion web site has an excellent short video on the making of BFD2 (interviewing BFD2's producer Gareth Green), and that's well worth a look.

A full multi-mic configuration was used for each element of the kit, including mics both inside and outside the kicks, snare top and bottom, overheads, close room mics (about eight feet away) and ambient mics (23 feet away and therefore catching more of the room sound). The user gains access to all these mic combinations via the BFD2 mixer, which allows extremely detailed control over the drum sound — either very

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pros

- Wonderfully detailed samples that sound fabulous.
- Excellent control over the drum mix.
- Great collection of grooves and good editing options.

cons

- A bit daunting at first.
- Can be a little resource hungry but offers good customisation options to cope with this.

summary

If you are serious about your acoustic drum sounds, BFD2 is a serious tool for getting the job done. Well worth the asking price and well worth auditioning.

dry from the close mics, very wet if you mix in a healthy dose of the various room mics, or somewhere in between.

In all, some 10 full kits were recorded, along with a range of extras (mainly additional hi-hats, cymbals and snares), giving you over 90 kit 'pieces'. Samples from the original BFD and any of the expansion packs can be used with BFD2, and users can also load their own samples (including multiple velocity layers), should they wish.

The comprehensive nature of the sampling strategy used allows BFD2 to offers two key performance elements. Firstly, each kit piece contains an amazing number of velocity layers (up to 96), which means that machine-gun-effect drumming is never an issue. Secondly, for the key kit pieces, multiple articulations are available. For example, the snares all feature normal, off-centre, sidestick, rim and drag-style hits. This creates the potential for very realistic-sounding drum performances to be produced.

Time For Tea

Unlike some products, BFD2 is provided in a suitably environmentally-friendly small box containing the five DVDs of content. While the user can choose different levels of content installation, given the relatively low price of hard drive space it almost seems pointless buying into the idea of BFD2 if you are not going to use its full detail. However, you could imagine installing a somewhat lighter version on a second machine — perhaps a laptop system for use on the move. Fortunately, the challenge-and-response copy protection is suitably flexible to allow up to three installs and FXpansion seem happy to deal with issues arising from system upgrades. The authorisation proceeded without any problems on my Internet-connected test system, but can also be carried out from a separate computer if you like to keep the Internet away from your main music-making machine.

Given the quantity of the data, full installation does take some time, so make a cup of something suitable before you get started. The install stage is also a good point to start work on the PDF manual. At 180 pages, this is perhaps best described as 'comprehensive' and, although not the most exciting of reads, it does give a sense of the real depth of functionality that lurks within BFD2.

Full Make-over

The BFD user interface has also undergone a pretty substantial revamp. While the Kit page provides the default screen upon start-up, the user interface can be switched between this and four other screens — the Mixer, Grooves, Keymapping/Automation and Preferences pages — via the large buttons lined up top-left along the control bar.

The Kit page is split into three main areas. The large image of the drum kit is retained from the earlier version and the sounds associated with each kit piece can be auditioned via the mouse. The bottom half of the display shows the specific drums loaded into each kit-piece slot. To the left of this display are three small buttons that allow you to switch between three different kit sizes; 10, 18 and 32 pieces. The drum-kit display changes to reflect this. For the 32-piece kit it adds a 14-piece percussion ensemble to the 18-piece kit display, but since there are not many percussion instruments included as standard with BFD2 you will need to add the Percussion Expansion Pack (£159) to make full use of this.

Controls along the bottom of each kit-piece display allow gain (trim) and pan to be adjusted, as well as providing mute and solo buttons. The 'X' button clears the kit piece and ▶

reduces the RAM used (as indicated in the status bar that runs along the base of the main window). A range of further controls is displayed at the top right for the kit piece that's currently selected. These include controls for tuning, dynamics, damping, and a variety of other tweaks that can be made to the sound.

The other key element of the Kit page is the mini-mixer display located at the bottom right. For basic blending of the various key microphone groups — direct, overheads, room and ambience — this might be all you need. Master dynamics, tuning and humanise controls are also provided here.

While individual kit pieces can be loaded into a specific slot, loading a full kit (as well as various other elements of a BFD2 setup) can be performed via the Load button in the control bar. A BFD2 'preset' includes all the BFD2 settings, while a 'kit' loads all the kit pieces of any one of the 10 sampled kits. Mixer, Palette (a set of related Groove patterns), key maps and automation presets can also be loaded separately via this menu. While the relationship between these various elements take a little getting used to, they do offer tremendous flexibility.

All Mixed Up

If you need greater control over mixing than is offered by the mini-mixer on the Kit page, the Mixer page is the place to go. At first sight, this appears a little on the busy side but FXpansion have crammed a lot of functionality into a compact space. The display is split into two halves. The lower section shows the usual virtual mixer channels. The exact format of the display can be altered via the six buttons located in a vertical strip to the far left. For example, toggling on the FX Send button in this strip switches between larger faders plus a drum graphic on the one hand, and shorter faders with insert FX and FX send slots visible on the other.

The upper portion of the display shows any effects that are inserted into the currently selected mixer channel. Up to four insert slots are available and BFD2 includes a nice range of dedicated treatments covering EQ, compression, delay, filtering, modulation and drive, amongst a few others. The compressor effects, in particular, sounded good to me, although it is worth noting that these are proprietary effects — they cannot be used outside BFD2, nor can VST plug-ins be used within the BFD2 mixer.

The mixer supports auxiliary channels, so you could, for example, submix all the snare mics to an aux channel and apply effects to just the aux channel, rather than having to insert compressors and EQs on all the individual snare mic channels. As BFD2 offers up to eight stereo and 16 mono outputs, it is also possible to pass individual drum mic outputs to your host sequencer, if you would rather do your mixing and effects processing in that environment. However,

For a mega kit-and-percussion combination, the 32-piece kit should be just the ticket, although you will need the Percussion Expansion Pack to really make the most of it.

The Mixer page provides a very flexible mixing environment.

Drummer Auditions

There's a handful of 'drummer-in-a-box' type products that are obvious competitors to BFD2, including Toontrack's EZ Drummer, Steinberg's Groove Agent, Digidesign's Strike, XLN Audio's Addictive Drums and Submersible's Drumcore. Each offers its own combination of features. For example, Groove Agent provides a wide range of drum-kit samples (acoustic and electronic), as well as an extensive collection of preset patterns and fills and, via MIDI, can be used to build a complete drum track or be used as a playback engine. Submersible's Drumcore also has an interesting take on the idea of the virtual drummer, combining audio loops and MIDI-triggered drum samples, with a series of 'drummer packs' available for extra content. In terms of detailed samples of acoustic drums, the most obvious comparison is probably with Scarbee's Imperial Drums XL. This is supplied with Halion Player as the front end, and the 50GB sample library also features close, overhead and ambient mics that can be mixed to taste.

Test Spec

- BFD2 v.2.0.3 build 15.
- Athlon dual-core 4400+ PC with 4GB RAM, running Windows XP Pro (SP2), with TC Electronic Konnekt 24D and Echo Mia 24 interfaces and Cubase 4.1.3.


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the bottom line here is that BFD2 now provides an extremely flexible mixing environment and offers all the sorts of possibilities that you might get with a multi-mic setup on a real drum kit.

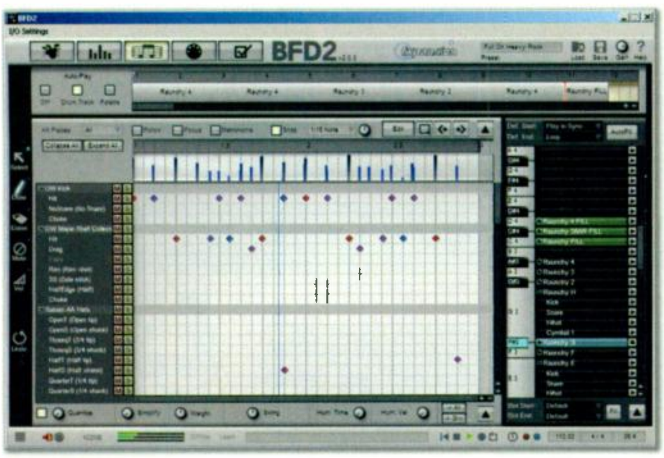
Groovy, Baby

BFD2's ability to work with Grooves has also been considerably expanded. Up to 128 Grooves can now be loaded at any one time. These appear mapped to keyboard notes on the right-hand side of the window (within the Groove Palette). Over 5000 Grooves are provided, and these can be previewed and loaded via the Load menu, either individually or in groups. The styles cover blues, country, rock, folk, funk, hip hop, house, drum & bass and African, and also a range of tempos.

The new Groove Editor occupies the bulk of the window. It's not dissimilar to the MIDI drum editors found in most major sequencers but includes dedicated lanes for the different articulations offered by BFD2, and can be collapsed and expanded as required. All the usual editing functions are provided, including a velocity lane that can be toggled on or off. By providing direct access to the different articulations, the Editor does encourage you to add embellishments, and it's amazing what the addition of the occasional dragged or half-edge snare hit can do to give a pattern a little more life. The various Groove FX functions are located along the base of the Editor window and include the very usable Human Time and Human Velocity controls, which add something of a random element to the performance.

Further flexibility is provided by the functions that allow you to audition a kit element from one Groove (for example, the kick drum) within a second Groove while it is playing back. If you like what you hear, the kit element can be dragged and dropped from the Groove Palette into the Editor where it can automatically replace, or be merged with, the current performance. This edited Groove can then be saved as a new Groove to an empty slot within the Groove Palette. You can, of course, create your own Grooves from scratch, either via the Editor or live via MIDI.

While BFD2 now uses a proprietary file format for Grooves, it can still open MIDI files, and Grooves can still be dragged and dropped onto a MIDI track in the host sequencer for arranging. However, the Drum Track, located above the Editor, provides an alternative approach, as Grooves can also be dragged and dropped here for arranging. They will then play back in sync with any host sequencer. While this Drum Track is perhaps mainly intended for those using BFD2 as a stand-alone application, or for those who simply want to quickly hear how two or three Grooves work together, it does work very well.



The highlights on the Groove page are the Editor and the Drum Track above it.

BFD2 provides extensive keymapping functions — great for using it with an electronic kit.

As with the Mixer window, there is an awful lot going on within the Grooves window and new users might find it takes a little time to explore. However, the flexibility and functionality are clear — this is a powerful editing/sequencing environment that matches the sophistication of the drum sampling with which it is used.

The Best Of The Rest

The MIDI Note Learn wizard within the Kit page provides an easy way to associate MIDI controllers with particular drum sounds but, for more comprehensive control, the Keymapping section of the Keymapping/Automation page is available. This would be particularly useful for those using an electronic kit to trigger BFD2 sounds. The Automation section of this page, as its name suggests, allows BFD2's controls to be automated via your host sequencer. Again, this is well featured.

Given the comprehensive nature of the samples, one might expect BFD2 to be a little hungry in terms of computer resources. Fortunately, alongside all sorts of other settings, the Engine section of the Preferences page allows the user to lighten this load. For example, under the Streaming Engine settings, a 16-bit mode can be selected and the number of velocity layers can be restricted. Having made changes of this kind, you can apply them with the Restart Engine button. This makes it easy to move between a 'lite' version of BFD2 while tracking and a 'full-on' version when performing a final mixdown.

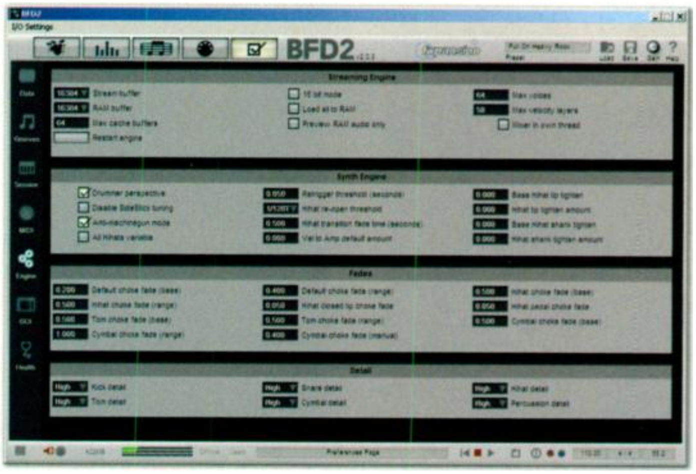
The other key feature worth discussing is the audio export, which allows a BFD2 performance to be rendered to a series of audio files. This process is initiated via the Export options in the Utility section of the Mixer page, and as any of the Mixer's channels can be armed for recording (via the 'R' button on each channel strip), it allows either a simple stereo mixdown or a full multi-channel drum recording to be rendered. The latter would be particularly useful either for transporting the BFD2 drum performance to a system where BFD2 itself was not available (when moving between studios, for example) or for archiving a recording project in an audio-only format.

Hit Me!

Of course, all this functionality is just so much froth if the drums sound like a bag of spanners in a tumble dryer. Fortunately, they do not; BFD2 simply contains some of

System Requirements

- **Mac: G5 or Intel CPU, 1GB RAM, 60GB hard disk space, Mac OS 10.4 or higher, DVD drive, internet connection (for authorisation).**
- **PC: Pentium IV CPU, 1GB RAM, 60GB hard disk space, Windows XP or Vista 32, DVD drive, internet connection (for authorisation).**



There are plenty of ways to reduce the load on the host computer, via the Engine settings on the Preferences page.

the best multisampled acoustic drum sounds currently available. I did the majority of my testing of BFD2 within Cubase 4 and the experience was a wholly pleasurable one — the plug-in version of BFD2 behaved extremely well throughout my testing. While it certainly does place a load on the host system, the flexibility offered by the Preferences page ensured that this never became a real issue.

The ability to drag and drop grooves into the host sequencer worked a treat and I found myself auditioning grooves, or small numbers of grooves together, within BFD2, before dragging them to a track in Cubase for arranging. This made it very easy to build complete drum tracks. Alternatively, the BFD2 Groove Editor is familiar enough, compared with the Cubase MIDI editing environment, to make composing or tweaking Grooves inside BFD2 a straightforward task right from the off. BFD2 is, however, a deep application and to get the best out of it does require some time — new users take note.

Conclusion

Given the combination of price and features, BFD2 is perhaps not for producers who prefer their drums pre-packaged as loops. However, for those serious about their acoustic drum sounds, BFD2 is an equally serious application. It might be a little daunting at first but this just reflects the sophisticated control on offer. The level of attention paid to the sampling borders on the obsessive, and for many users might be a little over the top, but the end result is a set of acoustic drum sounds that could grace almost any recording, with quality that would be well beyond what most home and project studio owners could dream of achieving in their own studios. These Big F***ing Drums are definitely B***dy F***ing Good! SOS

information

£ £242; upgrade from BFD, £124. Prices include VAT.
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Tascam US1641

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Put simply, if you want to send lots of audio inputs into your computer simultaneously, but don't want to pay too much for the privilege, the US1641 could be exactly what you're looking for...

Paul Sellars

The Tascam US1641 is a flexible, 16-input, 24-bit/96kHz audio interface, with basic MIDI I/O and the connection to the host computer provided via USB. The unit is housed within a solidly constructed 1U, 19-inch rackmount case and enjoys its own built-in power supply — something of a rarity in 1U rack interfaces at this price point. Supplied with the US1641 is a standard 'kettle' lead, a USB cable and a CD-ROM containing drivers and documentation. A printed copy of the manual is also provided, along with another CD, containing a copy of Tascam's CV Piano plug-in instrument and a DVD-ROM including Steinberg's Cubase LE 4 (see the 'Bundled Software' box for more details).

Installation

Setting up the US1641 is simple enough — the first step is to install the device drivers. Windows XP and Vista are supported (although only the 32-bit versions so far), as is Mac OS X (Tiger and Leopard, on both Intel and Power PC architectures). The manual recommends checking the Tascam web site for updates to both the device drivers and the device's firmware, and this was the first thing I did.

Installing the device drivers also installs the Control Panel software, which allows access to a few basic settings, such as Audio Performance (basic latency adjustments can be made between 'lowest' and 'highest'), Sample Clock Source ('automatic' means that the device will follow an external clock signal if one is connected, while 'internal' means that external clocks are ignored), Digital Output Channels (the US1641's digital output can have 'line out 1&2' or 'line out 3&4' routed to it), and Digital Output Format (S/PDIF or AES/EBU). Apparently the 'Audio Performance' option is not provided in the Mac Control Panel — although most audio applications provide their own buffer settings with which to adjust latency.

The Control Panel also displays the device's current firmware version. The review unit arrived with firmware version 1.0, so I decided to download the update to version 1.02. This takes the form of a small self-contained updater program and the update process is as simple as running it, clicking a button, and unplugging the device when prompted. When the device is plugged back in, the Control Panel reports the new firmware version. This worked perfectly for me, the whole process taking little more than a minute to complete.

Two Point Oh

It's important to note that the 1641 supports USB version 2.0 only — so although it is physically possible to connect the device to a computer with an older USB 1.1 port, you shouldn't do so and expect it to work. Tascam point out that the USB 2.0 protocol supports data transfer rates of up to 480 million bits per second, allowing greater bandwidth than either Firewire (400 mbps) or USB 1.1 (12 mbps).

The US1641 interacts with your computer's operating system as a 'multi-client' device. What this means is that it can handle audio output streams provided via different protocols — for example, ASIO and WDM — simultaneously. This means that on a Windows machine it would be possible to monitor the output of an ASIO application such as Ableton Live, a WDM application such as Sonar, and Gigastudio (which uses its own GSIF2 driver protocol), at the same time.

Connections

The US1641 provides a total of 14 analogue inputs and two digital channels. These can be used simultaneously, for a total of 16 ▶

SOUND ON SOUND

Tascam US1641 £279

pros

- Plenty of analogue inputs.
- Switchable phantom power for the XLRs.
- Cubase LE 4 provided free.
- The cost, or lack of.

cons

- Slightly limited MIDI functionality.
- Front panel knobs are a bit fiddly.
- Crackles when switching phantom power on and off.

summary

A flexible, multi-channel audio interface with more inputs for the money than most. Would suit a budget-conscious home studio with lots of analogue sources (and not too much MIDI gear).

input channels. Ten of the analogue inputs are located on the front panel: eight on balanced XLR mic inputs and two on quarter-inch jack sockets. The latter are switchable, operating as either unbalanced guitar inputs or balanced line inputs. The XLRs can supply +48v phantom power, which must be switched on or off in two groups of inputs 1-4 and 5-8 — so it isn't possible to switch phantom power on and off for an individual mic.

Ten small knobs allow the input gain for each of the front panel sockets to be adjusted — from -2dBu to -58dBu for the eight XLRs and from +4dBu to -42dBu for the jacks. Each one has an accompanying LED, which lights green when the signal reaches -30dBFS and then red as a clip warning at -2dBFS.

Keeping our attention on the front panel, there's a standard quarter-inch headphone socket with an accompanying level knob, and a separate monitor level knob. There's also a mix knob, which controls the balance of signals sent to the headphone outs. With the knob all the way to the left, a mix of the analogue inputs is heard. With the knob all the way to the right, the signal output from the computer via USB is heard. In intermediate positions a mix of the two can be heard, allowing for hardware 'zero-latency' monitoring of the inputs.

Four further analogue inputs are available on the back panel. These are balanced line-level inputs on TRS (Tip, Ring, Sleeve) quarter-inch jack sockets. The input level for these is switchable between -10dBv and +4dBu. The remaining two input channels are digital, in stereo S/PDIF format, on the usual RCA socket.

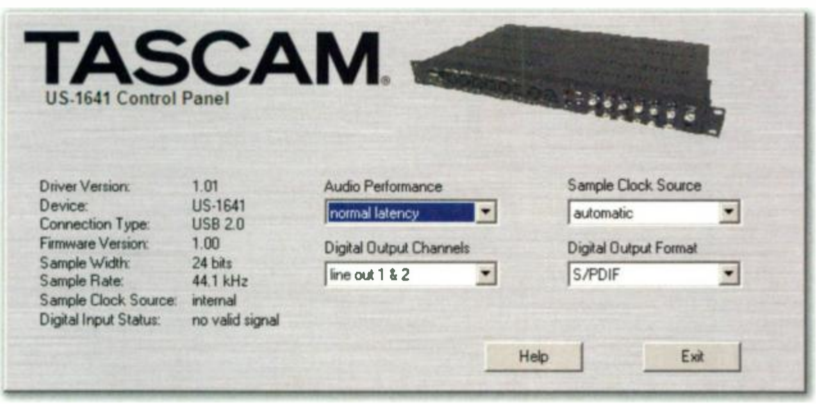
Outputs are less numerous, but there's still a fair selection: a stereo S/PDIF digital output, to go with the input; four quarter-inch TRS jack sockets, configured as two stereo pairs, and another pair that serve as dedicated monitor outputs. Finally, there's a standard USB port, and two MIDI sockets (one In and one Out).

The number and variety of inputs available make the US1641 a good bet for basic home studio applications. Recording 'live' takes of a band, for example, you might have four mics on the drum kit, two more on guitar amps, with bass and keyboards DI'd, and perhaps another mic for a guide vocal — and still have some inputs to spare.

One thing the US1641 lacks is serious multi-channel digital capabilities. The S/PDIF pair is certainly useful, but would be no substitute for, say, the eight channels available via an ADAT interface. MIDI I/O could also be more flexible than the basic one In and one Out provided. If your studio has a lot of external MIDI

Nvidious?

In an addendum to the US1641's manual, a slightly worrying note warns that PCs with Nvidia USB controller chips "may not offer optimum performance for audio streaming over USB 2.0, resulting in audio artefacts". Tascam's proposed solution is either to increase the application buffer size (and hence the latency) or to fit a third-party USB 2.0 card — so presumably this is not something that's going to be fixed by a driver update. PC users with Nvidia USB controllers be warned!



The US1641's Control Panel software is particularly simple, offering just four variable controls.

Alternatives

The Alesis IO26 Firewire interface offers similar features, in a tabletop rather than a rackmount unit, for about the same price. Edirol's UA101 is another USB 2.0 device and is also comparable, having slightly fewer inputs, but more outputs. M-Audio's Delta 1010 PCI interface offers eight analogue ins and outs, and two digital, plus basic MIDI I/O. However, it costs about £100 more than the Tascam and it is, of course, PCI rather than USB based.

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ADAM's unique folded ribbon diaphragm moves air four times faster than any driver in any other professional monitor, resulting in incredible clarity, breathtaking detail and imaging like you've never heard before. You'll work faster, better, and more efficient than ever.

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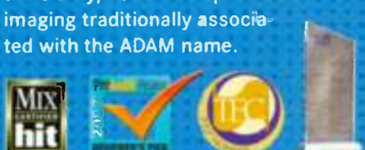
The multiple awarded A7 combines ADAM's renowned ART (Accelerating Ribbon Technology) folded ribbon tweeter with a state of the art 6.5" carbon fiber woofer, resulting in an extremely accurate monitor with all of the clarity, detail and spectacular imaging traditionally associated with the ADAM name.



P11A

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The P11A is a two-way shielded active monitor that produces unique imaging and outstanding transient response at an attractive price point. It was primarily designed for small to mid-sized project studios; however, it can also be used comfortably in any application where a small powered monitor is called for, from broadcast/post applications to surround monitoring.

"Hear them at your peril-you're unlikely to leave the shop without them!" (Sound On Sound, April 2006)

S3A

Member of S-series

The three-way/three channel S3A's utilize the ART tweeter to produce a wonderfully open sound field and pristine transients to 35kHz.

The award-winning S3A's have found critical acclaim from enthusiasts around the world, including professional recording studios in Hong Kong, European broadcast facilities, and major Hollywood film-scoring rooms and stages.

"The S3A is the best nearfield or midfield monitor that I have ever heard. They sound fantastic – but at the same time they are brutally honest, revealing aspects of my mixes that other monitors have failed to do." (Tape Op, Feb/Mar 2003)

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hardware, the US1641 may not be your first choice for that reason.

In Use

Sixteen inputs notwithstanding, the number of tracks it's possible to record simultaneously depends upon your computer's capabilities, in terms of hard disk performance, processor speed and available RAM. Tascam's specified minimum system requirements for Windows are a Pentium 4, AMD Athlon or equivalent processor at 1.4GHz or faster, at least 512MB of RAM (more would be better) and 1GB or more of free hard disk space. For Mac users the recommendations are much the same; a 1GHz G4 or 1.5GHz Core Solo processor is the minimum, and Mac OS 10.4 or later is required. In both cases a DVD-ROM drive and an internet connection are required to install and activate Cubase LE (see the 'Bundled Software' box).

I tested the US1641 on my 1.8GHz dual-core Pentium machine with 1GB of RAM; not enormously powerful by current standards. Even so, the US1641 performed well. Input latencies of under four milliseconds were easily achieved (by selecting the 'lowest' latency setting in the Control Panel software), making ASIO input monitoring quite feasible.

Setting up multiple recording buses in Cubase LE is easy, and in no time I was able to route signals from any or all of the US1641's inputs to different tracks as required. With all 14 analogue channels in use, a tiny bit of cumulative hiss was

As you can see, the rear of the US1641 is a pretty straightforward affair, as a good deal of the connections are found on the front. But what's that there on the far right? Yes — it's got an internal power supply!

sometimes detectable. Nothing excessive, and more noticeable when listening on headphones than via monitors. Turning down the front-panel gain knobs for any unused channels helped reduce this.

With Cubase configured for 16-bit, 44.1kHz operation, recording all 16 inputs to separate tracks simultaneously posed no problems. Adding effects gradually increased the load, but not problematically. Switching to 24-bit, 96kHz operation naturally made more demands of the system, and although there were no serious problems, I did start having to think in terms of larger buffer settings and fewer real-time effects.

In general, I found that recording and playback via the US1641 was clean and trouble-free. The only minor glitch is that activating, and particularly deactivating, phantom power for the mic inputs seems to produce an unpleasant crackle over the monitors — not loud enough to do any damage, but still noticeable.

Overall, the US1641 is easy to get along with. Connecting and swapping mics and other sources is easy, thanks to the front-panel sockets. Even if you are not likely to be using all the inputs simultaneously, you can treat the device as a basic patchbay, leaving different sound sources connected and switching between them as recording sources in your software as required.

One final grumble, though: the front-panel knobs are made of chrome-effect plastic, which is smooth and in fact slightly slippery to the touch. They're also quite closely-spaced (although this is all but unavoidable in a 1U device) and as a consequence can sometimes feel a bit awkward and fiddly.

Bundled Software

The centrepiece of the software bundle supplied with the US1641 is Steinberg's Cubase LE 4 audio and MIDI sequencing packaging. Cubase LE is a cut-down but still very powerful version of Cubase, offering 48 audio tracks (supporting 24-bit, 96kHz recording), 64 MIDI tracks, the ability to use VST effects and instrument plug-ins, and more besides.

Installing Cubase LE is more awkward than it really ought to be. Without going into excessive detail, I'll just say that the Syncrosoft challenge and response copy protection system is fiddly and slow, and that it would be a big improvement if the US1641 itself was used as a dongle (so Cubase wouldn't run without the device attached).

Also included in the bundle is a copy of Tascam's CV Piano instrument plug-in, based on their GVI software sampler (which itself is based on GigaStudio). It's an easy-to-use and pleasant-sounding piano instrument for VST and RTAS applications running under Windows (there's no Mac version). CV Piano is also available as a free download from www.tascam.com/products/gvi.html.

The US1641 comes bundled with Cubase LE 4, a cut-down, but nonetheless useful, version of Cubase 4.

Summing Up

The US1641 fits plenty of features into a small and sturdy box, and rivals some more expensive audio interfaces for flexibility. If you need multiple analogue input channels, and mic inputs with phantom power, the US1641 could be just what you're looking for; it's a very straightforward, workable package that's good value for money. While the 1641 is not entirely unique, there aren't many competitors in this price bracket, and Tascam's long experience in the field should certainly count for something. SOS

information

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Inside Track

Secrets Of The Mix Engineers: Joe Zook

OneRepublic's route to the top has been tortuous, and their massive hit 'Stop & Stare' was actually recorded more than three years ago. Engineer and mixer Joe Zook recalls the sessions.

Paul Tingen

Ever since Decca Records turned down the Beatles in 1962, tales of record companies getting it dramatically wrong have been commonplace. Columbia dropping OneRepublic in 2005 was a case in point. At the time, the band had just completed an album, produced by Greg Wells and engineered and mixed by Joe Zook, but Columbia apparently decided that it wouldn't fly and so the band was shown the exit door.

A year later, OneRepublic became the most popular music act on MySpace, mostly because of 'Apologize', one of the songs from the as-yet-unreleased album. Timbaland subsequently signed them to his Mosley Music Group label and included a remix of 'Apologize' on his Shock Value album. Jimmy Iovine, head of Interscope, reckoned that OneRepublic's album contained five singles, and co-signed the group. Timbaland's remix of 'Apologize' was released in the Autumn of 2007 and promptly went multi-platinum, topping charts around the world.

Nothing to be sniffed at, of course. But following this enormous success, it was still possible to see OneRepublic as a one-hit wonder riding Timbaland's coat tails. Their second single, 'Stop & Stare' put paid to that perspective. It has been a top 10 hit in at least 12 countries, including the US and the UK. The album, *Dreaming Out Loud*, was hurriedly released to tie in with the single, and also rode the higher echelons of the charts in a great number of countries.

If this appears an amazingly steep rise to the top, it has to be pointed out that several of the people involved in the making of Dreaming Out Loud already enjoyed considerable pedigree. OneRepublic's singer, songwriter, pianist and guitarist, Ryan Tedder, had already worked as a producer, and collaborated with Timbaland during 2002-04. As of today he has written, and sometimes co-produced, hits for Jennifer Lopez, Natasha Bedingfield, Leona Lewis and many others. Moreover, *Dreaming Out Loud* was recorded at Rocket Carousel Studio in LA, the studio of producer and musician Greg Wells, who at the time had already worked with Rufus Wainwright, Hanson, Louise Goffin and Natasha Bedingfield, and who has since gone on to success with Mika.

Blueprint

Another original participant in the *Dreaming Out Loud* sessions was Joe Zook, who has worked as an engineer and/or mixer for Tricky, Mika, The Hives, Modest Mouse, Pink, Lindsay Lohan, Courtney Love, Puffy Ami Yumi, Liz Phair, Remy Zero, Feeder, and many others. He recorded 12 songs during the mid-2005 sessions, and mixed nine of them, including 'Stop & Stare', in October of the same year, at his LA studio 4014 Mixing. After the band had been dropped and re-signed by Interscope/Mosley Music, another three songs were recorded by another engineer in late 2006 and mixed by Zook. Because three years have passed since the recording and mixing of 'Stop & Stare', Zook needed to extensively consult his recording notes and the Pro Tools Session file to remind himself what went on.

The Pro Tools Session for 'Stop & Stare'; drum tracks (below) were bounced to analogue tape at various different level settings to obtain tape compression.

"We recorded from 11 in the morning until seven at night, Monday to Friday," Zook recalls, "and we would do a song in about three days. It was very much a spontaneous process, and we wanted to get the songs down quickly and roll with it and commit. There was no pre-production. With each song we'd listen to the demo, talk about what we wanted to do, record the song live with the whole band for a scratch version, and then we started tracking the drums. We usually did the bass the same day or first thing the next day, then we'd record the guitars and third day the vocals and a rough mix. Occasionally there was a fourth day for some extra things. And then we'd move to the next song.

"For instance, I recall that we started on 'Mercy' on a Monday, and by the following, that song and 'Stop & Stare' were finished and rough mixed. We revisited 'Stop & Stare' two weeks later to add the chorus background harmonies. 'Apologize' was the third song we cut, and it was revisited several times. The band were very young, almost all of them were 20-22 years, and they'd hardly had any studio experience, yet they were all very competent and played really well. It was their competency that allowed us to work like this.

"For this record everybody was trying to create a blueprint rather than follow one. We wanted originality. For me, the songwriting and Ryan's voice provided a lot of inspiration for how to create something unique and

Joe Zook & The Unique Studio

Zook began his music career in his native Boulder, Colorado, but moved to Austin, Texas, at the age of 19. In 1996, he moved to Los Angeles, where he ended up assisting the legendary production duo of Mitchell Froom and Tchad Blake. "They did everything on 24-track analogue, 15ips, Dolby SR. Tchad Blake is the Jimi Hendrix of engineering to me. He doesn't do that much, it's all very simple, but even when you've seen him do it a million times, you can't duplicate it. When he left he would let me use his gear, and I would do everything I'd see him do, and it would still sound like a lame imitation. Tchad taught me to stop trying to be somebody else and just be myself." He moved on to work with the equally legendary Jack Joseph Puig for almost two years, and following this he went freelance, in 2002.

Zook established his own 4014 Mixing venture in 2004, with an unusual and extensive combination of state-of-the-art digital and obscure analogue gear. An unusual aspect of 4014 is the absence of a mixing desk, despite the fact that Zook professes that he "can't mix without analogue gear" and that he's "never been happy with an in-the-box sound".

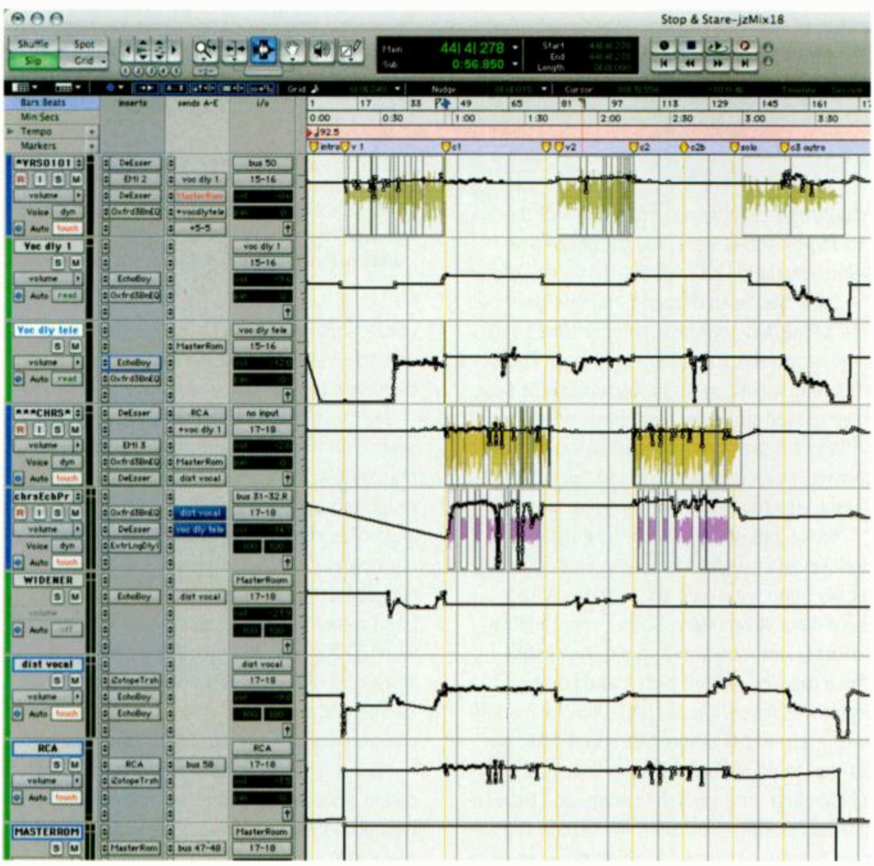
"The mean reason for having my own studio is that it's much more creative," he explains. "It can take five days for the record company executive to listen to your mix, and the band may be in one state or country and the producer in another, and the manager also wants a say, and any of them could ask for a recall. If you've been working in a commercial studio, you often won't be able to get back into that room several days later. At my own studio I take really great notes, and I take digital photos of all my gear, so I have been able to do fast recalls.

"I use a Command 8 for fader rides, I need to be able to get my hands on faders. I need the different colours that my outboard gives me, but plug-ins have become so much better during the last three or four years. You have to remember that someone like Dave Hill from Cranesong used to design tube preamps and half-inch machines 30 years ago. And now he does plug-ins. The same with Bobby Nathan of Unique Recording Software. Universal Audio do a similar thing. The stuff these companies make is good!

"The main problem in working in the box was the sound. I don't like summing amps either, so I was designing things and had things built for me. It's been a real struggle, but I ended up with something that I think is fantastic. It's not easy to summarise the way my studio works, but the basic idea is to do summing like on a vintage Neve console, with line level taken down and summed by a passive summing system, and then amplified back up with Neve 1271 and 1272 preamps. My system is based on 32 channels of Folcrum. Most of the OneRepublic record was done with preamps made from UTC transformers and original API 2520 amps, but for 'Apologize' and 'Say' I used Neve 1272 preamps for a slightly less gritty sound.

"I work with really great techs and talk to them about how I want it to sound and they build a few things and I'll pick what I like and tweak it until I'm hearing what I'm after. As far as I'm concerned it's still a work in progress. I'm inspired by people like Tom Dowd, who built Atlantic Studios, and (probably) invented what we know as faders in the process, along with some of the first eight-track recorders. All the great studios, from Abbey Road to United Western to Crystal Sound to Sunset Sound, were just making shit up that they thought sounded cool, because they couldn't just go out and get a great-sounding studio at the local music store. They all had a beautiful unique sound because of this. Making a studio should be like making a record: you don't just use the stock preset on everything. You tweak it and make it fit the way that you feel music."

interesting. I would work off these when searching for new sounds and things that were exciting. We recorded to Pro Tools, often via Greg's vintage Neve 5114 desk. We used a lot of the equipment in Greg's studio, and I also brought down quite a bit of my own gear. Because we wanted to record things fast, I decided to put up lots of different microphones in addition to the normal drum microphones that were set up in Greg's studio, and would then mute the ones I didn't want while tracking. I probably had 18 to 20 drum microphones up, of which about 10 were standard."



The vocal sound on 'Stop & Stare' was crafted using numerous delays and other effects.

Zook explains that he can't remember some of the drum and room mics that were used, saying "I'd never used many of these mics before. Greg had things set up, and he has good ears, so I trusted him. It was a matter of making what was there work. I recall that the room mics were Chinese MXL and the overheads were some kind of Gefell. The kick drum mics were an AKG D112 on the inside, going through an API 550 EQ and a Distressor compressor, and I had a Crown PZM mic in front. I also had an NS10 on the beater side, which was being pummelled and distorted, and went through a Universal Audio 2610 preamp and a Chandler EMI compressor. On the toms was, again, some Chinese mic, and there was probably a 57 or a Beta 57 on the snare. I also had a [*Shure Green*] Bullet mic on the snare, which I ran through a Deluxe Memory Man, a Ross flanger and a little Danelectro Nifty '50s amp, miked with an SM57. Bass drum, snare and overheads went through Brent Averill BAE 312a mic pres."

Dreams Of Fleetwood Mac

According to Joe Zook, the song 'Dreams' from Fleetwood Mac's *Rumours* album became a reference point for 'Stop & Stare', because "it has a similar quarter-note feel with hypnotic eighth-notes throughout the song". The engineer/mixer explains how this informed a number of choices that were made while recording and mixing. "When we recorded the drums, we ran into the problem that there was too much of a scene change between the verse and the chorus, when the drummer starts bashing the cymbals and this big drum room sound comes in. It wasn't working, and I remembered that on 'Dreams' the drums had been recorded without cymbals and the cymbals were overdubbed later on. So used that technique, and it worked great. It kept the mood throughout the song."

About half of the drum microphones went through the studio's Neve 5114 desk, while most of the remaining mics went through the BAE 312a preamps. An Alan Smart compressor and a GML EQ were used on the room mics. "We wanted it to sound like drums in a room, and it had to sound unique and somewhat dark in terms of mood, otherwise it would appear as if the drummer wasn't getting it. After recording the drums we moved on to the bass, which ended up being the scratch bass on almost all tracks. The bassist, Tim Myers, was really good and he nailed everything during these first scratch takes. I suspected that it would be possible to use the scratch takes and I had simply taken a DI out of the bass player's Ashdown amp head, going through a Neve V72 tube DI and preamp and an 1176 compressor. I recorded a lot of compression on some of the drum mics and on some other things when we knew we wanted to commit to that. The only exception was the vocals — we only compressed the vocal sound in the monitors, but didn't record it.

"The band have a unique approach to playing the guitar. Nobody is playing power chords and one of them is always playing high up the neck, single notes, with lots of delay. To give them a big guitar sound, the guitars needed lots of space around them. So I printed big stereo room mics. They were the same room mics as I'd used for the drums, in the same position, with the same signal chain, Alan Smart and GML EQ. I think the intro and the verse on 'Stop & Stare' were played via an open-backed Fender Deluxe in one room with two relatively close mics on it, a Royer 121 and a 57. In the other room there was a Marshall cabinet that was running through any one of the seven or eight heads that were in the control room, with effects committed to Pro Tools 90 percent of the time. Pedals were too numerous to mention, but 40-50 percent of them were delays using the Deluxe Memory Man. The guitar mic preamp was a Cranesong Flamingo, which went into an RCA BA25 compressor, from the '50s. It's all tubes, an old dinosaur kind of thing. Sometimes I was just running things through the RCA without adding an effect. In general the guitars were compressed very little, if at all.

"The acoustic guitar was recorded with my Dave Pearlman U47 mic. Dave is a few blocks away from me, where he rebuilds U47s. Before I bought the mic, I did a shoot-out with other U47s that I rented, and the Pearlman sounded incredible. The acoustic guitar mic also went into the Cranesong Flamingo and the RCA. I don't think I used any EQ. I simply moved the mic around until it sounded right. Recording acoustic instruments is a team sport. Nine times out of 10 a great acoustic sound happens when the player makes subtle adjustments, such as moving a couple of inches or changing picks. It's all in the hands of the player. I just try to stay out of the way of a great performance.

"The vocals were recorded with a Neumann U67. Ryan has one of these voices that engineers love. Whatever mic you put in front of him, he sounds like he is already EQ'ed. I wanted something soft and warm on him, and I didn't want him to start sounding thin when he went into the upper range. The U67 worked really well for that. Those mics need some EQ at their top end, in my opinion, but they always remain full in the mid-range, so they sound great when people sing high. We tried a few different preamps on him, and I was shocked to find that the Avalon 737 worked the best. I'm generally not a fan of the Avalon, but it was perfect on his voice. As boring as the Avalon ▶

compression is, it was very good at allowing an overall hotter level to be recorded without hurting the sound. I also added about 6dB around 32k, to give it more air. On the Avalon 737 the high-frequency shelf selections are 10kHz, 15kHz, 20kHz and 32kHz. The shelf is super-wide so when you grab 32k it extends pretty far down to audible frequencies. And that was it for recording. I didn't use plug-ins on the way in. Thirty tracks of drums were recorded, and 20 used after recording cymbals separately, the guitars were around 12 tracks, and there were six tracks of vocals.

'Stop & Stare'
Written by Ryan Tedder
Produced by Greg Wells

Joe Zook: "I find that mixing songs that I have also recorded can be tricky, because on some occasions I have tended to blow the track up and forgotten the song that's there. When I get tracks in that I haven't recorded I never have that problem, I usually get it right the first time. But with songs I've recorded I often go through the struggle of mixing them twice before I play them to anybody. I think it is because when you've recorded a track, you have preconceived notions about it. The challenge in the mix is not so much the sounds, because they usually are close to the way I want them, but more the levels.

"When I first mixed 'Stop & Stare', I made the drums and the bass massive. It became this huge grunge thing, and it just didn't sound unique or fresh. I'd started at five or six in the evening and I went home late at night being really proud of myself. But when I came back in the next morning, it was like, 'F**k, where did the song go?' There were all these great sounds, but it didn't feel right. The wall of jangly distorted guitars made it sound like from the '90s. I think I did email this mix to Greg just before I was about to take it all apart again, and he agreed, and said 'The rough mix is better.' I'd become too focused on all the unimportant aspect of the engineering, rather than the music itself.

"So I started again, and what I normally do in a mix is take my cue from the lead vocal, from what he or she is trying to do. Sometimes it even helps me to solo the vocal and listen to it from start to finish, without any effects. I just crank it up really loud, and I can hear how somebody is breathing, I can hear all the little noises that sometimes get lost in a track. I always get inspired by that, and I listen to whether different instruments want to support the emotion or juxtapose it. After putting up the vocal I bring in individual instruments and I ask myself what is helping the guy tell his story, and what is interacting and conversing with other tracks in the song. I like to find things that bounce off each other and converse. In this song it was the ambient verse guitars with the drum room sound, and eventually the vocal ambience. Once I'm clear on this, I pull everything back, and I start adding effects to the vocals. In the case of 'Stop & Stare' there were lots of them."

This composite photo shows some of the more obscure analogue gear at 4014 Mixing. Much of this was used on 'Stop & Stare', including the Collins 26U compressor (top), Cinema EQ (the dark unit with two large knobs about a third of the way down), Blonder Tongue B9b graphic EQ (centre), Gates Level Devil compressor (with the bright red light) and CBS Audimax II compressors (bottom).

Vocals: RCA BA25, Cinema EQ, Izotope Trash, Chandler EMI, Bomb Factory Cosmonaut Voice, Waves De-esser, Sound Toys Echoboy, Ross flanger, Vox AC30 amp and head, MicMix Master Spring

"I thought that Ryan's voice was really intense and emotional, and his consonants sounded really cool. So I put an RCA BA25 compressor on an insert on the vocal to bring that out. But when you are sending things really hot to the RCA, it starts to blow up and do really crazy things to the consonants, so I put a Cinema passive EQ before it. The Cinema was made in Burbank, probably in the '60s, and it has a high-frequency and a low-frequency knob, and you can get huge curves in the EQ. It has a beautiful tone and is wonderful for shaping vocals. I then added the Izotope Trash plug-in, to bring out some aggression. You can hear the Trash kick in in the choruses. This is called 'dist vocal.' If you put things on an insert you lose a generation, but since the effects are not subtle, who cares?

"On another vocal insert I had the Chandler EMI compressor, which I used more on the verses. I would then ride these two with the clean lead vocal. The lead vocal was actually doubled, and I had the Cosmonaut Voice plug-in on the double. The chorus double sounded too 'pop' untreated, like some other record. It didn't fit. So I really mangled the double vocal sound in the chorus with the Cosmonaut Voice plug-in to make it sound haunting. I don't know who makes this plug-in [*it's part of the Bomb Factory collection now owned by Digidesign themselves — Ed*], but it's great. It's a super tiny-sounding filter. It has pretty much only one sound, and a couple of seconds after the main sound ends it will either send out a walkie-talkie sound or a beep. So I always have to mute it immediately after using it to stop these noises coming through.

"Another plug-in I used was the Waves De-esser, because that is what I had at the time. I nearly rented the Dbx 902, but I got the Waves plug-in to work well enough. Today I use the Universal Audio De-esser in preference to any outboard de-esser. Then I also had three or four delays on the vocals.

The way Ryan is singing is so cool, it kind of cried out for little voices in the background with a kind of twisted, distorted, warbling sound. So I played with some freaky-sounding delays. You can't see the whole delay story from the one screenshot, you'd need four: volume returns, volume sends, return mutes and send mutes.

"To try to summarise the delays, I had two sets of delays using Echoboy plug-ins with different sounds and different timings. One had heavy modulation, distortion, repeats and EQ to emphasise the singer's frustration, and the other was more subtle and had less regeneration and seemed to emphasise hope. I just rode them in and out where it felt right, just a couple of passes and then some fine-tuning here and there. The 'widener' and 'voc dly tele' tracks are both mislabelled. I started using a widener on one, and tried for a telephone effect on the other, but ended up using an Echoboy delay, and never renamed the tracks.

"To get a heavier, more aggressive sound and a call-and-answer effect I also sent the vocal through a Ross flanger pedal, a Vox AC30 head and a 4x12 guitar cabinet, and then recorded it back in and cut it up and moved it in time. This wasn't as involved as it sounds, as I have it all set up in my studio through a routing matrix, so I can hear a variety of amps and heads and combinations of them in just a few minutes.

"These days I use Echoboy and several other delays, because there are now a bunch of plug-in delays that are good. I love the Massey TD5, it's an analogue-sounding thing. And I often use the Universal Audio Roland RE201. Actually, UA has four delays that sound great and that I use all the time. I've also been using PSP's copy of the Lexicon PCM42. Plus, I often do a slap tape echo with my Studer A827 on an insert. Oh, and I use delay pedals, like the Deluxe Memory Man pedal. I must have 15 different delay options!

"Finally, the reverb on the lead vocal is from the MicMix Master Spring Room Reverb. It's big, beautiful, warm, dark and just nails the emotion. I used a lot of it on the whole album. It's a big long thing. You can't control the time, it just has a big chamber sound. As you can see from the screen shot [*see page 136*], I also did a lot of rides on the vocals, just to get everything to sit right. I kept tweaking until it sounded good. I'll go down to one syllable. Sometimes eight words will sound amazing and then all of a sudden there's a moment, usually at the end of a word, where there's too much breath sucking, or something else that needs to be ducked. I am always looking for the emotion, and sometimes it comes up easily, other times you need to tweak a lot and do a lot of rides to get the maximum expression. This was one of those cases."

Drums: Valley People Dynamite, Empirical Labs Distressor

"As I said earlier, one of the biggest challenges of getting this song right was making the drums smaller. It's why we recorded the cymbals separately. When I did my first mix, the drums still were so huge, and the mood got lost. So I took all the faders down and took all the effects off everything, and stopped blowing it all up and then put the faders back up to where they were when I recorded them. This was much closer to what the song was supposed to be.

"The whole time I had been recording the drums I knew that it was missing analogue tape compression. So I went to Ocean Way, transferred the drums to an Ampex 124 and back to Pro Tools. The numbers at the end of the Pro Tools track titles are references to me recording to tape at different levels. So 18 was a little more than 14. You can see that I hit the rooms harder than the other elements, which had lower numbers, because I wanted to see how the transients in the room mics would respond to the tape. When I do this I'll sometimes cut between different levels. In some songs it may be better to have a more smashed-up sound in the chorus, for instance. But in 'Stop & Stare' the sounds go right through the whole song.

"The Bullet with which I had recorded the snare had lots of bleed on it, so during mixing I put a gate on it triggered from a snare sample. I also used the Valley People Dynamite and the Emperical Labs Distressor tube compressors on the snare and kick. I sent them out to a send and brought that back in. But for the most part the drums were recorded the way they sound on the record. If I needed more depth or another effect, I simply turned up one of the many microphones I'd used. There were other, tiny, cut-fader rides that I did, adding a few dB here or taking out a dB there. It was all about fader levels."

Bass: Sansamp Classic, Ashdown amp, Gates Level Devil, Blonder Tongue EQ, Collins 26U

"Because I had recorded the bass via a DI, I wanted the acoustic sound of an amp in there, so during mixing I ran the DI through a Sansamp Classic pedal and then through a Ashdown 900 head and a cabinet, one 18-inch with two 10-inch speakers, and a Crown PZM mic four feet in front of it, on the floor. It was pure violence! The ceiling might have caved in, even though you don't hear that on the record. There was not too much distortion, but it was working really hard. I wanted to get some depth and some air behind some of the attacks. I ran the DI and reamp mics through a Gates Level Devil — a '60s tube compressor — and my 1959 Blonder Tongue B9b Graphic Tube EQ. The Blonder is a weird, rainbow-looking thing and it's my favorite thing on the bass. I got the phase the way I wanted it, and also added a Collins 26U compressor, and that was the bass sound."

Guitars: Collins 26U, Cranesong Phoenix, Moogerfooger phaser, CBS Automax

"I didn't do anything to the verse electric guitars. They were were recorded the way we wanted them to sound. The electric guitars in the chorus were treated with a Collins 26U tube limiter. They brought out a bit of grain and added more weight to the single-note guitars. One electric was panned right and the other left, and the one on the right, which was played by Drew Brown, I used the Cranesong Phoenix Luminescent plug-in, which brings out the 300-800 Hz range and overtones, to give a nice full sound. The room sound that came with the guitar solo was treated with the Moogerfooger phase pedal. There were no other effects on the solo.

"Way underneath these guitars there are electric guitar overdubs in the chorus, playing open chords. I think one is a Telecaster and the other a Gibson 335 and they're playing through open-backed cabinets, probably a [*Fender*] Deluxe and a Twin. My first mix had these up loud with the acoustic in the middle. When I turned them down and the acoustic up, everyone was happy. As far as the acoustic is concerned, that initially wasn't quite working the way we wanted to. Ryan had played it four or five times, and I went back and grabbed another performance and stuck it on the track in the chorus. I think I used the CBS Automax tube compressor on it. Sometimes you have to go to great lengths to get things to work and I'll do whatever it takes to get there." SOS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Digidesign C24

Control Surface For Pro Tools

The Command 8 and Icon have revitalised the affordable and the professional ends of Digidesign's controller range. Now Digi have turned their attention to the middle of the market, with a replacement for their tried and tested Control 24.

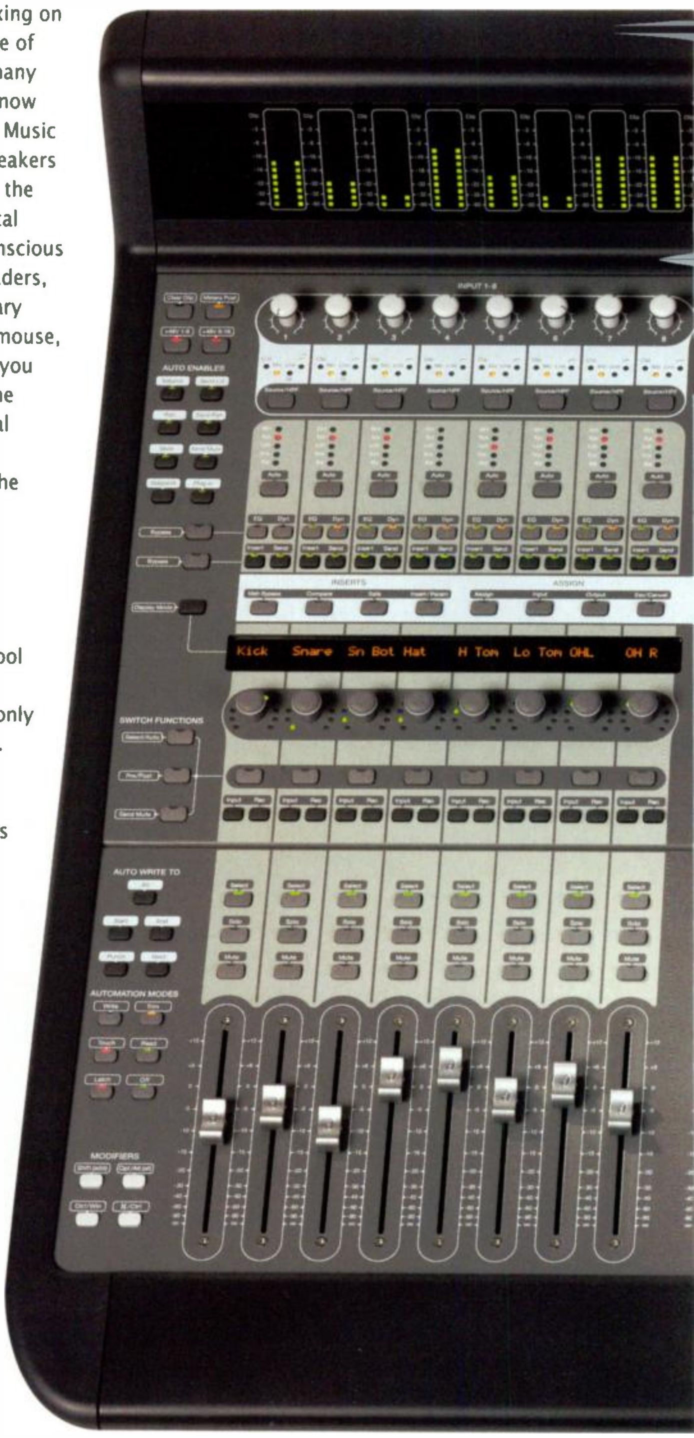
David Mellor

In days gone by, the centrepiece of any recording studio was the mixing console. You couldn't have a studio without one, and for preference it had to be huge. Then progress was made: the 1991 unveiling of Pro Tools marked the beginning of the transition from mixing by console to mixing by mouse. And indeed it is entirely possible to mix using a mouse-and-keyboard interface. Even the largest Pro Tools rig with a gazillion tracks can be controlled via an object no bigger than the palm of your hand.

But there is a drawback. If you have ever had the opportunity of mixing on a conventional analogue console of a decent size (and these days many people haven't), then you will know the joys of mixing instinctively. Music emanating from the monitor speakers enters the brain, filters through the arty, non-analytical, non-technical regions, and almost without conscious thought the fingers move the faders, and reach out and twist the rotary controls. When you mix with a mouse, you have to imagine the sound you want to achieve, then engage the brain's number-crunching central processor to make it happen.

And there is one thing that the mouse can never do, and that is balance one sound against another — the level of the bass versus the low-frequency EQ of the rhythm guitar for instance. This is a technique that old-school engineers will know very well indeed. With a mouse, you can only control one parameter at a time. One more example? Have you ever seen a highly experienced engineer drape his or her fingers over the faders and control several at the same time? The mouse would retreat back into its mouse hole, curl up and die.

The above operational reasons are enough to justify using a control surface in preference to a mouse, even though the control surface may be literally thousands of times the price. There is another reason... Although it is possible to make a complete multitrack recording and mix entirely within Pro Tools, a Pro Tools system in itself doesn't fully equip a 'proper' recording studio. And by 'proper' recording studio I mean one with a separate control room and recording area, where you can record a band in an efficient, productive manner and mix efficiently, too.



SOUND ON SOUND

Digidesign C24 £7044

pros

- Intuitive tracking and mixing
- Powerful automation facilities
- Easy access to frequently-used functions
- Goes a long way to eliminating reliance on keyboard, mouse and monitor
- Digidesign's experience in control surface design shows.

cons

- Some operations will require a learning process, but the massive capability of Pro Tools makes this an inevitability.

summary

For a Pro Tools user, buying a control surface is a matter of 'when' rather than 'if'. With the introduction of the C24, 'now' could be the time!

Among other features, a talkback mic will be needed; a listenback mic is nice. There will be alternate sets of monitors to be switched, external stereo inputs to listen to. Oh yes, and you will need to have your head up and be actively communicating with the other people in the room, not focusing your concentration deep into the monitor screen.

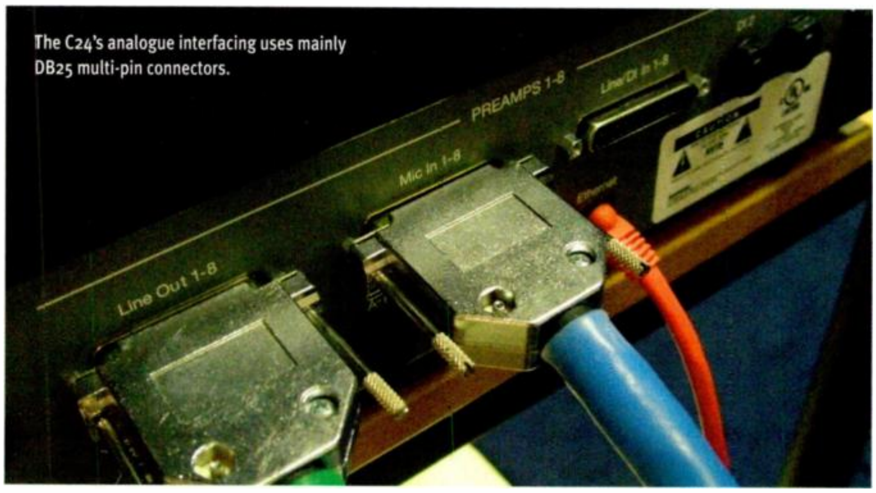
So, in a nutshell, the above reasons summarise why you should consider supplementing your £5 mouse with a seven-grand control surface. You really should. Whether you have a full-blown Pro Tools HD system, or Pro Tools LE running through a 003 or even an Mbox interface, the C24 could be the control surface for you. Want some more reasons? Read on...

An Analogue Console? From Digi?

It may seem counter-intuitive that the C24 is largely an analogue audio device. But this is indeed so, simply because all the audio digits are inside your Pro Tools rig, to which C24 attaches via an Ethernet cable. The digits within the C24 are there to control Pro Tools and display parameter values, not to store or transport audio. So on the rear of the console you will find a multitude of analogue connections. Let's explore...

Any analogue mixing console will have a good number of microphone inputs. We have come, over recent years, to see the microphone preamplifier as an outboard device. But the C24 puts it back where it belongs: inside the console. The C24's 16 preamps boast a frequency response that is flat within 1dB up to 100kHz, equivalent input noise of -127dBu and total harmonic distortion plus noise of 0.004 percent at 1kHz. These specs, though we might have ▶

DIGIDESIGN C24



The C24's analogue interfacing uses mainly DB25 multi-pin connectors.

liked to see distortion figures measured up to 20kHz, spell C-L-E-A-N. And of course there is no reason why you can't connect your boutique preamp as well. Naturally, these inputs can be switched to accept line levels or high-impedance instrument inputs too, and there is an additional eight-input line-level submixer, which we will look at in due course.

You might expect microphones to connect to the C24 via XLR connectors, but this is not so. The majority of inputs and outputs are on DB25 connectors. Since there are 13 such connectors you should make a generous allowance in your budget for cabling, or prepare for a marathon soldering session. You can, of course, buy adaptor cables from Digidesign!

The C24 has plenty of inputs; it has plenty of monitoring options too. There are, for instance, connections for no fewer than four monitor input sources — two six-channel surround and two stereo. There are two cue inputs and two outputs, one of the outputs being six-channel surround too. In fact, when you think about how much I/O capability the C24 provides, it starts to look like very good value almost irrespective of its control functions.

One feature that may be simple yet exemplifies the C24's dedication to traditional studio practice is the studio loudspeaker outputs. Yes: the band can listen to their take on loudspeakers (via a power amp of course) without having to traipse into the control room. Any engineer with experience of band members invading the control room will greatly appreciate this. I don't want to make too much of this one point, but it is a small nicety that anyone who didn't understand studio practice would overlook. Its inclusion demonstrates that Digidesign do indeed understand, and we can expect the rest of the facilities of the C24 to be as well in tune.

Hooking Up

Now that we know the ins and outs, let's examine how the C24 can be integrated with your Pro Tools system, and the rest of your studio.

The first thing to remember about the C24 is that it is not an interface to your Pro Tools system; it works with your existing interface. So the C24 will send analogue signals to and from your Digi 003 Rack, for instance, but it is inside the 003 that the A-D and D-A conversion takes place. So your microphones will connect to wall boxes in your recording room (it's a 'proper' studio, remember!) and the wall boxes will connect to the mic preamp inputs on the C24. Your line-level sources can connect to the line-level inputs and your guitars to two separate input jacks. The outputs of the preamplifiers are analogue and at line level, and they connect to the line-level analogue inputs of your interface. Clearly, if you have an interface with only eight inputs, you can only send eight channels to it. You can route any combination of eight, though. Actually, I'm a little bit surprised not to see an ADAT optical output on the C24, but evidently Digidesign made a policy decision not to do this — they wouldn't just forget.

The analogue outputs of the 003, or whatever interface you're using, connect back to the Pro Tools inputs of the C24. Typically you might configure two outputs to be your stereo monitor outputs, and two more can be your cue outputs (pre-fade auxiliary sends) for foldback. Obviously, monitor and cue levels are controlled — in the analogue domain — by the C24.

Engineer's View

That's enough time spent round the back. Once the C24 is installed, you shouldn't ever want to see it again anyway. Let's look

Control Surface Universe

Digidesign have a lot of experience in control surfaces, and they make an impressive range. At the top end is the Icon, which integrates Pro Tools into the truly massive D-Control or merely large D-Command control surfaces. You have to see the D-Control in the flesh to realise that Digidesign's intention was to grab the large-format console market. "Is your SSL getting a bit long in the tooth? No problem, a D-Control will replace it nicely, and its size will impress your clients just as much." The D-Command is smaller, but still satisfyingly chunky.

At the compact end of the range (it doesn't sound right to call it low-end) is the Command 8, which features just eight faders in a desktop package. And then there is Digidesign's earlier generation of control surfaces, from which lessons have been learned. Perhaps you have a Pro Control, or a Control 24. If so, you might be wondering whether it's time for an upgrade. The Pro Control, which was discontinued in 2004, was Digidesign's former flagship control surface. If you have one of these, the D-Command or D-Control is your upgrade path. If you have a Control 24, which is still in some dealers' warehouses, then the C24 will definitely be your upgrade. If you find all these 'Controls' and 'Cs' confusing, well so do I. Don't worry — all will be well once your shiny new control surface is installed.

What do the Icon D-Command and D-Control have that the C24 doesn't? Pro Tools users like myself want Digidesign to stay in business making all that lovely software and hardware for a long, long time. So we don't mind that they need a differentiated product range where people who have more money to spend can avail themselves of more facilities, and it's a simple fact that the more expensive Icon range must be able to do more than the C24 can. This is what the C24 is missing:

- **All rotary controllers on the D-Control and D-Command are touch-sensitive, so true touch automation is possible. Only the faders are touch-sensitive on the C24.**
- **The D-Command and D-Control have dedicated EQ and dynamics sections. However, they are physically very much larger consoles.**
- **The D-Command and D-Control allow custom fader groups. It is possible to group non-contiguous channels, or lock certain channels to certain faders while other faders are paged.**
- **The D-Command and D-Control allow plug-ins to be mapped to faders.**
- **The D-Command and D-Control have 'custom masters mode' and 'VCA spill'. (In the interests of brevity, at least two-thirds of a page of difficult explanation is hereby omitted!)**
- **The D-Command has two and the D-Control has six rotary encoders per channel, allowing immediate access to more parameters.**

Perhaps a good summary is that the D-Command and D-Control are both like mixing consoles with a DAW built in, while the C24 is a control surface for a DAW.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at the C24 from the engineer's point of view, which is much more interesting.

The C24 has a meter bridge, as you would expect — although you might have expected it to be an optional extra, as would be more usual. Thankfully, it's built in as standard. The meters reflect the on-screen software meters, so they don't tell you anything you couldn't find out otherwise, but they are exactly where you want them to be and I wouldn't want to be without them, nor the clear timer display.

Just below the meters are the controls for the microphone preamplifiers. I'm not going to be old-fashioned and carp that the gain controls are not calibrated in dB, because the preamps are fully integrated into Pro Tools and all you have to do is wind on enough gain, as shown on the meters directly above. There is of course a clipping indicator. Just one button control, which cycles through Mic, Mic with high-pass filter, Line, and Line with high-pass filter, is provided, and there's no phase button, but hey — that's the job of a plug-in! Phantom power is switched in blocks of eight preamps. Personally I don't find this slight lack of flexibility an issue but I know there are other shades of opinion on that.

You might have noticed that this is a 'C24' — note the '24' — and there are only 16 mic preamps. Well, space that might otherwise have gone vacant is used for an eight-channel line-level mixer. This has its own pair of output jacks, so you can do what you like with it — you can connect it to a pair of inputs on your interface if you like, or you can route internally direct to the monitor.

I'll skirt around the channels for the moment, because clearly they will need special attention. To the left of the console you will find a lot of buttons that are relevant to automation. If you currently mix with your mouse, you probably spend more time inserting and editing automation on screen than controlling it with the faders. But when you have a control surface the whole point is to use the faders and other physical controls. Why else would you be spending all this money? So Digidesign have really gone to town on this. There are no fewer than 19 buttons controlling automation globally. These cover automation modes, enables and 'write to'. 'Write to' takes the current values of automation parameters and writes them to other parts of the tracks, in a variety of ways — all the way to the end, for instance. You will use it a lot when it's right there in front of you ('write to' is only available on Pro Tools HD, unfortunately).

To the right of the console there is the monitor section, which is very well equipped with controls for stereo and surround monitoring, alternative speakers, cue (foldback), talkback and listenback. There is no oscillator! (You'll have to use a plug-in if you need to generate test tones.) Next come four buttons for the four edit modes in Pro Tools: Shuffle, Spot, Slip and Grid. View controls select the default view of the console (where the rotary encoders in the channels control pan), the previously selected view or a view of an external Digidesign microphone preamplifier unit, if you have one connected.

Refreshingly, the meter bridge is fitted as standard.

There are editing buttons that pull out important features of Pro Tools into the control surface, such as Trim, Select, Grabber, Pencil and the Smart tool. You can select audio, scrub, trim and perform edits right from the console. My instinct is telling me that because editing is so easy using the mouse and the smart tool, the mouse will still find itself popular for this application. I sense that with a little familiarity one's mouse hand and one's C24 hand would find themselves coordinating very nicely on this. Consideration should be given to creating some 'mouse room' right next the C24. You won't be using the computer keyboard an awful lot, but easy access to the mouse will be useful.

But, you know, I'm going to stop myself there... Doubtless, if you look hard enough you will find some function you occasionally use not represented on the control surface. But you won't be able to complain about it because Digidesign have done a pretty damn good job of placing everything you need right under your fingertips. And, by the way, the transport controls have a very pleasant, positive feel too.

The Channel Strip

On a conventional analogue mixing console, the channels are vertical strips that work independently of each other. That is so here too, except that it is also possible to display and control plug-ins using the channels' LCD displays and rotary encoders spread out across the width of the console. Despite the fact that there is so much going on inside the C24 that is of an analogue nature, the digital side of its character brings in multiple-function controls. This part of the operation of the C24 will require a learning process. Clearly, in an ideal world it is always preferable to have only one function per control. The human brain responds well to that. But the functionality of Pro Tools is immense. To map that to single-function controls would require a console of enormous size. It would not be even remotely feasible on the compact C24.

Let's start by looking at the facilities provided within the channel strip. At the bottom (you thought I was going to start at the top!) is the fader. I start with this because it is pretty much the *raison d'etre* of the control surface. I know plenty of people who do mix by dragging attractively

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Manley Mini Massive

Passive Stereo Equaliser

This EQ borrows circuitry from the classic Massive Passive design, but can the Mini really be 'massive'?

Hugh Robjohns

Getting hold of Manley products in the UK has been rather a struggle until recently, but as distribution now seems to be properly sorted out, we decided to celebrate with the first Manley review to appear in *Sound On Sound*. I looked at the Langevin Mini Massive, a dual-channel, two-band, solid-state equaliser. Launched in 2006, it is closely related to its bigger and older sibling, the Manley Massive Passive EQ (which was introduced in 1999 and has been likened to a kind of supercharged Pultec). However, the Mini Massive fits into 1U of rack space and costs over £1000 less than the Massive Passive in the UK.

More Than A Name

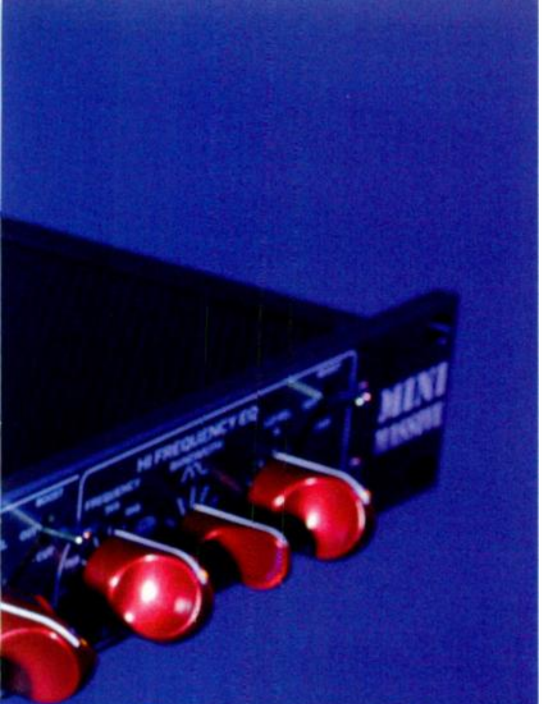
The similar name is no accident — the Mini Massive shares much of the same passive EQ circuitry and components that are used in the top and bottom bands of the classic Massive Passive four-band design. However, it also includes several extra features and circuit refinements that are intended to make it better suited to mastering applications. As well as being smaller and cheaper, the Mini Massive complements the Massive Passive by being cleaner sounding, while offering similar EQ characteristics.

Instead of using valve amplifier stages (as the original Massive Passive does), the Mini Massive uses two of Manley's own bespoke 'Rapture' amplifier stages in each channel. These are encapsulated, discrete, solid-state units that were apparently designed for a cost-no-object D-A converter, with the aim of being the 'cleanest and most musically involving gain stage.'

Attention to detail is evident throughout the product, and especially so when you examine the large circuit board. The EQ sections and signal paths are adjusted via sealed relays, the power supply incorporates both series and shunt regulation, with completely independent sections for the two channels, and premium audio-grade components cover the high quality PCB. Four small daughter circuit-boards carry additional relays and a few passive surface-mount devices, but the majority of the design uses conventional components.

Much like its bigger brother, the Mini Massive is designed to be run with very high signal levels and it will happily accept +26dBu on the input and provide up to 30dBu at the output. The down side of this approach is that the noise floor is relatively high, at -74dBu, but if you are prepared to drive it hard, and can cope with the high output level, it will return a dynamic range of 108dB, which is very respectable indeed.

Rather than risk compromising the output quality with a one-size-fits-all output stage, the Mini Massive can be switched on the rear panel to provide either balanced or unbalanced outputs at a nominal +4dBu, or



Photos: Mike Cameron

SOUND ON SOUND

Manley Mini Massive **£1756**

pros

- Musical and versatile sound.
- Clean and fast solid-state gain stages.
- Configurable output drivers.
- Adjustable-shelf EQ slope.
- Separately bypassable EQ sections.

cons

- Potentially noisy if not driven hard.
- Transformer mode switch on rear panel.

summary

A relatively simple, two-band, two-channel equaliser, but with some very well thought-out features and facilities, and exceptional build quality.

an unbalanced output at -10dBV — with each option being driven optimally. There is also an option to have transformer outputs fitted, employing the same 9811 transformer as used in the Massive Passive. Another rear-panel switch allows the transformers to be bypassed altogether, or used in one of two modes. The first introduces a fairly subtle 'warmth' (called Iron) while the second ('vintage') exaggerates the effect to bring a more obvious vintage character designed to emulate 'classic British consoles'.

Mini Tour

Starting at the rear, the fixed-voltage mains supply is via a standard, fused IEC inlet socket. Towards the two outside edges of the rear panel are I/O connectors for each channel. male XLRs for the outputs and combo-XLRs for the inputs (where the jack socket is designed to accept an unbalanced -10dBV source).

A pair of rather exposed three-way toggle switches configures the output stages of both channels simultaneously. One switch adjusts the output level between +4dBu (balanced or unbalanced), and unbalanced -10dBV. The second changes the mode of the transformer output (if fitted) between bypass, 'Iron' and 'Vintage.'

The low-frequency band can be switched between 22Hz, 33Hz, 47Hz, 68Hz, 100Hz, 150Hz, 220Hz, 330Hz, 470Hz, 680Hz and 1kHz, while the high-frequency band provides options of 560Hz, 820Hz, 1.2kHz, 1.8kHz, 2.7kHz, 3.9kHz, 5.6kHz, 8.2kHz, 12kHz, 16kHz and 27kHz, So, as you can see, there are closely-spaced frequency options for precise control, and with a generous overlap between the two bands.

"This equaliser is just brilliant for the gentle tonal shaping and moulding of a sound."

Moving around to the stylish black front panel, with crisp white legending, the controls are grouped into four clear sections, with distinctively sculpted red rotary controls and silver toggle-switches.

In the centre, the two toggle switches are used to turn the unit on and bypass the entire equaliser circuitry. A dual-colour LED illuminates when the power is switched on, showing green when the EQ is active and red when it is bypassed. To either side are the two equaliser band controls for each channel. In each case, there's an 11-position rotary switch to select the frequency, and continuous rotary controls governing bandwidth and gain, the latter providing up to 20dB of gain.

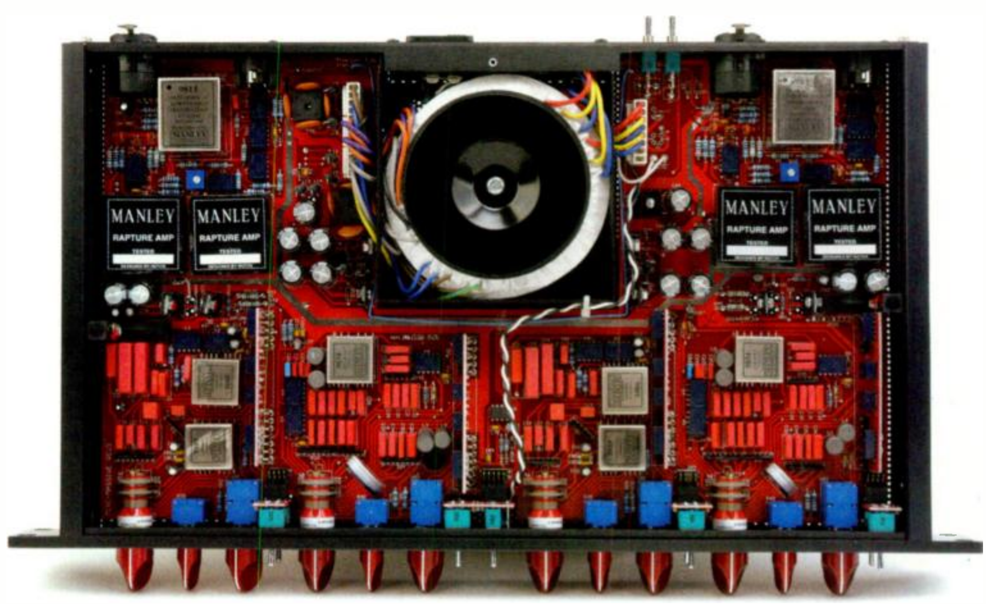
The equaliser can be changed between shelf and bell modes with a toggle switch, while a second toggle can be used to flip between boost or cut modes, and there's also a central section-bypass position. For the high-frequency bands, which are broadly similar, the EQ-mode switch has three positions, allowing you to choose a second bell option that offers narrower bandwidth settings for the top four frequencies.

This combination of range and overlap also makes it perfectly feasible to chain the two channels together to make a single four-band equaliser, if necessary.

An unusual (but very welcome) feature is that the bandwidth control works in both bell and shelf modes. Its function in the bell mode is obvious, but in shelf mode it acts to adjust the slope of the shelf, with a slight overshoot at the narrowest setting.

Apparently, Craig Hutchison, Manley's chief designer, has tweaked the parameters of the bottom four frequency settings of the LF band, and the top four of the HF band, compared with the Massive Passive, as well as adding the higher-Q bell option for the four highest frequencies. The idea is to let the bottom end sound even fuller and more Pultec-like, with more options for air and sparkle at the top.

Like the Massive Passive, the two EQ bands' controls are interdependent to a degree, because they operate in parallel instead of in series. The response is also different when cutting to when boosting, so if you cut and boost the same frequency using two bands, the result isn't flat. All of



A look inside the Mini Massive reveals a high-quality PCB covered in premium, audio-grade components, along with four small daughter circuit-boards that carry additional relays and a few passive surface-mount devices.

MANLEY MINI MASSIVE

which makes this a tool to use with your musician's ears, rather than with your engineer's hat on!

Massive Sound?

The Mini Massive is intended primarily to meet the needs of simple stereo tonal sweetening, either at the mix stage or in mastering. However, as I suggested above, the range of the EQ bands will also allow more comprehensive treatment of a single mono source by daisy-chaining the two sides of the unit together.

The Massive Passive has a reputation for sounding, erm... 'massive', with a particularly full and rich bottom end. I didn't have one alongside to compare directly, but the Mini Massive probably sounds very similar in range and scope, because the EQ circuitry is very similar, but I suspect that it's also a little cleaner and tighter, thanks to its solid-state gain stages.

Switches on the rear panel allow you to select either balanced or unbalanced outputs at a nominal +4dBu, or an unbalanced output at -10dBV.

The optional output transformers were fitted to the review model and the benefit is clear to hear. In the bypassed mode the sound is very open and extended at both ends of the spectrum, with a tight bottom end, and a clear and bright top. It sounds very modern, with crisp, clear transients and an overall neutral sound character that can be compared with the best solid-state units, such as the Millennia and GML offerings.

Switching to the 'Iron' position, to bring the output transformer into play, seemed to close things down a little — not dramatically, but you can hear it. The bass end becomes softer but fractionally richer and the mid-range slightly less clear. It's a subtle effect, although it becomes more pronounced if you daisy-chain the two channels with the transformers switched in.

Stepping up to the 'Vintage' mode, the whole effect becomes far more obvious and pronounced. When driving the output hard you can hear the saturation effects creeping in as the sound becomes compressed, and there's a lot more harmonic distortion going on. It is musical and interesting, and it definitely adds a whole new sonic flavour, but I don't think you'd want to use the effect all the time: I found it just that bit too strong for everyday use, but it would be ideal for thickening up some sources and for effect. All of which makes it such a shame that the switch is hidden away on the rear panel. In fact, I wouldn't be surprised if some people modified their units to bring that switch out to the front panel (this would be quite easy to do).

The equalisation is, above all else, musical and remarkably controllable. The small frequency-intervals allow accurate, repeatable tuning, with no fewer than 48 separate settings between the two sections per channel! The bandwidth control in bell mode provides a very useful range from wide to narrow, for broad tonal shaping or fine surgical precision. I also liked the fact that the level control uses the full 270 degree rotation to provide the 20dB gain range, with a separate switch to determine whether that is used for boost or cut. Essentially, it provides twice the resolution of conventional centre-detent boost-cut controls, and that really allows some accurate fine tuning.

The provision of the second choice of narrower HF-band bell-curve settings is interesting, and allows an alternative flavour of high-end sweetening. This facility isn't available on the Massive Passive, so I presume it has been added in response to user feedback on the limitations of the original. I particularly liked the ability to control the slope when using the equaliser in shelf mode. This enables a trade-off to be made between turnover frequency and slope, to determine just how much the mid-range is affected by the EQ curves.

Of course, it's almost impossible to describe the sound of an equaliser in words, but the Mini Massive is never less than musical. It did a fantastic job with gentle, wide-bandwidth shelving boosts at the top and bottom to provide the classic 'smiley' curve, freshening and lifting a mix. In bell mode, similar settings were able to bring out the fullness and punchiness at the bottom end, while enhancing the air and sense of space at the top end. Gentle mid-band cuts, carefully tuned to the right frequency, were also very effective at reducing the clutter in some mixes and improving perceived clarity.

Essentially, this equaliser is just brilliant for the gentle tonal shaping and moulding of a sound. I used it mostly in the content of full mixes, when broad, gentle brush-strokes were required. However, it is equally versatile when applied to individual instruments, and in this context you can often afford to be more heavy-handed, especially when cutting unwanted resonances or booms. In this kind of application I often found it necessary to daisy-chain the two channels, just to gather enough resources to tweak in the way I wanted, but the results warranted the extra effort, and on basses and guitars, daisy-chaining with the transformers in vintage mode brought some pleasing colour and richness to the party as well.

Alternatives

High-end two-channel equalisers abound, and choosing between them is very much a matter of personal taste and expectations. However, priced competitively with the Mini Massive and still offering a combination of solid-state clarity and some sonic flavour, the AMS Neve 8803 takes some beating. The Decca-pedigree DAV BG3 Mastering EQ is another solid favourite to audition.

If you want 'that' valve warmth sound but can't run to the full Manley Massive Passive, the hybrid Summit Audio EQP200B would be a good choice.

Conclusion

The Mini Massive is unlikely to be as characterful as the Massive Passive (how could it be without valve gain stages and with two fewer EQ bands per channel?), but that's not to diminish it in any way. This is a very sweet-sounding and musical equaliser that is fundamentally clean and neutral in character, but with plenty of weight and punch available at the bottom end, and a bright, airy nature at the top. The optional transformers bring some body and edge to the sound, and are a useful extra feature to have. Overall, it may not have the ultimate flexibility of some multi-band designs, but what it does have is a very fine degree of controllability for general tonal shaping, combined with superb ease of use. SOS

information

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WIN

The SOS £20k Dream Studio prize draw at the SRT show

The news you've all been waiting for...

The line-up of prizes for our massive Dream Studio gear giveaway, which will take place at the inaugural SRT show, part of the London International Music Show, has been confirmed. We've teamed up with 20 pro audio exhibitors to bring you 30 fabulous prizes worth over £20,000 that, together, will equip you with the home studio that's the stuff of dreams.

The prize centres around a super-fast, eight-core **Apple** Mac Pro with a 23-inch **Apple** Cinema Display running Logic Studio. Connected to this will be an **Apogee** Ensemble Firewire audio interface for all A-D and D-A conversion duties. Complementing the ample software suite in Logic are two additional DSP hardware processors: **SSL's** Duende PCIe, which fits inside the Mac Pro, and **TC Electronic's** Powercore X8, which connects via Firewire. These will open up even more options when choosing plug-ins.

Giving the winner hands-on control over their mixes will be the **Euphonix** MC Mix, the latest in control surface technology, while a **Roland** Fantom G6 will act as the master keyboard and a hardware synth/sound module. Additional data-entry equipment in the form of an **Akai** MPD32 and an **Alesis** Control Pad will enable the user to program their beats using MPC-style pads, or using good old-fashioned drum sticks, and a **Yamaha** Tenori-On — the first in the world to be offered as part of a prize draw — will let the winner sequence grooves using its innovative flashing matrix interface.

A generous handful of microphones are included, with the superb **Rode** K2 valve mic, **SE Electronics**' promising new SE4400a, a matched pair of **M-Audio** Pulsar II small diaphragm condensers (with the fabulous new **Rycote** Invision shockmounts keeping them rumble free), the solid all-rounder KSM27 from **Shure** and the mic that no decent studio is without: the **Shure** SM57. Mic preamplification will be supplied by the new **Allen & Heath** ZED 22FX compact mixer which has direct outputs that can be patched straight into the **Apogee** interface, but also from the **TL Audio** Ebony A1, a twin-channel preamp and DI device. If you're after even more bespoke mic preamplification, an as-yet unannounced **SSL** XLogic product enables further mic preamps to be used. For the guitarists among you, an **Emerson Williams** Bluestone Pro dummy load and DI box (a limited SRT '08 edition, no less) will let you get the sound of your amp at full volume without waking the neighbours.

Monitoring for the bundle will be handled by a **Presonus** Monitor Station feeding a pair of **Genelec's** 8040 nearfield monitors, which will be perched on **Primacoustic's** Recoil Stabilisers. **Ghost Acoustics'** Studio Kit 3 will help tame any nasty room modes and reflections, while two pairs of **M-Audio** Q40 headphones will allow performers to monitor signals from the DAW without sending spill into the mics.

Korg's brilliant DSD recorder, the MR1000, will ensure your audio backups retain the best possible audio quality, while the **Edirol** R09HR portable recorder will give you the chance to record 24-bit/96kHz audio anywhere at any time. Last, but by no means least, an **M-Audio** MIDISport 2x2 will let the winner plug any of their existing MIDI equipment into the shiny new rig.

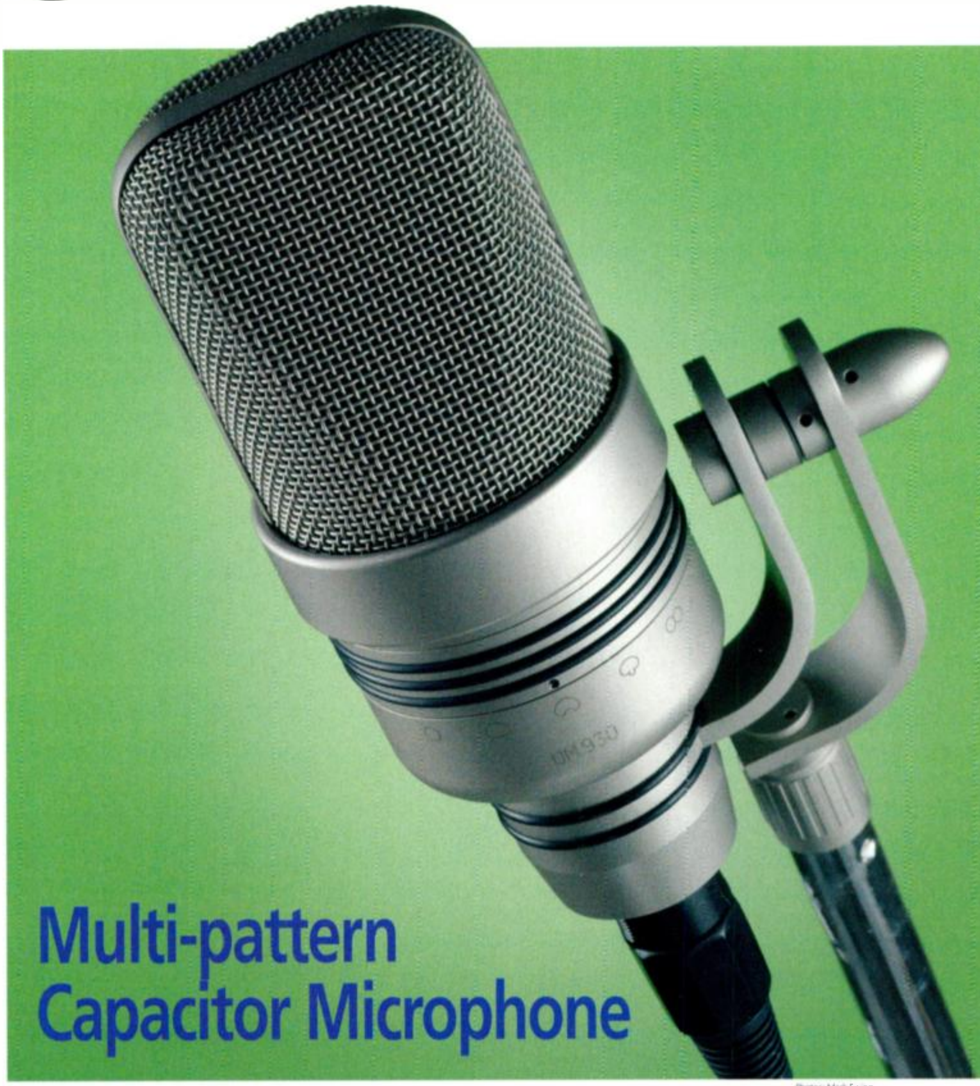
But there's more. The lucky winner of the home studio bundle will get their gear installed by SOS Editor In Chief Paul White and Technical Editor Hugh Robjohns using cables and priceless gadgets supplied by **Studiospares**, and they'll be featured in Sound On Sound Magazine! To top it off, **Alchemea** will supply the winner with fully certified Logic Pro training, making sure that they're a whizz on their brand new system.

Important! 1. Entries can only be accepted using the official entry system on the Sound On Sound booth (F10) at the Sound Recording Technology Show 2008. 2. The winner will be chosen after the show and notified by telephone or email. 3. Only one entry per person is permitted. 4. All prize winners must have suitable premises for the studio installation and agree to make themselves and their studio space available for the installation of the equipment on a mutually agreed date before 1st December 2008. All prize winners further agree to allow coverage of their studio installation to appear in Sound On Sound magazine and the web sites of contributing companies. All prizes shall remain the property of the prize draw organisers until these terms have been fulfilled. 5. By entering the prize draw, entrants agree for the details they submit to be passed on to contributing companies. 6. Employees of SOS Publications Ltd and their immediate families, and employees of companies contributing to the prize are ineligible for entry. 7. No cash alternative is available in lieu of the stated prize. 8. The organisers reserve the right to change the specification of the prize offered. 9. The judges' decision is final and legally binding, and no correspondence will be entered into.

Microtech Gefell UM930

At nearly twice the price of a Neumann U87, you've every right to expect this to be a very, very nice microphone...

Hugh Robjohns



Photos: Mark Ewing

Back in *SOS* January 2004, I reviewed the Microtech Gefell M930 large-diaphragm cardioid studio microphone, and was very impressed. However, there are times when it's useful to have something other than a cardioid mic, and the versatility of a good multi-pattern mic cannot be ignored. So the launch of the new UM930 intrigued me.

Overview

The UM930 is designed for the full range of studio and broadcast applications, including spoken voice, singing vocalists, and countless instrumental duties such as recording acoustic guitars and keyboards, percussion, wind and string instruments — and so on.

The mic offers five polar patterns (omni, wide cardioid, cardioid, hypercardioid and figure-of-eight), which are selected using a large ring just below the grille.

> **SOUND ON SOUND**
>
> **Microtech Gefell UM930**
> **£2535**
>
> *pros*
> - Very low self-noise and huge headroom.
> - Beautifully engineered and solid.
> - Interesting SCT and dual-output Twin models.
> - Various shockmount options.
> - Sound quality lives up to the price.
>
> *cons*
> - That price...
>
> *summary*
> A beautifully produced, multi-pattern, solid-state, large-diaphragm mic with a relatively neutral and accurate character and impressive technical specs. Only its cost counts against it.

Usefully, the side-address UM930 incorporates a discreet, green LED behind the grille on the on-axis side of the mic, along with the model number and polar-pattern symbols. The manufacturer's logo is etched onto the rear of the mic (and many people actually rig MG mics to face the wrong way because of that!).

Although sharing a similar model number to the M930, and appearing to have the same sort of grille and body proportions in the advertising photographs, the UM930 is positively elephantine in comparison. Whereas the M930 is roughly 120mm long and 45mm in diameter, the UM930 is roughly 40 percent bigger in all dimensions. It measures some 158mm long, is 65mm in diameter, and weighs a very chunky 930g (compared with the M930's delicate 210g).

Gefell's clever optical DC-converter technology from the M930 is used in the new mic too, and its technical specifications are very similar, with the same low self-noise of 7dBA, the same maximum SPL of 142dB (for 0.5 percent distortion), and fractionally lower sensitivity, at 20mV/Pa. Standard phantom power is required, of course, from which the mic draws 4.5mA.

Decisions, Decisions...

Available in either satin-nickel or dark-bronze finishes, the review UM930 was equipped with an integral, dual-axis 'elastic suspension'. However, a more conventional cradle shockmount is also available, as is a simple, single-axis microphone bracket. Whichever support is chosen, the mic is supplied in a traditional wooden box.

There are three versions of this mic: the Studio Condenser, and the Soundcheck Tool and Twin versions. The Soundcheck Tool (SCT) sends a 1kHz test signal down the cable

Alternatives

High-end multi-pattern mics abound, but the standard has always been the Neumann U87. For similar money, the Brauner Phantom V is an interesting alternative, and Neumann's TLM170R is a firm favourite. The AKG C12VR is in the same cost ball-park as the UM930, as is Gefell's switchable-pattern valve mic, the UM900, and — for ultimate pattern versatility — so too is the Soundfield SPS422.

if the polar-pattern switch is left midway between any two positions for more than about five seconds. The test-tone generator emits tone equivalent to a sound pressure level of 74dBA (20dB below the reference standard of 1 Pascal). The other model (the UM930 Twin) provides two simultaneous outputs. This mic is fitted with a five-pin XLR instead of the standard three-pin type, and it provides a fixed cardioid pattern on one output and the normal switched-pattern output on the other, thus allowing the user more flexibility in comparing polar patterns when recording. Another useful feature (on all models) is that the black rings around the pattern selector can be replaced with one or more coloured O-rings to aid identification. Green, red and blue rings are available.

In Use

The mic feels very robust, solid and — dare I say it — expensive. The polar-pattern switch is nicely weighted and detented for a smooth, positive action, and the sealed reed-relays used to change patterns should give a lifetime of reliable service. The output mutes for a few seconds when switching patterns (and the green LED turns off), to allow the electronics to stabilise, after which the LED illuminates again.

The published polar patterns are all fairly tidy and consistent up to about 8kHz — but above about 12kHz they collapse towards a narrow figure-of-eight response (as with most dual-diaphragm mics). All patterns have a broad 4dB presence peak centred at about 12kHz, but as the patterns become more directional the frequency response tilts more strongly up towards the HF. The 5kHz point is only 2dB higher than the 50Hz point for the omni pattern, but it's 4dB for the cardioid, and almost 8dB for the figure-of-eight pattern. The proximity effect counteracts this in most applications, however, and the rising response is not as obvious in use as the various plots suggest. The broad presence peak is nicely judged to give clarity and air without exposing sibilance.

In cardioid mode the sound is very similar to the M930, and the character is clean and reasonably neutral. I acquired some lovely recordings, ranging from male vocal to solo cello and silver cornet, and the mic's size and styling impressed the vocalist enormously!

Nice, But Pricey

Sonically, I can't fault the UM930, but I feel its price is a concern — even if you don't opt for the 24-carat gold version (yes, really!) — given that it is almost double that of the industry-standard Neumann U87ai, and over three times that of the fixed-pattern M930. The UM930 offers far more versatility and easily outperforms the U87 technically, but even so I think many will struggle to justify the cost. SOS

information

£ UM930 Studio Condenser £2535; UM930 SCT £2693; UM930 Twin £3316. Prices include shockmount or elastic suspension and VAT.

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Studio SOS

A dead live room, a live control room and faulty monitors... sounds like a job for the Studio SOS team!

Paul White & Hugh Robjohns

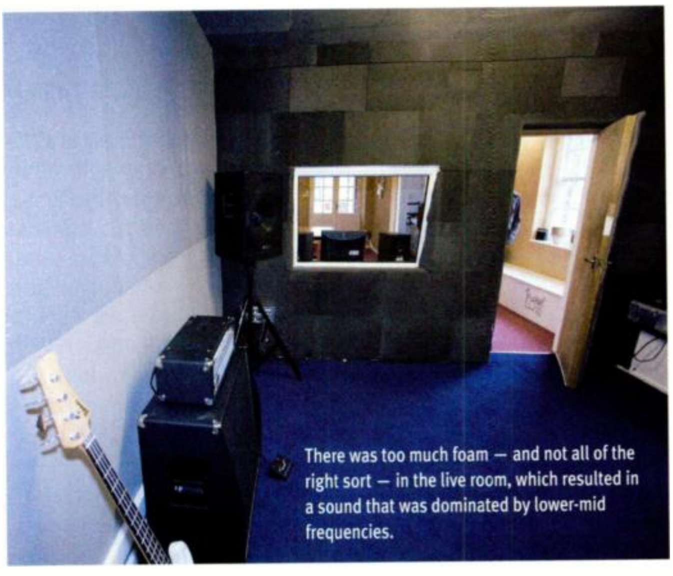
On completing his music technology education, Adam Dengel rented a property with the aim of setting up a rehearsal room and recording studio (DC Studio). The rehearsal part of his business seems to be ticking over just fine, but the studio he built for his own recording projects (and with the aim of attracting some commercial work) hasn't been performing as expected — and acoustics problems seemed to be to blame. Adam's studio is tucked away right in the centre of Barnsley in Yorkshire, and as I was born and educated there (but left because I couldn't learn the language!), I was keen to have a look around and see what had changed.

Clearly Adam's studio building has been there a long time: it's made from Yorkshire stone, with the familiar black coating that accumulated during the industrial and coal-mining boom. He has lined the walls of the live room with plasterboard on batten and then glued foam over every surface apart from around half the ceiling and the floor, which is carpeted. The end wall behind the drum kit is covered with Studiospares foam (around two inches thick) as is his approximately 2m x 2m vocal booth. I estimated that the half-inch industrial flat foam used on the rest of the room would have little effect below 1kHz or so. The control room was, by contrast, completely untreated when we arrived, and sounded distinctly live, so we were going to need to apply some treatment there to kill the worst of the reflections around the monitoring area.

Adam's Studio is based around Pro Tools M-Powered running on a PC, with a Project Mix I/O interface and a pair of Tannoy Mercury M2 passive monitors. These are fed from the powered sub of an M-Audio LX4 system and Adam also has the original LX4 satellite speakers, though one of the tweeters had been been damaged, so he replaced this with a similarly sized unit from the local Maplin store. He'd also bought a switching box for his speakers, hoping to be able to switch between the M-Audio and Tannoy monitors, but he hadn't been able to get it to work.

Keep Music Live

I'd already been in touch with Adam via email to discuss his live-room problems, and from his description it was evident that his over-enthusiastic foam treatment was killing all the high end but leaving lower frequencies effectively untreated, which meant that the sound in there was boxy and lifeless. As luck would have it, the panelling he'd built over the original walls was providing some accidental low-frequency trapping, but the room was still dominated by the lower-mid range. Short of ripping out all his hard work and starting again, it wasn't possible to come up with a perfect solution, but as the room was to be used mainly for recording drums and electric instruments we felt that a practical compromise could bring about a significant improvement for very little outlay.


There was too much foam — and not all of the right sort — in the live room, which resulted in a sound that was dominated by lower-mid frequencies.

Our strategy was to put some high-frequency reflection back into the room to balance the low end.

This was all very well in theory, but we needed to find something to use for this. At home I had a partly-used pack of rigid plastic foam sheets designed to go under laminate flooring. They were a couple of millimetres thick, and I reasoned that if these were glued on top of the foam they'd bounce some high-frequency energy back into the room, while still allowing lower frequencies to be absorbed by the flexing action of the panel on the foam. I brought along half a dozen of these sheets, which measured roughly 1200 x 600mm each, and we pinned them up to test their effectiveness in this role. Happily, they did exactly as we'd hoped, so we glued them to the foam on one of the side walls using Auralex spray adhesive, spacing them apart to help randomise the effect. The windows and untreated doors in the room also supplied some useful reflection. We didn't have enough panels to treat the opposite wall, and we weren't able to source any more at the local Wickes DIY store, so after dragging Hugh out of the electrical section (where he was buying some mains flex for use as speaker cable — because it's cheaper but just as good as dedicated speaker cable!) we decided to buy four 1200 x 600mm sheets of the thinnest MDF we could find and try that. Adam also bought a can of matte black paint to make them look less like boring sheets of MDF but, as he only had one can, we didn't have enough to paint them properly and ended up settling for a textured pattern. Adam said his girlfriend was an artist and could fill in the panels with something more interesting later! We stuck these panels directly to the foam (again, using the Auralex spray adhesive) and the difference in the feel of the room when speaking became noticeably less oppressive.

Adam's DIY vocal booth was a decent size and actually sounded pretty even — possibly due the panel construction beneath all the foam, and the fact that the booth was fairly large and irregularly shaped. However, as the singer faced the booth window, we felt it would be advantageous to cut down

Using a combination of plastic sheets and MDF panels, some high-frequency reflections were added back to the live room, without compromising the absorption of mid and low frequencies.

reflections from that quarter by putting an SE Reflexion Filter behind the mic. Sonic Distribution had kindly given us a few to use on our *SOS* visits, so we set one up after our usual rejigging of the mounting hardware to get the centre of gravity of the combined mic and filter somewhere close to the centre of the mic stand.

We Have Control

That left the control room, which we decided to deal with after a spot of lunch. This being Yorkshire, we had pork pies and corn-based synthetic pork scratchings rather than our usual feast of chocolate Hob Nobs! We'd brought four sheets of Auralex foam with us and decided to use these to kill the worst of the reflections around the plaster-walled listening area that was set up on the short wall of the rectangular room, but rather right of centre due to the location of a door. We fixed one panel to the nearest (right-hand) side wall at the main reflection point, split another panel to put next to the M-Audio monitors, which were mounted on small shelves, and used the remaining two panels above the control-room window and on the ceiling between the mixing chair and the monitors. The improvement in the stereo imaging was very noticeable and the liveness of the mixing area had been reduced, though some further absorption towards the rear of the room (which must have been six or seven metres long) would also be beneficial. A heavy curtain over the entrance doors would be an easy solution, and some slatted blinds over the left-hand windows might help to control reflections from there. A leftover piece of foam from Adam's store room was propped up in the corner to the left of the mixing position but we left it to Adam whether to fix this permanently or buy another piece of Auralex.

Having done what we could with the acoustics, Hugh set about installing the speaker switch-box using the twin-core cable we'd picked up from Wickes. This worked fine, though the subwoofer level setting was inevitably going to be a compromise, due to the significant difference in sensitivity between the Tannoy speakers and the M-Audio satellites. The sub's fixed crossover point of 140Hz was also a little too high to integrate properly with the Tannoys. The trick was not to turn the sub level up higher than strictly necessary, to fill out the lower octaves.

With everything up and running, switching between the Tannoys and the M-Audios revealed a curious out-of-phase character to the sound — not quite out of phase but definitely not in phase — and it was impossible to get a stable image from the M-Audios. Hugh checked all the wiring to make sure there wasn't a silly cable error, but everything checked out. Deliberately swapping the positive and negative wires over on one of the M-Audio satellites didn't cure the problem either.

At first we thought the replacement metal-dome tweeter Adam had installed in one of the M-Audios was the problem, as it did sound brighter than the original soft dome fitted to the other speaker. Clearly, the two speakers would have different frequency response and dispersion characteristics, but this problem seemed to be something more than that. After further listening using the white noise tracks on Hugh's BBC test disc, we thought it worth checking the tweeter to see if it was wired in the wrong phase relative to the woofer.

Unfortunately, Adam had poked the wires through the tweeter terminals before soldering them, and as soon as we got them hot enough to desolder, the plastic holding them just collapsed... so it was off down the road again to Maplin to buy another tweeter (which, at £2.99 each, wasn't too punitive). The plan was to buy two identical tweeters, so that the monitors would at least behave in the same way, but unfortunately there was only one in stock, so we had to settle for that.

Sub-standard

Before fitting the tweeter permanently, we tried wiring it in the opposite polarity to the previous arrangement, and the sound from the speaker heard in isolation was definitely better. However, it now sounded 'properly' out of phase when a mono signal was applied to both speakers. A quick examination of the woofer showed that the bass-unit wires were soldered to the driver rather than to the expected spade terminals, so rather than risk damaging it while trying to swap the wiring over, we reversed the speaker wires feeding the cabinet instead, and were rewarded by a far more satisfactory result. The left speaker was still slightly brighter, because of the replacement tweeter, but at least everything was now in phase, despite the terminals on the back of the box suggesting otherwise. Adam said he hadn't touched the woofer wiring, so there was no way he could have put it out of phase, but examining the two speakers showed that one ▶

Paul White performs radical open-tweeter surgery on a faulty speaker.

was internally connected using spade terminals and the other using soldered joints. As the system was originally an ex-demo model it may be that some earlier repair had been carried out, resulting in the internal wiring error.

Originally the subwoofer was placed under the desk, midway between the two speakers, but the panelled rear and sides of the desk created a resonant cavity that tended to boom. So to get the bass sounding as even as possible, we moved the sub out and stood it on the floor to the left of the desk, close to, but not exactly at, the centre of the wall. The improvement was very noticeable, with a much tighter and more even bass end. Adam had been worried about doing this, as he thought the bass might sound off-centre. With such a high crossover point that would be a risk, but the far smoother response easily outweighed any minor imaging issues, and in fact it seemed to integrate pretty well with the desktop speakers. The vibrations we could feel in the desktop were much less severe after we moved the sub, too, and we also put the Tannoy speakers on a pair of Auralex MoPads rather than the thin scraps of foam Adam had been using. The angled foam inserts were arranged to direct the tweeters up towards the engineer's head, which was another useful improvement.

Drum Recording

While we were in Maplin looking for replacement tweeters, Adam also picked up some interlocking heavy foam mats. When we returned, he assembled them to use under the house drum kit, to help prevent creep and to minimise noise getting into the structure. He was using AKG C1000 mics as drum overheads, which are less than ideal in this application, as they have quite a modest high-end response for a capacitor microphone, so he was looking at replacing these with some alternative small-diaphragm mics when his budget allowed. In the meantime, we found some foam offcuts and made holes in the middle. We could then slide them over the backs of the overhead mics, to further cut down on ceiling reflections getting back into the mics, as experience has shown that in smaller rooms the drier you can keep the overhead mics, the better your chances of producing a good, tight drum sound.

Adam's final query related to his clip-on AKG drum mics, which are back-electret models. He found that these always produced too much level, overloading his mic amps. He'd bought a pad adaptor to try — but this had a mono jack output which he'd fed into a line input, overlooking the fact that the mics need phantom power to operate. Even if they had been dynamic models, the combination of a pad and the reduced sensitivity of a line input would still have meant he got virtually no signal. What is actually needed is a balanced in-line XLR pad that still passes phantom power, or a separate bank of preamps with more headroom (or switchable pads) for use with the drum mics.

In contrast to the live room, the control room was largely untreated, so it was difficult to make accurate judgements about the sound.

Job Satisfaction

After a few hours work, the performance of both the live room and control room had been improved to a worthwhile extent at minimal cost. The original live-room problem was due to over-treating the walls with a material that was only effective at upper-mid and high frequencies, so it threw the overall tone of the room out of balance. Normally, foam-style treatments work best if applied to no more than 25 to 30 percent of the total wall area, rather than everywhere, and the thicker the foam (or the further spaced from the wall it is), the more effective it is at lower frequencies. By reintroducing some simple high-frequency reflection we helped to reinstate some kind of spectral balance.

The control room had no treatment at all, and while a rigorously-designed solution could be quite complicated and expensive, the simple expedient of controlling the strong reflections around the monitoring position, using Auralex foam, made a big improvement. Moving the sub and putting the Tannoy monitors on MoPads also tightened up the sound and made the bass end sound more even. By way of upgrades, obtaining better active monitors as soon as possible should be a priority, closely followed by upgrading the drum overheads for small-diaphragm capacitor models that have a nominally flat response up to 20kHz or above. SOS

Reader's Reflections....

Adam Dengel: "As a regular reader of *SOS*, it was a nice suprise for me when Paul and Hugh agreed to come to the studio armed with their years of knowledge and experience: after all, this was only the studio's second year of business. I'd invested a lot of time into trying to set up a decent music facility for the Barnsley area, but I'd hit a brick wall with my limited knowledge of acoustics, and needed advice.

Adam Dengel, in his control room after the Studio SOS treatment.

"The acoustic treatment in the mix room is fantastic, the monitor response is incredibly clear, and the room itself no longer sounds like a cave! I've been mixing a lot of tracks in Pro Tools this week and the difference is obvious. The bass end especially is more even, making it easier to mix. The monitor switch that Hugh installed for me has also helped my approach to mixing.

"The live room sounds brighter and it has a nicer 'feel' to it, especially when recording drums. I decided to leave the half-sprayed boards as they were, because I've had numerous positive comments on Paul's artwork!

"Finally, the SE Reflexion filter works a treat with vocal tracks, as sound bouncing off the window has been greatly reduced. Thanks again to the *SOS* team, Auralex and Sonic Distribution for helping and improving DC Studios."

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Overloud Breverb

Formats: Mac RTAS, VST & AU; PC RTAS & VST

Overloud's Breverb plug-in is designed to offer high-quality algorithmic reverberation, but with a low CPU load. In fact, Overloud claim to have opened 120 stereo instances on a 2.4GHz Core 2 Duo Macbook Pro, which is pretty impressive! It can be used in mono, mono-in/stereo-out or stereo-in/stereo-out configurations, supports all the major plug-in platforms, and is authorised to an iLok key.

Four algorithms emulate Halls, Rooms, Plates and Inverse reverbs. All are apparently designed to replicate the sound of classic hardware reverbs, so I might hazard a guess that they've used the classic Lexicon sound as their inspiration. I was rather hoping Breverb would also have a dedicated Ambience section, but unless one is planned as an addition for the future, I seem to be out of luck.

Once authorised or launched in its time-limited demo mode, the plug-in opens looking like a desktop remote control. You can have it expanded to show expert controls or keep it simple with just a handful of key parameters accessed by the knobs. The expert parameters cover different things depending on the algorithm selected but typically provide more EQ or diffusion options, as well as allowing adjustment of any internal modulation the algorithm may use. The expert section also lets you create gated reverb effects, with the usual gate controls joined by Shape and Slope parameters with which you can tailor the release curve.

The core knob-controlled features are divided into five pages, accessed by tabs. Three small buttons allow the fader section to be hidden or shown either below or at the side of the main section. Up to six faders can be shown, and they are freely reassignable to any adjustable parameter from any of the pages, by means of a drop-down menu above each fader. This mapping is saved with user patches. In the central area of the window you can select algorithm types or the presets associated with them, and you'll also find the usual save, load, delete and A/B compare buttons. Separate faders set the wet and dry signal levels, though the default when a preset is loaded seems to be 100 percent wet, which assumes you'll be using the reverb in a send. This can be changed in the preferences, along with other user attributes.

As with the reverb units it seeks to emulate, the reverb decays don't follow a simple exponential curve but are somewhat more complex, some of them building up more slowly before they decay. One of the reverb parameters controls the apparent size of the space being emulated, while another adjusts the shape of the room and determines how the early part of the reverb builds up.

The reverbs do have a 'classic' sound to them, but because of the restricted amount of CPU overhead allowed them, they don't have the same finesse as the more costly hardware units, and the synthesized reverb doesn't blend with the dry sound in such a natural way. With a Lexicon reverb, for example, there's a lot of early-reflection information that adds a sense of place to the dry sound, so the more you turn up the reverb level, the more the sound seems to recede back into the room. That doesn't really happen here to the same extent, though the treatments on offer can still sound extremely good in the context of a mix. You get the sonic 'glitter' that a good synthetic reverb provides, especially if you reduce the diffusion for a gritty, vintage sound, and you can add a fair amount of reverb without it getting in the way of the dry sound, which is always a good test. There's plenty of adjustability, and though you can't create really convincing small room ambiences, the range you do get covers most music-production territory including, some very believable plates and great-sounding rooms.

The claim to challenge high-end algorithmic reverbs may be stretching things a bit, but Breverb is still sonically impressive, especially when you consider its very small CPU footprint. It sounded noticeably more natural than Logic's Platinumverb and subjectively was at least on a par with IK Multimedia's CSR plug-in. If the reverb that comes with your DAW is a bit on the disappointing side but you don't want to throw away all your CPU overhead, Breverb is definitely worth checking out, and thanks to the free demo period, you can do this with no risk.

Paul White

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TC Tube-Tech CL1B

Formats: Mac & PC Powercore & TDM

TC Electronic's Tube-Tech CL1B plug-in, which is available both for TC's Powercore platform and as a TDM plug-in for Pro Tools HD and HD Accel systems, has been created using a digital modelling system that works by analysing the behaviour of individual components, to recreate the true analogue sound of Tube-Tech's rather lovely all-tube CL1B compressor/limiter. With a GUI that follows the hallmark blue front panel of the real thing very closely, the CL1B plug-in has a compression ratio continuously variable from 2:1 to 10:1.

One of this compressor's most distinctive features is the Attack/Release Select switch, which offers three options. Fixed disables the manual Attack and Release controls and sets an attack time of 1ms coupled with a 50ms release time, while the Fix/Man option combines the release times of the Fixed and Manual modes in such a way as to implement a shorter release time for brief peaks and a longer release time for longer peaks. This mode is designed primarily for compressing mixes or submixes. The third option is the Manual mode.

A three-position Sidechain/Link switch becomes active when the plug-in is inserted on a stereo channel, allowing you to set whether the gain reduction follows the signal in the left channel, the right channel or a combination of both. The VU meter can show the input level, output level or amount of gain reduction, in either channel or both. The In switch on the left of the panel activates or deactivates the processing, as you'd expect, but leaves the meter working with the plug-in still loaded onto the Powercore DSP, so that it can be engaged without delay; by contrast, the rotary Off/On switch frees up the Powercore DSP when the plug-in is off. Gain compensates for the gain loss during compression and, despite its physical position on the panel, comes after the gain-reduction section, with up to 30dB of available make-up gain.

You can run up to two instances of the plug-in (in stereo, 44.1kHz) per DSP chip on second-generation Powercore hardware (Compact, Firewire, Express, PCI MkII or Unplugged). The plug-in count is halved at 96kHz sample rates.

Like the original, the plug-in seems to have the knack of making things sound better just by passing them through the compressor, even when it is doing very little in the way of processing. Furthermore, when

you do add compression, you have to go a long way before it becomes too obvious. Most of the time, the mix simply sounds louder and more confident, with the individual elements blending together better. The Fix/Man mode works best on mixes where fixed attack and release times may not be ideal, due to varying dynamics in different sections of the song. I tend to prefer the low-threshold, low-ratio approach to mix compression, but even with more assertive settings, the end result almost always sounds well-controlled and musical. Vocals really come to life at medium-ratio settings, and you can afford to apply more gain reduction than you can with many competing designs or other plug-ins before the side-effects become noticeable. Similarly, you can really beef up the sound of a kick drum or complete kit by combining a longer attack time with a shortish release. I also found that I could afford to be quite heavy-handed with the compressor when it came to treating bass parts, and when the side-effects finally do become audible, they are still quite musical. With most compressors I find myself holding off a bit in case I damage the sound I'm trying to process, but this one just seems to egg you on to try adding more.

There's no denying that as plug-ins go, this one is quite expensive, and for a little over four times as much you could own the real thing. Nevertheless, the real thing won't allow you to use multiple or even stereo instances without spending more cash, and though there will always be heated discussions as to how accurate any form of modelling can be, the CL1B plug-in sounds gorgeous and is very forgiving even when you lay it on with a trowel! It's probably fair to say that when it comes to emulating high-end analogue gear as plug-ins, Universal Audio have always carried the torch, but now it seems TC have demonstrated their ability to be just as meticulous. *Paul White*

£ 560 Euros including VAT.

W www.tcelectronic.com

Fatar Numa

MIDI Controller Keyboard

Long-term players in the controller keyboard game, Fatar have brought all of their expertise to bear on the Numa — so is it as sophisticated as it is stylish?

Nick Magnus

Growing demand for hands-on control of DAWs, software instruments and effects has meant that today's master keyboards, from the full-scale, weighted, 88-note, feature-laden types down to the nattiest two-octave, battery operated varieties, come bristling with knobs, faders and buttons. Fatar have been making master keyboards bearing the Studiologic name for ages, their most recent models being the control-laden VMK range, reviewed in the February 2007 issue of *SOS*. It is, therefore, interesting that their latest 88-note weighted Numa keyboard takes an altogether different approach, by foregoing knobs or faders and focusing on the performance and quality of the keyboard itself. Bold claims are made for this particular keyboard, making the Numa potentially of interest to those who up to now have not been convinced by the feel of weighted MIDI keyboards.

Designer Keys

The Numa's design and construction are nothing if not bold statements. The chassis is constructed entirely from ABS, which brings the benefit of reduced weight compared to other 88-note boards. This keyboard is clearly conscious of its looks — if it were alive it would spend hours

SOUND ON SOUND

Fatar Numa £899

pros

- High build quality of the keyboard mechanism.
- 'You Play' system provides customisable velocity response to suit your playing dynamics.
- Sleek, arty design.

cons

- Coarse resolution and discontinuities of controller wheel and aftertouch data.
- Awkward positioning of the wheel.
- Only one wheel with no centre detent.
- Touch-sensitive controls behave erratically.
- Pricey, considering the basic nature of its facilities.

summary

The Numa provides a superior playing experience when coupled with a good piano sound source. However, various operational aspects of the instrument seem subservient to the physical design, and there are a number of software issues that need addressing. The lack of a detailed control surface may limit the Numa's appeal for computer musicians, unless they are happy to use it in conjunction with a separate MIDI controller device.

peering into a mirror looking for blemishes. Despite being made of plastic, the whole assembly feels reassuringly solid — nevertheless, I wouldn't fancy the Numa's chances of avoiding a nasty crack if dropped onto its end. For that reason, gigging musicians would certainly need a sturdy flightcase to maintain its appearance, which naturally puts the weight up again. The Numa's design could be described as elegant minimalism — it wouldn't look out of place in a modern art installation (more Duchamp than Tracey Emin) or featured in a lifestyle advertisement in the corner of some expensive open-plan loft apartment on London's Bankside.

To the left of its otherwise featureless, glossy white surface is a small, shiny, black control panel that houses the Numa's user interface. The only other interruption to this sleek minimalism is the brushed aluminium upper-rear surface, which curves round to form the rear panel. This can be slid backwards to reveal a slot for the included perspex music stand. Most usefully, though, it increases the depth of the top surface to form a convenient platform for other devices — a second keyboard, a drum machine or, perhaps more essentially, a knobby USB MIDI control surface.

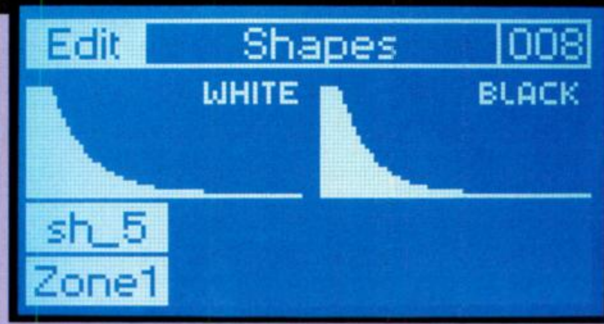
Connections & Editing

Out of necessity (due to the sliding aluminium upper-rear surface) the Numa's connections are located on the left-hand end cheek instead of the rear-panel, as you might expect. These are quite basic, consisting of a single MIDI output, two assignable controller pedal jacks, a USB connection and a DC power input jack. Also found in this rather unorthodox location is an unsprung controller wheel — more on which later.

You Play

Fatar's much-vaunted 'You Play' system, which allows the user to create their own custom velocity curves, works very effectively. In You Play mode, the Numa analyses your performance for 30 seconds and then creates a velocity response curve that complements your playing dynamics. Fifteen such curves can be saved, enabling the Numa to be customised for a number of different players or situations. Fatar claim this is a unique system. However, Kawai actually got there first with a similar system on their MP8 Stage Piano (reviewed *SOS* November 2005).

Where You Play *is* unique is that it generates separate curves for the black and white keys, recognising that players tend to apply different degrees of force to each of these. The result is noticeably smoother and more consistent dynamic control, with fewer of those rogue 'stick-out' or 'lost' notes that often plague other weighted keyboard designs.



'You Play' generates independent velocity curves for black and white keys.

The Numa's control panel is finished in a shiny black surface that Fatar call 'Gloss Metacrilato'. Stylish and elegant, this panel consists simply of a blue backlit 128 x 64 pixel LCD, below which are four buttons flanked by two circular 'dials', the sum of which serves as the Numa's user interface. Actually, these are neither buttons nor dials in the usual sense — they are capacitive touch-sensitive pads, a rather sci-fi approach for a distinctly sci-fi looking keyboard. The left hand dial is divided into four segments that function as up/down, left/right cursor buttons. The right-hand dial functions like a standard endless rotary encoder — just slither your finger round the ring to change values. In the default Play mode, the up/down cursor buttons are used to select Patches. The first five of these are 'templates', and permanently resident in flash memory, with up to 59 more locations available for user Patches, storable to internal EEPROM memory. The left/right buttons step through five 'quick edit' pages from where temporary changes can be made to each of the Numa's four key Zones. These include MIDI channel, Zone mute, Transpose, Program Change and Volume. However, these changes are lost if the Patch number is changed. To make more detailed ▶

The Numa has a particularly simple interface.

changes that can be saved permanently, you must enter the 'proper' Edit mode by touching the left and right cursor buttons simultaneously.

Edit mode offers more detailed parameter settings, including those covered by 'quick edit', across 14 pages. Setting the four Zone key ranges is easy — select a Zone, play the lowest note followed by the highest, and it's done. Three key velocity modes are selectable per Zone: Dynamic, Organ and Staccato. Both Dynamic and Staccato modes make use of the currently selected velocity curve (soft, medium, hard or one of 15 user-definable curves — see the 'You Play' box). Organ, on the other hand, delivers a fixed velocity of 127, but with a useful bonus. Rather than having to depress a key for nearly the full depth of its travel before a note triggers (around 8mm) in Organ mode you trigger notes sooner, at around 5mm of key travel, making organ playing techniques easier to execute. This might also prove invaluable for playing other non-pianistic synth sounds; however, Organ mode does not output any velocity other than 127, making it of limited use in this respect.

The difference between Dynamic and Staccato modes is, put simply, the point at which note-on and note-off events occur relative to a key's physical travel. In Dynamic mode, note-ons happen near the bottom, whilst note-offs are near the top, analogous to the hammer/damper motions of a real piano. In Staccato mode, note ons and offs are at the same point, at the bottom of the key's travel, enabling much shorter notes to be performed than in Dynamic mode. It's a subtle difference, but of potential benefit when using Staccato mode to play synth sounds.

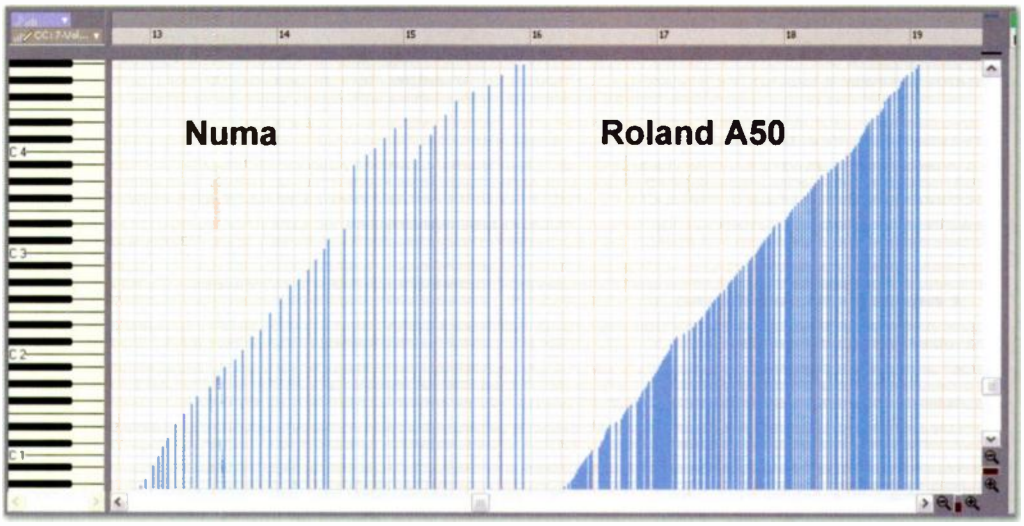
The Keyboard

The Grand Touch TP400 keyboard mechanism has a very high-quality feel, partly due to the use of 'full body' solid black keys, as opposed to the hollow ones found on many instruments. The keyboard also features taller white keys, for a 'throw' distance that Fatar describe as mimicking a full concert grand piano. The action is graded (heavier at the bottom, lighter at the top) and it certainly feels well-balanced, although I still found it generally heavier overall than most real pianos — the lighter weight of the top keys is arguably closer in feel to the real thing.

Fatar's claim that this is "the most inspiring, pianistic action ever conceived in an electronic instrument" is a bold one; nevertheless it does provide a pleasurable playing experience, and certainly seems to offer more consistent control over dynamics than many other weighted 'boards.

Control

The wheel, aftertouch and the two controller jack sockets all operate in either positive or negative polarity, and are freely assignable to any controller number from zero to 127.



The control-data curve on the left shows the coarse resolution and discontinuities generated by Numa's wheel, compared to the same data generated by the mod wheel of a Roland A50, on the right.

They can also be individually disabled per Zone. Aftertouch can also (unsurprisingly) generate aftertouch data, while the foot jacks can be defined as continuous, open or closed switch controller types.

The wheel can additionally be assigned to output pitch-bend data. However, because the values output by the wheel run from zero to 127, 'pitch zero' (value 64) is at the centre of its travel. Since the wheel has no centre detent position, locating 'pitch zero' quickly and accurately is going to be difficult, making it impracticable as a performance tool in this context. The wheel's resolution is also very coarse, with drastic data discontinuities that can be seen in the screenshot above. Aftertouch also suffers from overly coarse resolution and similar discontinuities, failing to generate virtually any data at values less than 42.

USB Functions & Support Software

The Numa offers two modes of USB operation from its system menu: USB MIDI and USB Virtual Com. The instrument starts by default in USB MIDI mode, and connecting the Numa to my PC via USB produced the expected result — the Numa was recognised as a 'USB Audio Device' and became available as an additional MIDI port in Sonar. USB Virtual Com mode enables the Numa to communicate with PC editing applications, and although none were available at the time of writing, 'Numa Interactive' software should be downloadable from Fatar's web site by the time this review goes to press. This will allow updating of the Numa's operating system (as and when updates are made available) and will also store Patches to a PC. Additional editing software is scheduled for the coming months, which, according to Fatar, "emulates the current Numa LCD display and patch editor functions for computer use."

Alternatives

There is a number of 88-note controller keyboards on the market at the moment, some with built-in control surfaces and some without, and at a range of different prices. Fatar's own VMK188 plus (fully weighted, full control surface), is an obvious alternative — and was reviewed in the April 2005 issue of *SOS*.

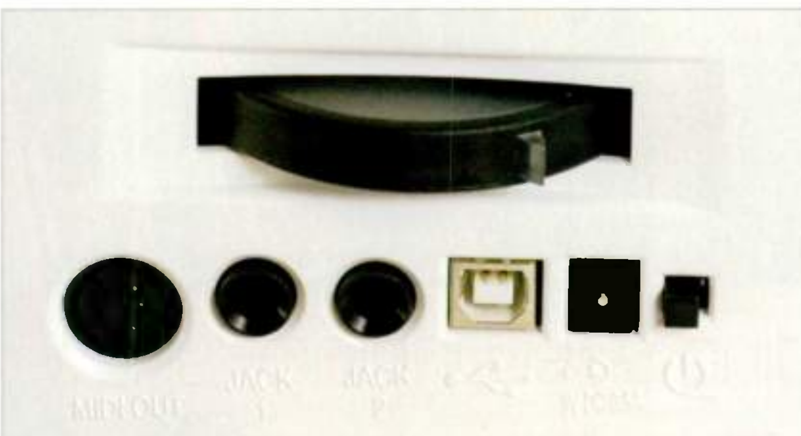
M-Audio's Keystation, also featuring a fully weighted keyboard and a control surface, would also be worth considering. Lastly, the CME VX8 and UF80 both offer weighted keyboards and control surfaces and both have recently been reviewed in *SOS*.

The wheel itself is an enigma; its physical location makes it uncomfortable to use, as you have to reach almost to the rear of the end panel to find it. It is also completely invisible from any practical playing position, which causes one to wonder why Fatar went to the trouble of making it translucent and embedding it with attractive blue LEDs that glow brighter as you increase its value. No-one, least of all the performer, is ever going to see it!

Using the footpedal jacks with a continuous pedal also revealed a shortcoming: the maximum value generated was only 123. The same pedal I used for testing (a Roland EV5) is capable of generating the maximum value of 127 when used with other devices, so the fault clearly lies within the Numa.

Further software issues include a Panic button function that succeeds in resetting the controllers but fails to send any note-off information, and Program Changes which, when stored in a user Patch, fail to be transmitted when the Patch is selected. Many of the above problems are presumably due to this being the initial release version of the Numa's operating system (version 1.0), so one can only urge Fatar to resolve these issues at the earliest opportunity.

Finally, although one can appreciate the intent behind the hi-tech touch-sensitive controls, they proved to be more of a hindrance than a help. Occasionally they would fail to react at all on the first touch. More often, however, they were hugely over-sensitive, tending to react when my finger merely hovered over them. This resulted in erratic behaviour such as patches changing unintentionally, pages cycling as my finger passed over one button on the way to another, and edits being lost. Although options are available in the System menu to alter sensitivity, the problems are still exhibited even at the lowest settings. Nice idea, Fatar, but in this case not the wisest choice.



All of the Numa's connections are located on the left-hand end cheek of the unit — along with the single, assignable control wheel.

Conclusion

The aesthetic side of me likes the Numa purely for its appearance, but the all-important practical side recognises various causes for concern. Software issues can be fixed, but hardware ones (such as the location of the wheel) are less likely to be. Anyone needing more detailed control will also have to take account of the additional cost of a MIDI control surface, which could add up to a fairly expensive package. However, if all you need is the ability to play piano sounds on a luxurious keyboard, then the Numa could be exactly what you're looking for. SOS

information

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Equipped Music Premier Beats

Multi-format

Equipped Music just go from strength to strength with their series of trip/hip-hop loop libraries. This latest hip-hop release, for example, kept bringing to mind classic Gangsta records like Snoop Dogg's *Doggystyle*, Dr Dre's *The Chronic*, and Tupac's *The Don Killuminati*, as well as Notorious BIG's *Ready To Die* — which can only be a good thing, though whether it's safe to mention all those guys in the same sentence, I'm not quite so sure...

Spanning a tempo range of 85-95bpm, the drum loops exhibit a deep and easy swing reminiscent of Tupac's 'Live And Die In LA' or Ice-T's 'It Was A Good Day'. Kicks throb powerfully, as if through a low-rider's bodywork, while snares combine rich sustain with plenty of punch. Hats take a more background role, merging with the general vinyl patina to subconsciously support the general groove. What always impresses me most about the drums on Equipped releases, though, is their musicality — where most other developers just seem to be programming patterns, Equipped have found the magic ingredient that makes beats sound alive and dangerous. Certainly the drums on offer here are more than a match for Smokers Relight (which I praised back in *SOS* August 2007), while providing a more saturated and gritty sound that provides a useful contrast. If anything, the harder-edged feel here appealed to me personally even more. As usual, every effort seems to have been made to increase the usability of the core beat loops, from supplying various simplified versions to including a capacious one-shot section on DVD two. If this isn't enough, of course, the REX 2 files expand the possibilities even further, providing complete multitrack control over the mix.

Moving on from the drums, each beat has an accompanying mixed music loop replete with the kind of sultry added-note electric pianos, vinyl hits and crusty tempo-delays for which Equipped have such a reputation — imagine a kind of scummy mash-up of Marvin Gaye's 'What's Going On'. However, the sounds are all shot through with a consistent sweltering languor, which again contrasts pleasingly with the somehow cooler-sounding nocturnal material on Smokers Relight. There was the odd occasion when I felt that maybe the company were resting a little on their Smokers-based laurels here, but there's no getting away from the fact that this is still all really evocative material, with a disjointed, old-school feel that gels beautifully with the beats.

The music loops are also available in deconstructed one-shot form, and the electric piano and bass hits in particular are brimful of Roni Size-worthy character and grime: pop half a dozen of those and a couple of the beats into an MPC and you're laughing. As an added bonus, there's also the best collection of vinyl-noise samples I've yet come across tucked away in there, so you can dirty up any other sample library to blend in seamlessly with a Premier Beats-based arrangement.

Despite the large amount of raw material included on the two DVDs (over 2GB of loops and a further 1.2GB of one-shots, with duplication across Apple Loops, Refill, REX 2 and WAV formats), the quality is maintained at such a consistently high level that it feels like there's no end to it. In short, I can recommend this library without reservation to anyone with a penchant for powerful G-Funk swagger. Go listen to the demos, but I'm warning you now — once you've finished weeping, you'll have little choice but to flex that plastic. *Mike Senior*

£89.95 including VAT.
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www.timespace.com www.equippedmusic.com

Soniccouture Hang Drum

Kontakt 2

This is a rarity: a sample library dedicated to a handmade instrument unknown to 99.5 percent of the musical community. The Hang (from the Swiss-German dialect word meaning 'hand') is an acoustic tuned-percussion instrument built along the lines of a Caribbean steel drum. Made from two domed steel shells joined at the rim, it looks like a flying saucer from the side, and resembles a kettle-gong from above. Distributed around the upper playing surface is a series of dimpled indentations, which the player strikes by hand to produce notes. The sound is softer, more melodious and sustaining than a steel drum and lacks the Caribbean pan's stridency.

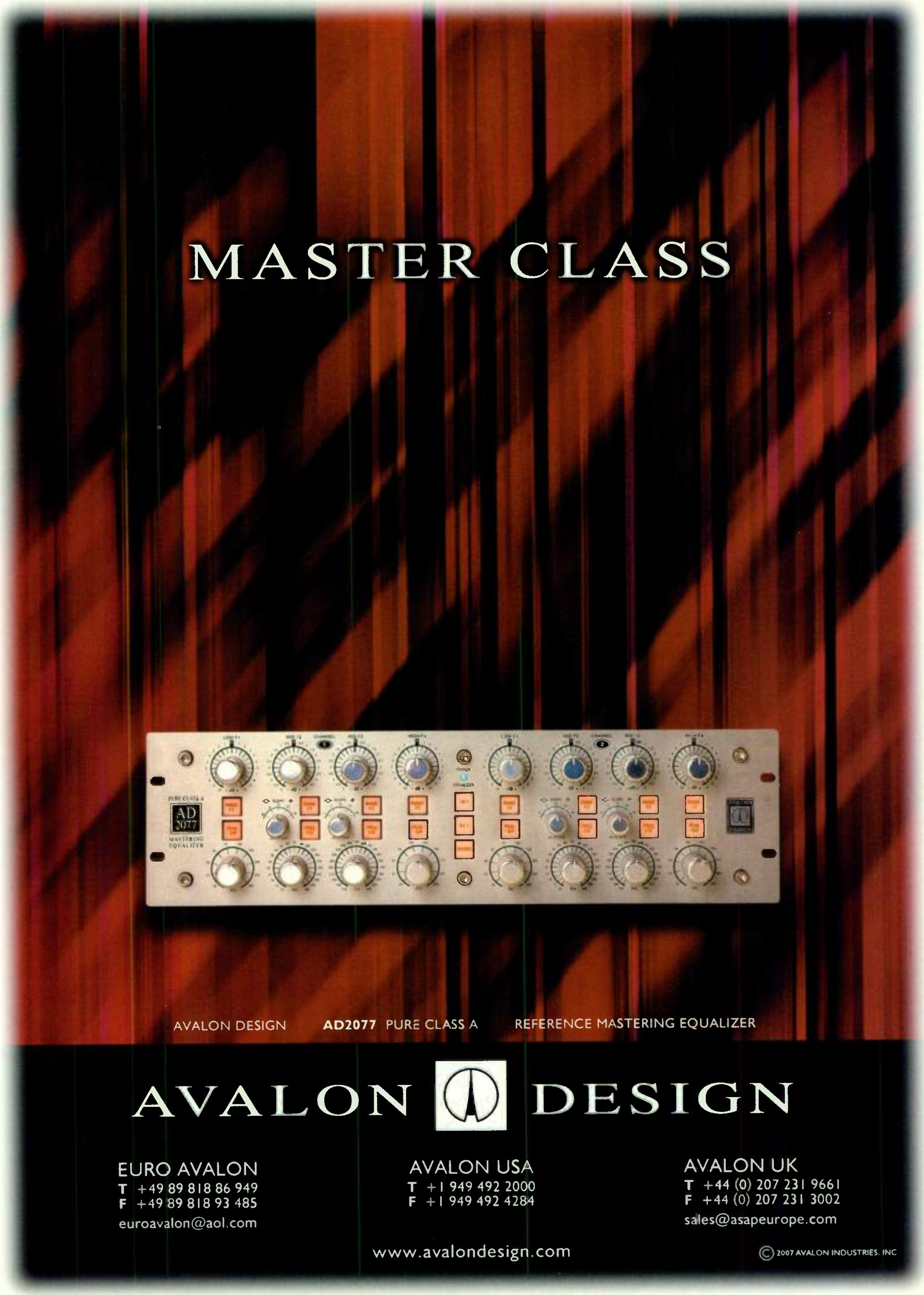
Hang drums (as they're commonly known) are generally tuned to concert pitch and play a diatonic scale of seven or eight notes that varies from instrument to instrument. There's a waiting list for orders and according to its inventors, "further collaboration between art and science is needed to make it possible that other hangmakers may exist in future". I guess that means it won't be available in Woolworths just yet.

In order to bring the sound of this cultish 'sounding sculpture' to a wider public, the enterprising UK-based company Soniccouture sampled two Hangs: the first plays the notes F3, A3, B♭3, C4, E4, F4, A4, B♭4 and C5 and the second (a later model) operates in the minor scale of D3, A3, B♭3, C3, D4, E4, F4 and A4. Both have an extra, rather impure and clay-drum-like fundamental tone, pitched an octave below the bottom melody note. The impact of flesh on metal gives a soft, organic yet distinct attack and the ringing notes are clearly pitched, open-sounding and easy on the ear. As with the Javanese bonang, each note's fundamental frequency is augmented by a haze of untempered upper harmonics, creating an exotic, ethnic-sounding timbre.

The Hangs were sampled at many different dynamics using palm slaps, knuckles and fingers hits, resulting in 2.47GB of data. The finger strikes were performed in three zones round each of the dimpled 'tone fields', the central zone giving the plummiest tone. There's a choice of keyboard mappings, including one that arranges the samples chromatically across nearly three octaves. Alternatively, you can limit yourself to the Hang's original pitches and use a handy Kontakt control to transpose them into any key. A hugely entertaining 'Hang Jammer' script generates endlessly-repeating notes at a user-selected rhythmic value, with total control over the tempo, degree of pitch variation, randomisation and octave range — a great way of creating unpredictable, slightly deranged, gamelan-like grooves.

Despite my views on capital punishment, this is one Hang I'm happy to see brought back. There's nothing quite like it on the market, the producers have done a diligent and exhaustive sampling job and it's sensibly priced, so if you're on the lookout for new sounds I suggest you buy a copy before it gets plastered over umpteen movie soundtracks. *Dave Stewart*

DVD-ROM £61, download £58. Prices include VAT.
www.soniccouture.com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Playback
Readers' Music Reviewed
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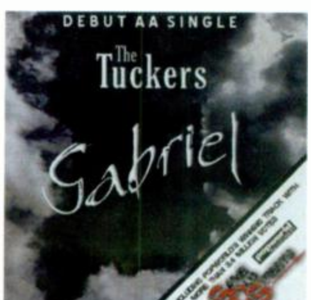
aZaidou
Butterflies

There are many reasons why you don't see a lot of rock bands with singing pianists. For one thing, the stage piano has yet to be invented that can be carried on public transport. For another, the range of rock & roll shapes that can be thrown with two hands on the keyboard and one foot on the sustain pedal is limited. However, I suspect the biggest factor is just that singing and playing the piano at the same time is bloody difficult. Or is that me?

Whatever the reason, it's refreshing to hear a band like aZaidou, who use the instrument as a central element of their sound, but don't restrict themselves to down-tempo ballads or allow jazz and blues influences to dominate. If the three tracks included here are anything to go by, they should be a fine live band: the musicianship is excellent, the songs are very well constructed, and there's an oomph to the whole thing that you don't often find in keyboard-led bands.

However, aZaidou's demo CD also shows up one of the other problems that can arise when you try to put the piano centre stage in a rock band: it's just too big, and not only in the physical sense. The sound of a piano spans an enormous range of frequencies, many of which it shares with drum kits, basses and guitars. It takes both sympathetic arrangement and serious engineering and mixing skills to resolve these battles, and whoever recorded this CD hasn't quite succeeded. Drums and bass come across well, but the vocals are distant, the DI'd acoustic guitar is scratchy, its electric counterpart is muddy, and the electronic piano sound is a thin and apologetic tinkle. Ultimately, though, it's the music that matters, and none of this really stops aZaidou's strengths coming through. A band to watch. *Sam Inglis*

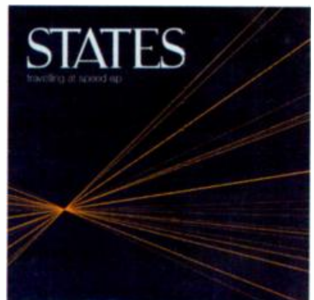
W www.myspace.com/azaidou



The Tuckers
demo

In the spoof documentary series *Yacht Rock*, Donald Fagen of Steely Dan speaks in a private language that only Walter Becker can understand. The Tuckers' biog makes me wonder if something similar is going on in their universe. What, exactly, do they mean by describing themselves as "a self-acclaimed 'H-indie' band"? What madness made them adopt as their logo a badly drawn naked female body, on which the band name appears to be carved in blood? Since they've sent in a blank CD, I guess I'll never know. *Sam Inglis*

W www.thetuckersonline.co.uk



States
Travelling At Speed EP

Did someone change the meaning of the word 'eclectic' while I wasn't looking? I must've missed the meeting where Croydon band States chose it to describe their 'mix of influences', which covers Coldplay, Radiohead, U2, Oasis, the Verve, the Manics and Jeff Buckley.

I realise that Croydon may not be a melting pot of the world's cultures, but to my mind, that's about as 'eclectic' as the ethnic makeup of the Ku Klux Klan. Not that there's necessarily anything wrong with liking those artists, but you don't go making a song and dance about a record collection you share with every other binman indie band on the planet.

States' four-track EP is perfectly competent, then, but sounds exactly like you'd expect a band who only listen to mainstream indie-rock to sound. Guitarist owns a delay pedal? Check. Anthemic choruses? Present. Glib, empty lyrics full of metaphors about drowning? Here, miss. Careworn male vocals, with occasional notes sung falsetto for no obvious reason? Check, check and double-check. States are, not to put too fine a point on it, predictable, and their music would surely benefit from an injection of actual eclecticism. Look it up, guys, it's in the dictionary. *Sam Inglis*

W www.statesmusic.com

Alex Sheppard
demo

Here are some facts about Alex Sheppard: she is a 20-year-old singer-songwriter from Norwich, who's been compared to the incorrigible but downright cuddly Kate Nash. Already she has appeared on BBC local and national radio stations, caught the attention of the *NME*, and has supported ex-Libertines frontman Carl Barat and his current band, Dirty Pretty Things. Not bad going, really.

But let's go back to this comparison with the aforementioned Nash. For those of you unaware of her exceedingly annoying yet utterly catchy talent (I'm not sure her cock-er-ney tones have crossed the north Atlantic yet), she's popular with the young population of the UK, mainly because most 15-year-old girls can 'sing' just as well as Nash, or at least do a good impression of her. They can even play the stupidly simple piano riffs that seem to pass for the basis of many of her songs.

Given these points, I can only surmise that the reason that Nash is so popular with the record execs (who are mostly fully grown adults) is that her songs are very well written and were seemingly unique on first hearing. Somehow, Sheppard's songs don't seem to keep me interested like Nash's do. Maybe it's because I've heard them before (on Kate Nash's debut album, I reckon) or perhaps it's because they need a good polish to bring them up to the grade.

I can see a lot of potential; Alex has a pleasant tone to her voice and can pitch well when she wants to, but her songs either need fleshing out in terms of arrangement, or cutting almost in half time-wise. Whichever approach she takes, I'm sure some discerning adult will be able to sell her music to the kids. *Chris Mayes-Wright*

W www.myspace.com/alexsheppard

Ahoora

All In Blood With You

The freedom to play progressive thrash metal is, arguably, not the most fundamental of human rights, but it's one of many that is denied to Iranian citizens. It's thus pretty impressive that Ahoora have managed to record anything at all, and although their EP *All In Blood With You* is not stunningly original, it sounds remarkably contemporary given the band's enforced isolation from the rest of the metal world.

No recording or equipment details are available, but the band's technical ability puts to shame many of the demos I've heard from more liberal regimes. They play with an assured power and precision, led to good effect by what is either an excellent but uncredited drummer or a very nicely programmed machine.

Any form of criticism seems redundant in the circumstances; there is some muddiness to the proceedings, which could be due to the necessity to distribute the EP as an MP3 download, but I suspect that limited choices of amps and microphones might also play a part. There's also a tendency for the vocals to get lost in the mix, but it's understandable that singer Ashkan Hadavand doesn't always sound confident. I dare say few of us would, if we played under the constant threat of being thrown in prison. *Sam Inglis*

W www.ahoora-band.com

Hinterland

demo

Hinterland are what you might describe as an old-fashioned guitar band, who seem to be targeting their appeal at people who find Queens Of The Stone Age a bit too modern and sophisticated. There's a satisfyingly brainless, Route One quality to their songs, a feral rawness to their sound and an impressively bone-headed dedication to the way they play simple riffs over and over and over again. I don't often find myself agreeing with a URL, but in this case, I'm happy to concur that Hinterland do, in fact, rock. Dot com. *Sam Inglis*

W www.hinterlandrock.com

Jon Chapman

demo

Oddly enough, it's not often you hear a demo that hints at unfilfilled potential. Usually, if someone's capable of writing a decent song, they'll also have the wit to create interesting arrangements and the wherewithal to record things competently.

Jon Chapman might be an exception. He's a good singer and a mature songwriter who obviously has a clear idea of the music he wants to make, and there are moments of brilliance on his demo. His imaginative vocal harmonies, for instance, are as well-executed as they are unexpected, and the second track, 'Lose It All', boasts a lovely lazy groove overlaid by jazz guitar and shaky eggs. Yet at the same time, Jon's music has a strangely unfinished feel to it. The songs would benefit from editing, the playing is loose to the point of sloppiness, the kick drum sounds like it's been high-pass filtered at 400Hz, and the strong vocals are undermined by weird reverb choices. What's needed here is a producer. Any volunteers? *Sam Inglis*

W www.jonchapman.co.uk

Chalou Saint Jude

demo

A bad start for Chalou Saint Jude: a riff poached from Eminem doesn't sit well in the opening track, 'One Of My Personal Favourites', while elsewhere, the influences of the Rolling Stones and Kings of Leon strangle the band's own songwriting and musicianship. Track two, 'End Of The Road', could indeed be mistaken for the brothers Leon and, happily, shares their great feeling and real energy. Occasional timing issues, maybe due to sync problems, might detract from this recording for a rhythmic OCD sufferer, but weren't a critical point for me.

The spell of the early Rolling Stones makes its presence felt on the third song, a slower, Sunday afternoon blues that is decorated with interesting wordsmithery and performed with an effortless roll. Even though the vocal suddenly turns Irish, it's an elegant track that actually works very well! *Chris Sibbons*

W www.chalousaintjude.com

Jazz Thrash Assassin

Eclecticore

Listening to this four-track EP provoked me to try to list all the bands that Jazz Thrash Assassin sound like. Their songs constantly switch between styles as diverse as thrash metal, disco, jazz fusion, ska-punk and cabaret, so I came up with a very long list. However, one very important name was missing: Jazz Thrash Assassin are so busy mimicking Napalm Death or Isaac Hayes that they forget to ever sound like Jazz Thrash Assassin.

The band's motorway pile-up of styles certainly does justice to the title, but for all its cleverness, it's deeply unsatisfying to listen to. *Eclecticore* is the musical equivalent of having your TV remote grabbed by a compulsive channel-surfer. There is no whole, just an awful lot of parts, and no sense that the band has any real investment in any of the genres on offer. This is the sound of people trying too hard. *Sam Inglis*

W www.myspace.com/jazzthrashass

Ear Candy

Filtered 'drops' are easy to set up and ever-popular. Here, Multimode Filter's Frequency parameter is being automated.

DP Audio Production Tricks

Robin Bigwood

DP offers an excellent environment for working with the little pieces of 'ear candy', such as filtered drops, beat-mangled breaks and reversed audio phrases, that can make such an impact in pop and electronica production. We look at some of the techniques involved.

There are many ways to create the pleasing little isolated and infrequent audio treatments that can keep a track sounding fresh and interesting for the listener. In fact, the more you experiment, the more off-the-wall techniques you can come up with. But several basic approaches stand out as being both effective and easy enough to work with: using plug-ins in real time; applying plug-ins off-line; and combining specific MIDI and audio techniques. All of them require that your audio is being handled internally by DP, so any external sound sources will either already have been recorded into audio tracks, or will be coming into DP via an audio or aux track, prior to being recorded.

Real-time Treatments

One straightforward approach is to use audio plug-ins on individual tracks, aux track submixes, or indeed a master fader, and to automate their bypass status (and maybe their parameters too) to make them active for a short period of time. A common example is the 'thin-sounding intro' — using EQ, you roll off bass and treble at the beginning of your song, but bypass the EQ when the first verse or chorus begins, to deliver extra impact. Here's how you'd do it (see top screens, opposite page).

1. Instantiate an EQ (perhaps a Masterworks EQ) in an insert slot on a master fader.
2. Set up the EQ so that it has high-pass and low-pass bands enabled, and roll off your frequency extremes to taste, while listening to the intro of your song.
3. In the Sequence editor, insert an automation event to enable the plug-in at the start of the sequence. Do this by clicking on the Track Settings button next to the master fader track's name, choosing Insert, then the name of your EQ, and then Bypass. This 'loads' the pencil tool with a bypass event, and you choose whether to write a 'bypassed' or 'enabled' event by clicking in the upper or lower part of the track lane. DP informs you which in the Track Info bar (DP4 and DP5) or the Cursor Info palette (DP6).
4. The resulting automation data is shown in its own track layer as a line with breakpoints. If it's dotted, automation is not enabled, and you'd need to switch it on for the track by clicking on the Track Settings pop-up once more, choosing Automation, and clicking Play.
5. Finally, we need to write the EQ bypass event, to restore the normal sound of the mix when the first verse or chorus is reached. Locate to the appropriate point in the track and place the mouse pointer on the automation line. It turns into a pointing hand cursor and a click now will write the new breakpoint. Drag this breakpoint upwards so that it becomes a 'bypassed' event, and you're there.

Playing the sequence should now result in the EQ plug-in automatically being bypassed and enabled in the appropriate places. You could, of course, go on to write more of the same kind of data, to bring back the bandwidth-limited effect later on in your mix. Also consider trying other plug-ins that suit being suddenly bypassed or enabled. Digital degraders such as MOTU's own Quan Jr or the freeware MDA Degrade are excellent for brief, end-of-measure drum breaks, for example, and another favourite is the 'knackered LP' effect that can be easily achieved with iZotope's Vinyl plug-in (also freeware, and in MAS format too). Short sections of very heavy reverb also work well — the sudden disappearance of the tail as you bypass the effect can be startlingly effective. Don't forget DP5's Pattern Gate either: by enabling most of its steps but setting the envelope to a fast decay, you can create some very effective 'stutter' effects.

Dynamic audio treatments can be set up by writing a few plug-in automation events in the Sequence editor.

For more subtle and 'evolving' treatments, it's better to automate plug-in parameters rather than just their bypass status. The classic (and rather well-worn) example is the closing and re-opening low-pass filter sweep so beloved of euphoric techno and its pop spin-offs. For this you use a similar approach to that described above, but automate a filter plug-in's cutoff parameter with an automation data *ramp*. Here's how you'd automate a low-pass Multimode Filter to close from 20,000Hz to 200Hz, before re-opening:

1. Enable Multimode Filter on a master fader track and make sure it's set to low-pass mode, with wet/dry mix at 100 percent wet, and resonance to suit. Disable its on-board modulation by setting the range parameter to zero.
2. In the Sequence editor, locate to where the filter sweep should begin. In the master fader's track lane, click the Track Settings pop-up, choose Insert / Multimode Filter / Center Freq and click in the track to write the automation event. Don't worry too much about the exact frequency value of the event you're writing, because...
3. Precise automation values (and locations) can be entered numerically. With the breakpoint still selected, click on its value as displayed in the Event Info bar (DP4 and DP5) or Event Information palette (DP6) and type in the starting value of 20,000 (Hz), then hit return.
4. Now locate to where the filter should be fully closed. Click on the automation data line to write a new breakpoint and, as you did in the last step, enter the precise value of 200Hz numerically.
5. At the point where the filter should be fully opened once more, click to create another new breakpoint. For this one, enter a value of 20,000 again. The breakpoints are connected by smooth ramps, causing the filter to close and re-open smoothly (see screen at the start of this article).

For greater flexibility when entering and editing automation graphically, you can draw upon DP's huge data-editing power, using the Reshape Flavor pop-up in the Tools palette to shape your automation line into all kinds of parabolas, cyclical waveforms, or even random madness. See the DP column from June 2005, on the *SOS* web site, for more suggestions.

Also, as I mentioned above, you don't always have to work with a master fader. You could just as easily apply automated plug-ins to individual tracks or instruments, or submixed instrument 'stems'. For example, you could try routing every single track in your mix *except* vocals to an aux track. Automating a filter sweep on the aux track would then isolate the vocals and leave them starkly audible even when everything else was murky and muted. For that matter, you could have automated plug-ins working on individual tracks, stem mixes *and* master faders all in the same sequence, to really play around with the sense of scale and coherence of the mix. The sky (or, at least, the grunt of your processor) is most definitely the limit.

Going Off-line

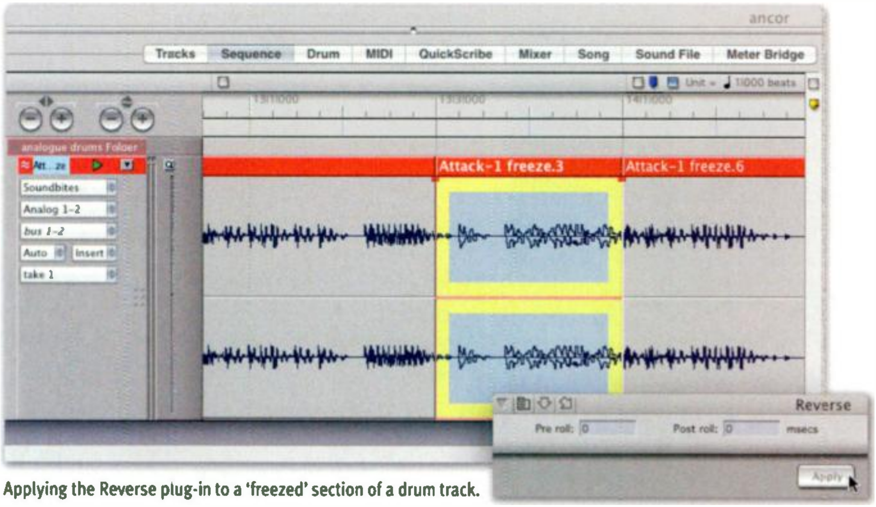
For some effects it's essential, or helpful, to *not* work in real time, and to apply plug-ins as an 'off-line' process to a soundbite or selection within an audio track. For example, this is the only way to work with true reversed (backwards) audio. It's also useful for experimentation with plug-ins that operate somewhat randomly; you can just repeat the process until you get something you like, and then you've got it for good.

The quintessential off-line process is reversed audio, which simply can't be achieved in any other way. It can be a particularly sweet little piece of ear candy, and is easy to do. In this example, the last few beats of a drum fill are reversed.

With the drum track 'freezed' or bounced on to an audio track, select a region of it that corresponds to the last couple of beats of a fill before a new section. You could do this by dragging over the region with the I-beam tool, or with the crosshair pointer that appears when your mouse pointer hovers over the bottom quarter of a track lane. From the Audio menu, choose Audio Plug-ins and then Reverse. Click Apply in the little window that appears. Easy as that! ▶

Soundbite Triggering With Nanosampler

You don't actually need MachFive or any other expensive add-on sampler to do the 'vinyl brake' effect described overleaf — DP5's bundled Nanosampler will do just as well. But for some bizarre reason Nanosampler can't handle split stereo files, so if you try to drag a stereo soundbite into its waveform window in DP5 you'll only get one of the channels loaded. Rather daft that DP's own sampler can't handle its native audio format, but there we are! This will probably be a thing of the past if you're using DP6 and working natively with its AIFF or BWAV (Broadcast WAV) interleaved audio formats, but for a workaround in DP5, select the soundbite you want to load into Nanosampler in the Soundbites window and export as an interleaved stereo file. Drag and drop this file *from the Finder* into Nanosampler. It will load in stereo, ready to use.



Applying the Reverse plug-in to a 'freezed' section of a drum track.

Obviously, the musical success of longer rhythmic and melodic phrases, when reversed, can be a bit uncertain unless they're carefully planned. So one way of retaining something of their original structure is to subdivide them into multiple soundbites, then reverse all those soundbites *en masse*. Taking the same drum fill from a moment ago, here's the alternative approach. First, select the scissors tool and manually cut around each visible beat. Or, if your audio has been beat-analysed, enable the beat grid (by selecting the blue tickbox at the top-right of the Sequence editor in DP5, or in the Snap Information palette in DP6) and simply drag over the soundbite with the scissors tool, to automatically make a series of precise, beat-based cuts.

Next, select all the resulting beat-long soundbites and apply the Reverse plug-in, as before. DP cleverly just reverses each individual soundbite, rather than reversing their order, and you get your drum fill as it was programmed, but with all the hits played backwards.

Another plug-in I love working with off-line is the Replicant beat mangler, by AudioDamage. It re-sequences audio according to various user-definable processes, but always with a degree of randomness. Using it off-line is a way of ensuring you produce something to your liking! As with Reverse, I just choose Replicant from my Audio Plug-Ins submenu, but then repeatedly apply it until I get an effect I really like.

Devious Means

If you're up for more of a challenge, you could try some effects that are achieved with a multi-step process involving both MIDI and audio. One is the 'vinyl brake' — as if a turntable's drive motor was turned off while it was playing your song. DP has no straightforward way to directly apply the drastic varispeed that is required. But it can be done in another way, using a sampler instrument such as MOTU's MachFive 2. You load the sampler with the section of audio you want to 'brake', trigger the sample via MIDI at the correct moment, and then use pitch-bend to drop the sample pitch:

1. Bounce your song to a new audio track, so that it's in your sequence as one long soundbite. Then instantiate your sampler on a new Instrument track, and create a MIDI track that drives it alongside.
2. Locate the point where you want to place the vinyl brake effect, and use the scissors tool to cut either side of it, to create a new soundbite.
3. Drag the soundbite into MachFive's keygroup editor, placing it with a root key of, say, C3. Set the MachFive part's Bend range to something suitably large, such as 36 semitones.
4. Now, in the MIDI track, use the pencil tool to write a new note event, with a pitch of C3, at *exactly* the location where the soundbite you created a moment ago begins. Make its duration as long as the soundbite, too.
5. Use the I-beam tool to drag a selection over the last two-thirds of the MIDI note, then from the Region menu choose Create Continuous Data.
6. In this dialogue box, select Pitch-bend, and enter values of 0 and -8192 (the minimum value) in the 'Change smoothly from' boxes. Set Minimum value and Minimum time changes to a value of 1, and finally hit Apply. This writes a lot of very smoothly graduated, downward pitch-bend events.
7. Finally, delete or mute the soundbite that you created in step 2. This is now replaced with its MIDI-triggered (and pitch-bent) counterpart. If the effect isn't quite right, try a different bend range, write the pitch-bend data later or earlier, or experiment with the curvature value in the Create Continuous Data dialogue.

Another combined MIDI/audio technique you might like to try is a little obscure, but subtly effective. It's another reversed audio effect, but it can give results that are smoother and not achievable in any other way. Here's a quick description of what it is and how it works. I'm applying it to a little piano break, but it sounds great on drums and almost any other MIDI source too.

First, make a precise selection of the MIDI events in the piano-track phrase to be treated. It can help if this begins and ends precisely on a beat. Now, from the Region menu, choose Retrograde. This reverses the

Respecting The Edit Grid

Many of the techniques described this month involve writing data or making selections in Sequence editor tracks. If it ever appears that you can't select or write where you want to, check to see whether the edit or beat grids are enabled. If they are, DP will only let you make selections to and from grid divisions, and you won't be able to write data between them either. But grids can also be your friends. When they're enabled, precise beat-based selections and placements are much easier to achieve, and dragging over a soundbite with the scissors tool, for example, will automatically make a series of beat-based cuts that could take a long time to do manually.

The edit-grid toggle buttons can be found in the top right-hand corner of the Sequence editor and Graphic editor in DP4 and DP5, and in DP6 they're located in the Snap Information palette. You can temporarily toggle their status by holding down the Apple key during editing actions.

Using the edit grids can really assist you with various selections and edits. Their toggle switches are shown here in DP5 (top) and DP6. The DP6 screen grab is from a pre-release version, though, so might be a little different in the final release.

order of the notes, so that they read the same backwards as they did forwards. Bounce or freeze this same region to a new audio track, and reverse the resulting soundbite. Finally, delete or mute the original MIDI events.

This process restores the correct order of events, but each individual event is heard backwards. It's a lot smoother and more reliable than bouncing first and then reversing every individual soundbite, and also works well for more sustained sounds. Sometimes you may need to tweak the soundbite's location to restore exact timing, but it's well worth the effort.

Freak Out!

As many studio people have observed in the past, some of the very best effects can come from wild experimentation with signal routing and processing, or even unintended accidents, as you tweak settings in real time. If you like to work in this way, it's crucial to make sure that you're recording your experiments, as those mind-blowing sounds generated by delays, freezable reverbs and granular synthesis plug-ins are all too often gone in a flash.

One straightforward approach is to use a completely separate application to record your DP noodlings. Rogue Amoeba's Audio Hijack Pro is great for this, and any gems that it captures can easily be re-imported into DP, though they will probably require some editing and then 'spotting' into position.

If you've got a recent MOTU audio interface, there's another option: in the CueMix Console application there's a File menu option called 'Mix1 Return Includes Computer Output'. With this selected, you can choose 'Mix 1 1-2' as an input to a stereo audio track in DP, to record both the external signals coming into your interface and the output of DP and other applications. There is a potential for feedback, so for safety keep the audio track that is recording the 'interface mix' muted while it does so.

Tools Of The Trade

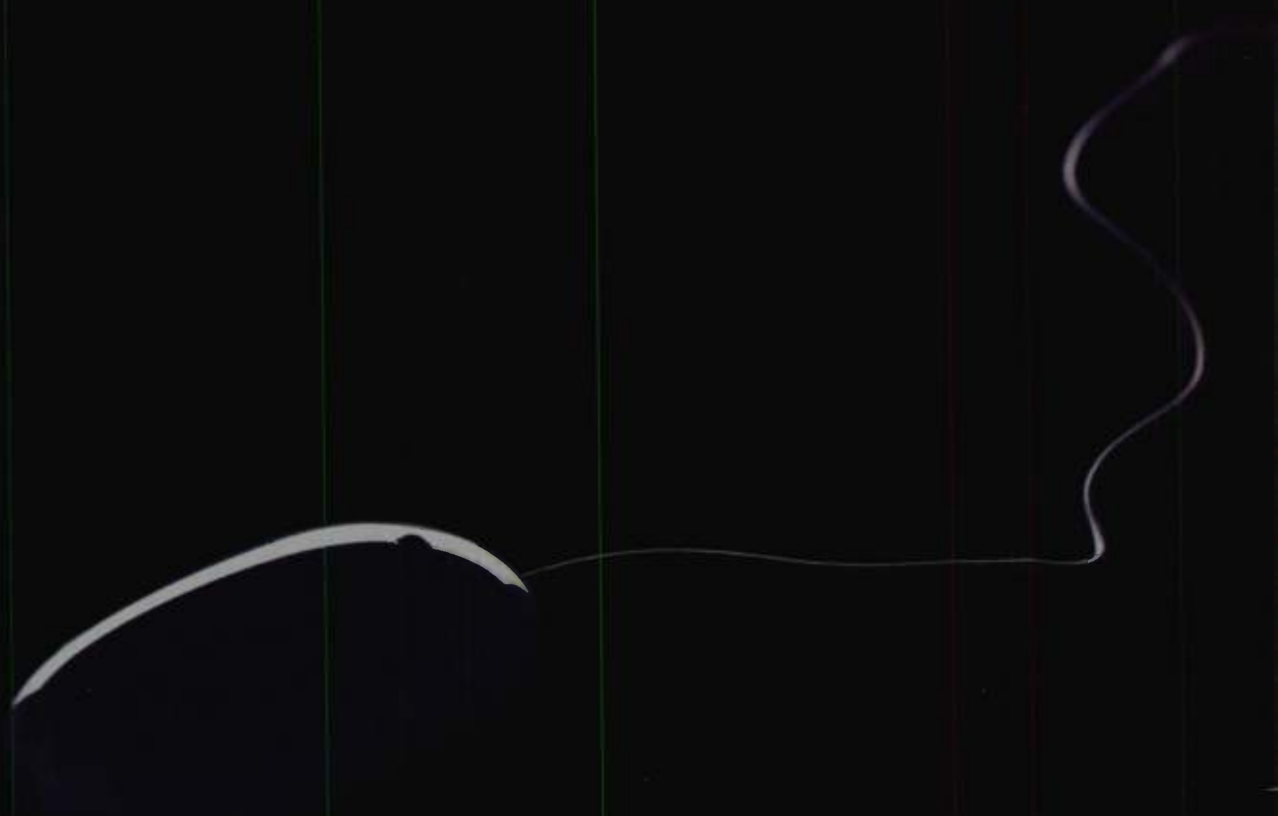
Virtually any plug-in can be useful in some way for the real-time and off-line treatments suggested this month. But in addition to those I've already mentioned, there are others that stand out as particularly powerful, interesting and colourful.

You can achieve super-saturated, pumping dynamics treatments with both the MOTU MasterWorks Compressor and the Limiter. The seemingly simple Chorus, Autopan and Tremolo can produce weird vibratos, shimmering stereo and rhythmic gate-like pulsations that all work great when used sparingly. Try the Echo and Delay plug-ins for gloriously weird tonal treatments when using short delay times and medium to high levels of feedback, especially if you automate the delay times.

Turning to third-party developers, you're spoilt for choice of unusual, 'out there' processors. The freeware MDSP LiveCut is another beat slicer capable of phenomenal effects, and worth the effort despite its inoperative bypass and weird off-line behaviour. Some of my favourite hard-to-categorise textural tools include the deceptively simple Audiodamage Vapor, Cycling 74's Hipno suite, dfx's Transverb (another freeware gem), PSP's Nitro and PSP84, and Audioease's RiverRun. For distortion and speaker effects, try out MDA's freeware Combo, iZotope Trash, or Audioease's amazing Speakerphone. Finally, for just plain weird, there's DP's own Ring Modulator and other frequency-based effects such as the freeware MadShifta (from http://bram.smartelectronix.com) and Ohmforce's Hematohm. SOS better

faster

cheaper



Mystical Synthesis

Making More Of The Mystic Synth

We dive deeper than the presets to discover what you can do with Cubase 4's most complex bundled soft synth.

John Walden

Although it shares some controls with Prologue (which I looked at in *SOS* April 2008), Mystic is a very different beast, with a synthesis engine that's centred around comb filtering and capable of producing some really interesting sounds. The Plug-In Reference PDF does its usual efficient job and describes Mystic's key controls, but it is less useful as a guide to the synth's architecture or as a programming tutorial, so...

The Mystic user interface, with the 'Multiband' spectrum-filter curve selected.

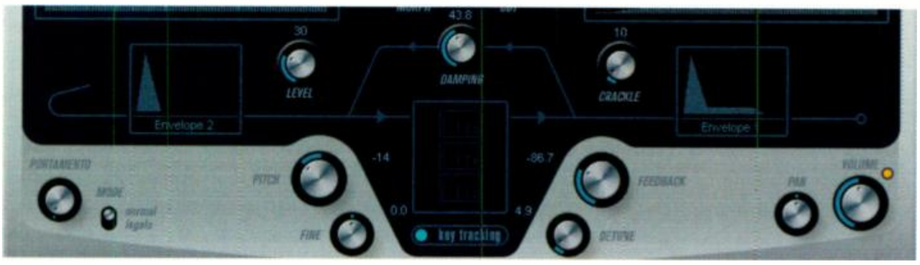
Magical Mystic Tour

I'll start with the basics. Sounds in Mystic are created in two main stages. The first involves the configuration of an 'impulse sound', which is fed into Mystic's three comb filters. The creation of the impulse sound is controlled using the topmost section of Mystic's interface, and the description of this stage in the Plug-In Reference is, I'm afraid, rather confusing. It tells you that the "Impulse Control section has two basic waveforms that are filtered through separate spectrum filters". While the two spectrum filters (labelled A and B) are present and correct and they can both be configured by the user (see the 'Bugs In Boxes' panel elsewhere in this article), only one Waveform menu is present, and this selects the waveform (from a choice of six types) to be sent through spectrum filter A. It isn't immediately obvious, but some comparative tests between the two waveforms leads me to believe that the waveform sent to spectrum filter B is fixed as a sawtooth (which would explain why there's no need for a second Waveform pop-up box to be provided).

The two spectrum filters offer a selection of preset contours, but you can also draw your own using the mouse. In normal use, setting a frequency contour in one display will configure an inverse contour in the other, but holding down Shift while drawing with the mouse allows the two filters to be configured independently. The shape of the contour dictates which frequencies are cut or boosted, and the effect of this can be heard clearly via a patch such as that shown in the screenshot above. Here, all the LFO, ENV, Event and EFX options have been turned off, and the comb filter has been removed by setting the feedback control to zero. In this configuration, and with a suitable ADSR envelope configured for Env 2 (by default, this is the envelope that is applied to the output of the Impulse Control stage), the influence of the spectrum filters on the impulse sound can be examined. The 'Multiband' spectrum preset provides a good way to demonstrate this, because it contains a series of extreme peaks and troughs. Setting the Morph control to either 0 or 100 (so that you only hear one or other of the spectrum filters operating) and then simply playing a series of notes up or down the keyboard allows you to hear where these notches or peaks are, as the frequency content of the sound will change.

The four other controls in the Impulse Control section (Coarse, Cut, Morph and Raster) are fairly straightforward. Coarse offsets the pitch of the impulse sound (+/-4 octaves), while Cut acts rather like a standard high-cut filter, with lower settings reducing the high-frequency content. As I implied above, the Morph control adjusts the balance between the output of the two spectrum filters. Finally, the Raster control is used to remove harmonics from the impulse sound. As with the Cut control,

The Impulse Control section.



The comb filter section.

lower values tend to produce a darker sound, although the effect is somewhat different.

Comb Together

The impulse sound is then passed to the second stage of the Mystic engine — the comb filters. As can be seen on the first screenshot, an envelope (the default is Envelope 2) and a Level setting are applied to the impulse sound prior to it being fed to the these filters. If the envelope includes significant decay, sustain and release times, the character of the impulse sound has a stronger influence on the timbre of the final output. However, for some presets the impulse sound is just an impulse, with very short decay and no sustain or release elements. With this sort of envelope shape, the detailed settings in the Impulse Control section are somewhat less significant.

The influence of this initial envelope stage can be illustrated by loading the 'Pizzicat' preset (it is worth disabling the Delay and Modulation processing in the EFX window to make the influence of the envelope more obvious). This patch uses an envelope that consists of an instant attack, a short release and zero decay and sustain times, and it produces a sort of bright, fizzy guitar tone. If you adjust the properties of Envelope 2 (in the ENV window), the change in the sound is quite dramatic, as more of the character of the impulse sound reaches the comb filters.

Bugs In Boxes

On my test PC, there's something a little odd about the behaviour of the displays within the Mystic spectrum filter boxes. The contents of the displays do not always seem to correctly update themselves when I'm browsing through the presets and, for example, if I have two instances of Mystic open and both have the same preset selected, the spectrum filters sometimes show different filter curves. This behaviour can be cleared by saving, closing and then re-opening the Project, but it is a little irritating if you're trying to study how some of the presets are constructed, as you can't rely on the displays to be accurate. I've reproduced this behaviour on a second PC so I suspect it is a genuine bug, although I'm not sure whether it is also an issue for Mac users.

A number of the other controls in the comb filter section are pretty self-explanatory, so I'll be brief: the Pitch (coarse pitch) and Fine controls provide overall pitch adjustment for the synth's output; the Key Tracking switch operates in the expected fashion; and Crackle also does pretty much what it says on the tin (it adds noise to the comb filters). For any sustained sounds, unless you're going for a special effect, a little Crackle goes a long way — although the 'Granulator' preset provides a good example of what's possible.

More interesting are the Feedback, Damping and Detune controls. Feedback determines the amount of signal sent back into the comb filters for reprocessing. If set to zero (the 12 o'clock position), there's no feedback, essentially turning off the comb filter — and while this may defeat the object of using Mystic in the first place, it is useful when programming a sound from scratch, because it allows you to hear the character of the impulse sound in isolation. Both positive and negative feedback values can be set. More extreme values of both generate sounds with longer decays, while negative values create a more hollow sound (good for bell-like tones) than do positive ones — and while it is by no means universally true, Mystic's more melodic presets tend to favour positive feedback values.

The Damping control influences the low-pass filter that is applied to the sound being fed back into the comb filters. By removing the higher frequencies, it causes the sound to become gradually softer over time (each time the sound goes through the feedback loop more high-frequency content is removed). This can be used to mimic the character of many acoustic sounds, such as, for example, a plucked string, where the higher frequencies decay more quickly than the lower ones. If you want to avoid this, turning the Damping control up to 100 will switch off the low-pass filter.

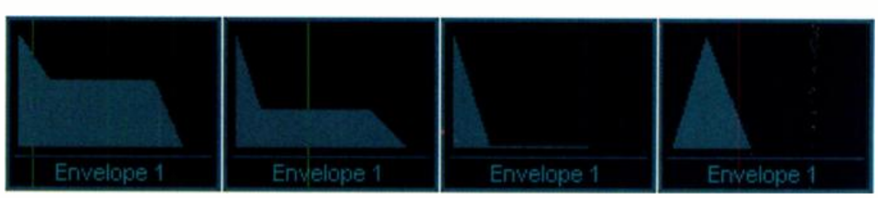
The Detune control offsets the frequencies of the notches within the three comb filters, detuning them relative to each other. At low settings, this can create a gentle thickening of the sound (not unlike a chorus effect). At higher settings, this detuning can get pretty extreme (as illustrated by the 'Doppler' preset) and the output becomes more sound effect or sound design than melodic.

The output from the comb filter stage passes through a standard ADSR volume envelope (which, by default, is Envelope 1 from the ENV window). The overall Volume and Pan controls operate as expected and Portamento (with normal and legato modes) is also included.

String Me Along

Mystic isn't an obvious choice for creating convincing emulations of real string-based instruments, but (as some of the presets show) it's capable of producing some interesting synthetic string-type sounds. As noted above, the 'Pizzicat' preset uses the comb filter input envelope to define a very short impulse sound, and this is typical of many of the 'plucked string' presets in Mystic that use similar forms of envelope shapes (for example, the 'Pluck One', 'Cello Two' and 'Plucked String' sounds). This makes sense, as the impulse sound is attempting to reproduce the effect of the initial striking or plucking of the string — which is a very short-lived event. Unsurprisingly, the majority of the string-style patches (including those listed above) use the String Body spectrum filter preset. However, some of the other six waveforms can be used, and the other Impulse Control settings can vary over quite a wide range and still be perfectly useable.

Many of the string-style presets share a number of other similar settings as well. Low (or zero) Damping values tend to be used, whereas Feedback values tend to be high (130+). This combination means that there's plenty of comb-filter processing, causing the sounds to lose higher-frequency harmonics quite quickly. Given that most of the string-style presets are intended to be melodic, low (or zero) values of Detune are the norm. The final common feature is the output envelope. Some examples from the presets listed above are shown in the screen shot below. The combination of fast attack, short decay and low sustain values (as used



The comb filter output envelopes for the 'Pizzicat', 'Pluck One', 'Plucked In' and 'Cello Two' presets.

for the 'Pizzicat' and 'Pluck One' presets) emphasises the initial attack of the sound. The 'Plucked In' preset is similar but the lower sustain value means that it is more like a plucked string than a bowed one: it doesn't simulate the continuous excitation of the string that would be provided by a bow. In contrast to all of the others, the 'Cello Two' preset features a slower attack phase and slower decay, and this results in a sound that is more suitable for slower, single note, melodic lines.

Finding A New Pad

Mystic excels in the creation of pad or soundscape-style sounds. Some fabulous presets illustrate this, ranging from the fairly musical 'At The Movies' (which seems to be both mellow and urgent at the same time), via the almost organ-like 'Rising Pad' and 'Trembling Pad' (quite foreboding in the lower register), to the more unsettling and abstract 'Arhythmic Chaos' or scary (and speaker damaging) 'Love No Screaming'.

When exploring pad-type presets such as those listed above, it soon becomes obvious that a wide range of settings in the Impulse Control section can be used. Indeed, any of the basic waveform types are suitable and the Randomize option for the Spectrum Filters can throw up some perfectly usable filter shapes. However, there are some controls in the comb filter stage where the settings show more similarities. Output envelope shapes (usually Envelope 1) often feature zero decay, high sustain levels (often 100) and long release times. Attacks can be short or longer (so that the sound fades slowly up to full volume when a note is played). In many cases, the comb filter input envelope (usually Envelope 2) has similar properties — very different from the string-style sounds discussed above — and this means that the tonal characteristics of the impulse sound have a stronger influence on the overall timbre. For all the presets listed above, Feedback levels are positive and in the mid- to high range. For more musical pads, Detune and Crackle values tend to be low (less than 10), but if you want to create a 'pad gone bad', slightly higher values for these two controls can soon add an unsettling feel.

However, the key element that adds interest to most pad sounds is the way in which they may change (subtly or otherwise)

Don't Just Read It: Hear It!

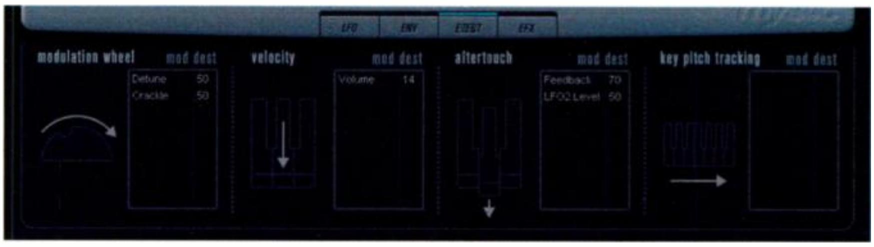
To complement this article, we've placed some audio examples, along with descriptive commentary, on the *SOS* web site, at: www.soundonsound.com/sos/jun08/articles/cubasetechaudio.htm

The LFO window is great for adding movement to a sound.

with time as the note is sustained, and to achieve this we need to turn to the ENV, Event and LFO windows. The last probably offers the most instant fun, so I'll concentrate on that here. Mystic (like Prologue and Spector) provides two independent LFOs and, as can be seen in the screenshot above, the speed, depth and wave type of both the LFOs can be easily configured. The LFOs can then be assigned

In this example, Envelope 3 is being used to vary Feedback, Damping and the depth of LFO1, with the last also influenced by note velocity.

to modulate a number of Mystic's key controls by clicking within the Mod Dest or Vel Dest window and selecting a parameter from the drop-down list. The editable value



Hands-on control of Mystic is also possible via the Event window.

next to the parameter controls the degree of modulation, and assignments within the Vel Dest box are also influenced by note velocity. As can be seen in the screenshot at the top of this page, an LFO can be assigned to more than one parameter.

Good potential targets for the LFOs are the Cut, Morph, Damping and Feedback controls, as all these parameters influence the tonal qualities of the sound. Negative values can also be set, changing the direction of the LFO modulation. If a slow evolution of the sound is required, a slow speed value and the sine, ramp up or ramp down settings provide a good starting point. A faster LFO speed can produce more rhythmic effects — and the 'Arhythmic Chaos' preset provides an excellent example of this. If you want to create something that sounds a little more disturbing (and less melodic), target the Detune, Crackle and Pitch controls... things can get ugly very quickly!

Both the ENV and Event windows give access to further options for developing evolving pad sounds. Envelopes can be assigned to modify Mystic controls. For example, the Cut value could be altered by Envelope 3 and therefore change with time. However, LFO speed and depth can also be controlled via an envelope, and any of the parameters controlled by the LFO will thus also be influenced by the envelope. For hands-on control, Mystic parameters can also be assigned, in the Event window, to the mod wheel or aftertouch (if your master keyboard supports it), and this allows you to control how the sound evolves as part of your performance.

Be Careful Out There

Finally, I must sound a brief note of caution... Mystic can generate a fairly powerful output when its controls interact, so make sure you practice 'safe sound-making': experimenting at a modest monitoring level will ensure that you avoid nasty surprises. This aside, it should all be plain sailing, with a surprising array of sounds waiting to be coaxed out of Mystic's depths. SOS

Space Exploration

Pro Tools Plug-in Masterclass: TL Space

Like many DAW manufacturers, Digidesign offer their own convolution plug-in as an add-on for Pro Tools. We explain how to get the best from TL Space.

Mike Thornton

This month's Pro Tools workshop is the first in an occasional series where we'll take a detailed look at one of the more complex plug-ins that Digidesign distribute. First up is the TL Space convolution reverb plug-in, which is available in both TDM and RTAS flavours. As with all of their plug-ins, a 14-day demo of TL Space can be downloaded from the Digidesign web site (see the 'Links' box), so if you're curious, you can install the demo and use it in conjunction with this workshop, even if you don't yet own TL Space yourself.

The TDM version actually comes as three different plug-ins, Short, Medium and Long, which offer progressively longer reverb times at the expense of using more DSP. The longer versions can 'bridge' multiple DSP chips in one instance of a plug-in, up to a limit of eight DSPs for a quad input/true stereo output instance, but the maximum reverb time available in the TDM version is 3.4 seconds. If you want to hear the nuclear cooling tower in all its glory, then, you need to use the RTAS version. This is included with the TDM version as well as being available as separately, and uses the host computer's processing power instead.

What Is Convolution?

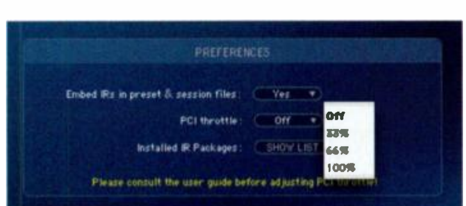
At a mathematical level, convolution multiplies every sample in one waveform or impulse by the samples in another waveform. When that second waveform represents an 'impulse response' created by recording the decay of a gunshot or sine sweep, it can be used to apply reverb 'sampled' from a real acoustic space. Once you have an impulse of your favourite space and put it into a convolution reverb, the result will sound as if the audio was played in that space. However, you can actually use anything you like as the impulse response: for a wacky example, try convolving someone saying 'Hey' with a snare drum hit. You will find that the snare drum says 'Hey' each time it is hit!

Space Is The Place

Let us move on from the theory and take a look at how TL Space implements these principles and how you can use them creatively. The top half of the TL Space interface is the display section, and can be switched between any of four modes. Waveform displays the impulse response waveform, while Picture Preview shows pictures of the location in which the IR was created. The third page is Snapshot: TL Space offers up to 10 Snapshots that store the IR waveform and all the control settings, and has been optimised for very fast loading. Switching between these snapshots can be automated, so enabling the user to program TL Space to change 'environments' very easily with a single automation function. The final page is Preferences. Among other things, this allows you to turn on and off the feature that embeds the IR waveforms in the presets and Session file. This feature is very useful when transferring Sessions from one system to another, as it means you don't have to remember to copy the impulse library too. The 'PCI throttle' on the TDM version is there to help if you get any dreaded -6042 errors, especially when using PCI video capture hardware.

The meter section stays visible all the time and will display the appropriate number of meters depending on the input and output configuration chosen by the user.

The five pages shown in the bottom half of the screen display the controls used to vary the settings. The first Levels page gives you separate control over the levels of the early and late reflections, and if you have a surround instance of TL Space, then on this page you will also see controls for



It's worth being familiar with TL Space's Preferences page. The 'Embed IRs' option is particularly useful if you need to archive your Sessions and open them on other machines.

Links

- **TL Space TDM**
  www.digidesign.com/index.cfm?navid=115&langid=51&itemid=4421
- **TL Space Native**
  www.digidesign.com/index.cfm?navid=115&langid=51&itemid=4463
- **TL Space Impulse library**
  www.digidesign.com/index.cfm?langid=51&navid=115&itemid=4782
- **Voxengo Deconvolver**
  www.voxengo.com/deconvolver/
- **Voxengo impulse responses**
  www.voxengo.com/impulses
- **FuzzMeasurePro 3**
  www.supermegaultragroovy.com/FuzzMeasure/
- **Cyber Kitchen Sound Design Enterprise impulse responses**
  www.cksde.com/f_6.htm
- **Numerical Sound impulse responses**
  www.numericalsound.com
- **Spirit Canyon Audio impulse responses**
  www.spiritcanyonaudio.com

Front, Rear and Centre levels.

The second page deals with Delays. TL Space goes beyond the normal pre-delay that most reverb units offer and provides separate control over the delay time of the late reflections. The surround version also features controls for the front, rear and centre delays of up to 200ms. The Pre Delay control offers between -200ms and 200ms of delay: negative pre-delay values can bring about subtle or radical changes to the sound. Small amounts of negative pre-delay will mean that the plug-in does not use the 'early reflections' part of the impulse response, while larger amounts of negative pre-delay enable you to isolate the end of the reverb tail as a reverb in its own right, in a way not available on conventional reverb units.

The third page is dedicated to the Early portion of the impulse response (IR), and so the early reflections of the resulting reverb. The Length control determines what TL Space will consider to be the early part of the IR. It is best to adjust this whilst viewing the waveform display; you will see that as you adjust the Length control, the section of the IR that TL Space allocates as 'early' is highlighted. The Length value is specified in 'm' rather than 'ms', but is actually measured in milliseconds, not metres!

On this page you can also adjust the relative volume of the early reflections, with the Size control, and enable a high-pass filter to reduce the low-frequency content of the early reflections. This can help with low-frequency boom in the reverb path and reduce LF cancellations when mixing reverb back with the dry signal. Finally, there is a Balance control, which enables the user to adjust the left/right balance of the early reflections. This control will help with the apparent position of the reverb input in the stereo image.

The fourth page, Reverb, controls the reverb 'tail' section of the IR. High- and low-frequency shelving EQs both have frequency and gain controls, and are prior to the convolution process in the signal chain. The Width controls the perceived stereo spaciousness of the reverb tail, but remember that if there is relatively little stereo content in the IR, this control won't have much effect. Balance controls the position of the reverb tail in the stereo image and, finally, the Reverse control reverses the IR waveform and will also control the total length. Again, it is best to adjust this whilst viewing the waveform display; very helpfully, this control displays its results in beats per minute, to help the user sync the reverse effect with the tempo of the piece. Any impulse response longer than five seconds will be truncated when Reverse is used.

Finally, the Decay page allows users to control the decay of the low-, mid-, and high-frequency elements of the IR, using a three-band crossover-type EQ section. When using the surround version there are separate controls for the front and rear channel decay.

Installing Impulse Responses

The boxed version of TL Space comes with a good selection of IRs, but if you are downloading the trial version of TL Space from the Digidesign web site you will need to download some IRs to get you going. All the IRs are available from the Digidesign web site, and as well as the Core packages that are supplied with the boxed version, there is a range of additional packages. Notice that most of the files are large, so be aware of this when downloading.

Once you have downloaded some IR packages, check that the downloaded file has been 'unzipped' and you know where the IRs are on your system. Open the IR Browser within TL Space, click on the Edit button and a menu will appear. Select Install TL Space IR Package and a navigation window will open. Navigate your way to where you left the unzipped file(s), which will be in the proprietary '.tls' lossless compression format, and install that package. You can repeat this process for as many packages as you have. Note that TL Space puts its own IRs into a special folder on your system.

You aren't restricted to Digidesign's own libraries, however. TL Space reads a wide range of IRs, including WAV, SDII and AIFF, with sample rates from 22kHz to 96kHz and bit depths from 16 to 32 bits. TL Space also supports JPEG files for the location images.

To use IRs in these formats, choose the 'Import Other IR Folder' option from the Edit menu in the IR browser. You will need to make sure that your third-party IRs are pre-arranged in a folder, as TL Space won't import single audio files that are not inside a folder. I downloaded a number of IRs and grabbed the relevant images from the web sites too. For example, I downloaded a response from a nuclear power-plant cooling tower, created a folder called Cooling Tower, put the IR and the JPEG inside it and pointed TL Space at that folder. TL Space read both the IR and the picture. Note that when importing third-party IRs, TL Space doesn't copy them to its own folder, so you shouldn't move third-party IRs once you have imported them. I have a dedicated folder on the same drive I store all my samples and sound effects and I make sure they are managed in the folder format that TL Space understands.

Although TL Space accepts a wide range of IR formats, you cannot import IRs from Altiverb after v4, Waves IR1 or the Sony hardware convolution processor amongst others; all these rival manufacturers keep their IRs in a proprietary format to protect the work they have done in acquiring quality libraries. There is, however, a good number of sites offering free IRs, although the quality varies — some have missing reverb tails or left-over sweep tones in them. There are also commercial IR libraries from companies like Numerical Sound and Spirit Canyon Audio. For more details and reviews of these and others, have a look at the article on convolution processing with impulse responses in the April 2005 issue of *SOS* (www.soundonsound.com/sos/apr05/articles/impulse.htm).

Creating Impulse Responses

The real fun with convolution lies in recording and using your own impulse responses, which isn't as difficult as it sounds. At its simplest, all you need to do is create an impulse sound like a gunshot from where you'd like the sound to originate from in the space, usually the stage, and record it by putting the mics where you want to listen to the sound, usually somewhere in the audience.

The first issue is how to create a suitable impulse sound — if you are going to create an impulse that requires no post-processing, some sort of impulse sound (as close as possible to the theoretical ideal of an instantaneous sound containing all frequencies at once) is going to be necessary to excite the space. Handclaps are not really suitable, as the attack isn't really good enough. Some people pop balloons, but unless you use really big ones, there won't be enough low-frequency content in the impulse. The oldest technique is to use a gun, and I have heard that some people work with a power-actuated nail gun that ▶

uses .22 blanks to fire a nail into concrete! However, using an actual gun — even a nail gun or starting pistol — probably causes more problems than it solves in the current climate. Other options include banging two books or wooden blocks together, but both will tend to produce a coloured sound.

Most professional impulse response libraries are recorded instead using a swept tone file. This method requires good-quality audio playback technology to replay the sweep tone accurately within the acoustic space to be sampled. You then record the sound of the swept tone in the room, and use a so-called 'deconvolution' algorithm to extract an impulse response. If you run a Mac and have Apple's Logic you already have the tools to do this: for a detailed guide, read the February 2008 issue of *SOS* (www.soundonsound.com/sos/feb08/articles/logictech_0208.htm).

PC users shouldn't feel left out, as there is a Deconvolver application from Voxengo that will take a recording of a swept sine wave and convert it into an impulse response. The full version of Deconvolver only costs $39.95, and there is a free demo version that will allow three conversions per session.

For Mac users without Logic, the best solution is a software package called FuzzMeasure Pro. The current version (v3) requires Leopard, but the nice people at Supermegaultragroovy can supply a copy of v2, which is designed for Tiger, and will be what most of us Pro Tools users will need at this time. It currently costs $150, with a 14-day demo available, and it should be remembered that it is really an acoustics analysis application. However, IR recording and analysis functions are all built into the application, and once you have swept your chosen space it is possible to export an impulse response, which you can then import into TL Space or any other convolution reverb. Remember to take pictures of the location, so you can see it when you select it in TL Space!

Impulse Hygiene

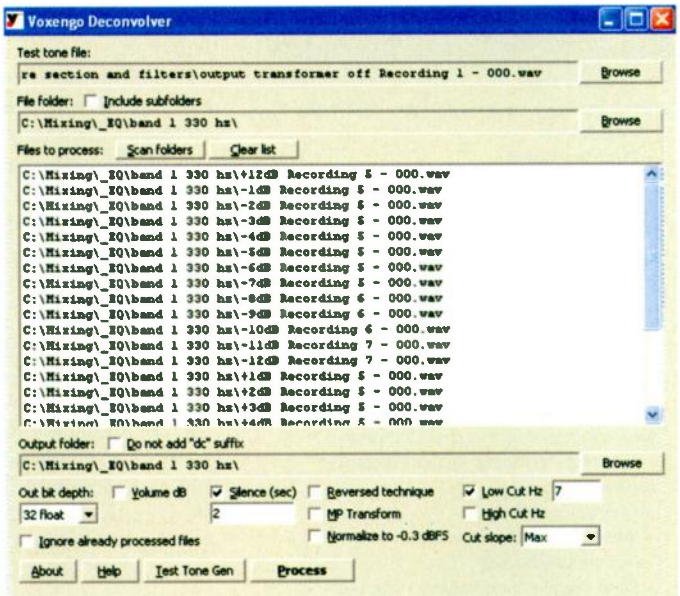
My first attempt at using FuzzMeasure Pro worked, but there was some low-level echo long after the main sound had finished, and the processed sound was quiet. So I imported the impulse I had made into Pro Tools, normalised it so that the impulse would be as loud as possible, then trimmed the front tight to the start of the impulse and finally trimmed the tail back to the impulse. I exported this file back to my impulses folder and re-imported it into TL Space. Both problems were solved. The processed sound was much louder and the low-level delayed echo had gone. The moral of this experiment is to always clean up your impulses before loading them into your convolution reverb.

It's worth remembering that you can 'sample' other audio gear, as well as acoustic spaces. The basic rule is that any linear process that is not time-variant can be successfully captured as an impulse response. This rules out compressors, distortion circuits and modulation effects such as chorus or flange, but includes reverbs, delays and loudspeakers, among other things.

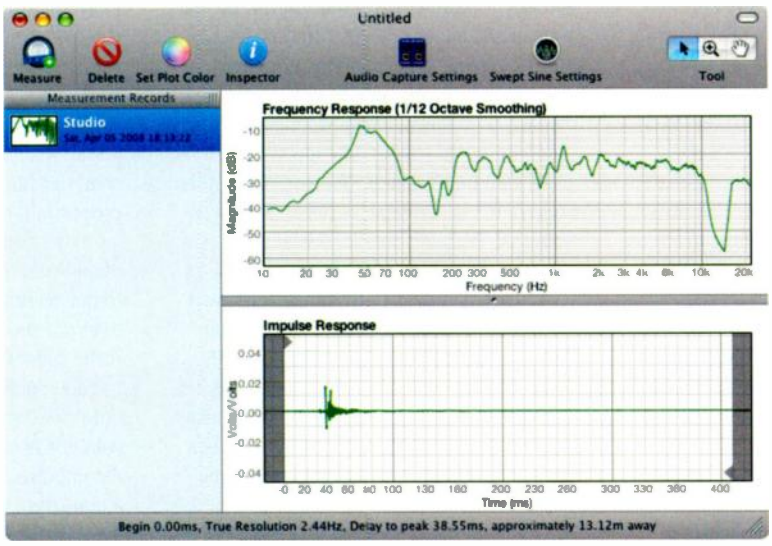
The easiest way to excite effects units and get an impulse response from them is to hit them with a single full-level sample, record the output, and use that sound as the impulse response. This way you can recreate patches from classic hardware reverb units and other effects processors and have them in a Pro Tools Session without having to own one of these classics. You should be able to track down impulse responses of most of the classic units on-line; indeed, quite a few come with TL Space, including the EMT250 and AMS RMX16.

You can extend this principle and take impulse responses from guitar amps as well as things like telephones and car radios. You can even take an impulse from a film location, so that when the dialogue is later replaced you can use a convolution reverb to process the replaced dialogue to match the original location. The TL Space Post Production set has a very good range of locations, and another set I find useful in post-production work is called Tiny Spaces and includes environments such as flowerpots, glass cups and a hose from a vacuum cleaner.

For those who might like to walk on the wild side or are into sound design there is a huge range of experimentation open to you by trying other sounds as impulse responses — I already talked about the idea of using a vocal 'Hey' as an impulse to process drum samples. Try taking the impulse response from the resonance of a violin body or piano soundboard, which can help sampled or synthesized sounds feel more natural. Using a drum loop as an impulse can produce some very unusual echo effects: pad sounds will result in drones and background textures, whilst short files produce complex comb-filtering sounds. You could even draw your own impulse responses using Pro Tools' pencil tool. Have fun and experiment! SOS



The inexpensive Voxengo Deconvolver is a PC utility that can create impulse responses from recorded sine sweeps.



FuzzMeasure Pro is a Mac utility that can 'deconvolve' a sine sweep to create an impulse response suitable for use in TL Space.

Better Logic Scores

Part 1: The Basics

If you want real musicians playing your composition but have no idea how scores should be presented for them, you can still make playable Logic parts with our simple guide.

Mike Saunders

Your orchestral MIDI track is coming along really well. You've spent hours playing in or entering MIDI information, and it sounds as convincing as it's possible to make it sound using samplers and software instruments, but it's not quite there yet — there's something missing. You feel it's lacking the authenticity of real instruments played by real players. This song is just too good to rely on MIDI: it needs a human touch and you decide that it's time to ship in a bunch of musicians as the icing on the cake. But musicians are going to need musical parts to play from, and you've never used Logic in this way before. What now?

Starters

If you can get the notes into the correct key and onto the stave in a neat and tidy manner, you'll be a long way towards achieving your goal. Let's have a look at how to turn the MIDI regions you have in your Arrange window into usable, playable, printed parts for the scrapers, blowers and thumpers when they turn up at your studio.

In this instalment we will start work on producing parts for a brass section (often referred to as a 'horn' section) and doing some basic formatting. The techniques I'll discuss in this section can then be applied to other solo instruments, such as flutes, basses and violins.

For this article a 'part' will be an individually printed piece of music for a musician to play from (for example, first trumpet or flute). A 'score' is the rather larger object that a conductor or producer would use to oversee all the individual parts and when they should be playing.

I have found that producing parts in Logic for musicians can be fairly quick and easy if you do things in the optimum order. In addition, there are some handy tips and tricks that can help in simplifying and speeding up the process.

The First Step

My first step is to save a new copy of the song file, renaming it something like 'mysongfilePARTS.iso'. I do this to reduce the risk of permanently altering my actual song file, to which I will ultimately record the real instruments. For safety's sake, having a copy to work on seems the best way to be sure you can always get back to your starting point. Even if you lose or ruin a part while experimenting and tweaking, you can always cut and paste from the original.

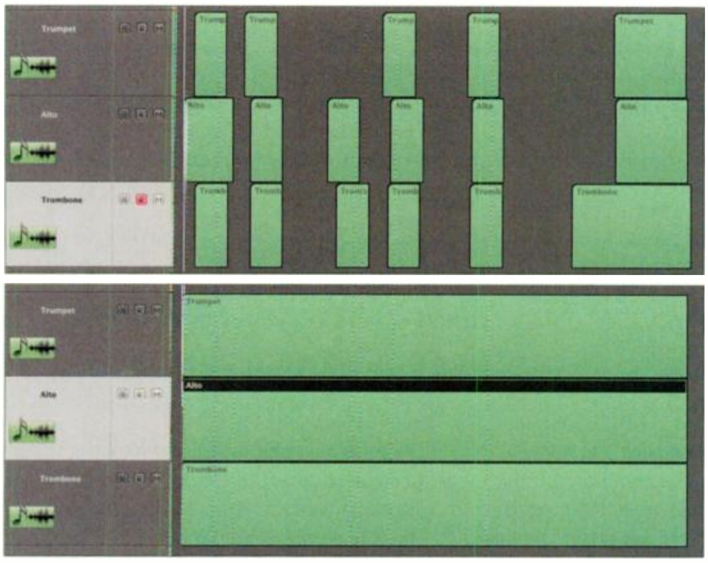
In the Logic manual you will read that there are a number of tools that can be used to produce printed parts without affecting the real-time playback of your MIDI data. We will have a look at these later, but for now let's look at the technique that I use for most of the time: deliberately affecting the MIDI playback of the song in order to achieve the best printed results.

Preparing Your MIDI Tracks

Let's assume that today's task is to produce printed parts for a four-piece brass-section. Our section will consist of two trumpets, an alto sax and a trombone. Before embarking on the actual score production there are a couple of tasks that you should complete in preparation.

You may have fragmented regions of MIDI information for each of your brass instruments, where you have played in just the bits you need throughout the song. Make sure that on each separate track you have merged the brass regions into one single region. Do this by highlighting all the regions on a track and using the Merge command (Region / Merge / Regions). If you don't have a single region, your printed part will end up with blank spaces rather than empty bars between the played sections.

Next, still in the Arrange window, if there



An unmerged region, as shown on the left (top), will leave blank spaces between the played sections on your printed part. Use the Merge command to create one region (shown left, below) and, with respect to your score, empty bars instead of blank spaces. This makes it much easier for your players to follow.

is a chance that you will ever want to use these parts live, make sure that the brass regions run from the very start of the piece to at least where your guys will finish playing, if not to the end of the piece. Drag the left-hand edge of each region to start at bar 1 1 1 1 (assuming this is the first bar of the piece), and, if necessary, elongate the right-hand edge to the end of the song. Do this for each of your brass-section tracks. If your section don't start playing until bar 68, you can start the regions nearer there, but starting these parts at the beginning can help the players with their orientation.

Splitting Your Parts

Usually you will need a separate printed part for each 'voice' or instrument of your section. You may have one region with what could be all four parts within it. You may, as above, have two trumpet parts in a single region on a track (as they are using the same sound source), the alto on its own track and the trombone separately. So you have three brass tracks for four parts.

To have these played by your real brass section you will need to produce an individual part for each of the brass players: the first trumpet (usually the highest part), the second trumpet (the remaining next lower part), and the alto sax and trombone third and fourth respectively. All solo/single-note instruments such as brass, woodwind/reeds and strings (first violin, second violin and cello, among others) will require this sort of treatment, as their players are not like pianists, who can read and will play multiple parts from one stave.

To extract two parts from a single region, first create another trumpet track below the first in the Arrange window. Copy the trumpet region onto this new track so that you have two identical regions on two tracks. This will force the score into providing another stave below the first. This gives you the separation that will be required for printing.

Work through the first region deleting the lower (second trumpet) part, then work through the second region deleting the first trumpet part, thus leaving different parts on two channels and resulting in four brass-section parts ready for action. Now that you're into recording real instruments, you could consider recording the MIDI data on separate tracks as you write your orchestration. This will reduce the likelihood of problems when you're at the stage of printing the parts.

Tidying Your Parts

If you have created a 'natural' sounding performance by 'humanising' your MIDI information, or you have played it in from a keyboard 'freestyle', rather than having it perfectly quantised, you will need to produce parts that look as though you've quantised them and played them like a computer, in order to get your musicians to play them how you want to hear them — like humans!

As you are using a copy of your original song file, you should be able to quantise each part in the Piano Roll (Matrix). This is the simplest way of tidying up the part, but make sure, even though it'll sound computerised, that it's still playing the notes in the correct places, with no strange overlaps or chords.

Alternatively, you can go into the Score window and use Logic's special score quantisation options to tidy up. This quantise differs from that found in the Piano Roll, as it doesn't affect the playback of the MIDI information, just the visual position of the notes on the printed part. This is a 'quick fix button' and it does work in most instances.

Select the part you are working on, open it in the Score window and select the stave by clicking on the lines (not notes or clefs) so that the lines turn blue. Click on the quantise box situated under the top bar of ▶

Stream, Beam & Broadcast

Rogue Amoeba's excellent Airfoil software, which allows you to stream any audio from your computer using Apple's Airport Express, has been updated to version 3. It can now transmit audio to wireless-enabled devices, such as other computers, Internet radios and Apple TV. I've been using it to beam audio from the G5 in my studio to laptop speakers and a TV sound system — it's very convenient not to have to burn a CD every time you want to check a mix. Using an additional FM transmitter (costing just a few pounds from eBay), I can also hear how my mixes sound through both transistor and car radios. So many professional mixing engineers have told me they check their mixes in the car that I've started to wonder why recording studios aren't built inside automobiles! Airfoil also synchronises video with audio, so you can choose to watch movies on your laptop while listening to the audio wirelessly through an Airport-enabled hi-fi. I recently mastered some tracks through the stereo in my living room using Airfoil. I used Logic's linear EQ and compressor to master, and acheived excellent results — it's the system I know the 'sound' of best. The latency means that you have to wait a second or so for any parameter change to be recognised, although it isn't too much of a trial even for an inveterate knob twiddler like myself, and it means that I spend more time reflecting over each change rather than moving on to the next frequency too soon.

If you co-produce with people in different geographical locations, it can be a real pain to keep sending off MP3s of mixes for evaluation. Believe me, when you've emailed 25 versions of a track in a single afternoon to California, it doesn't feel as though technology makes things any easier! Rogue Amoeba have come to the rescue again with Nicecast. This utility is designed to let you set up your own Internet radio station, and can allow distant colleagues to hear any mixes in real time, so that conversations (and blinding rows!) can be conducted via Skype or iChat. In addition, the company's web site states that you can use Nicecast to broadcast 'live events'. I've been using it to send audio over the Internet in real time and, as long as your bandwidth is sufficient, it works pretty well. There is also the added benefit of being able to distribute your finished music via your own radio station.

Airfoil costs a mere $25 ($10 to upgrade from version 2), while Nicecast is $40. Both are available from www.rogueamoeba.com.
Stephen Bennett

the left column and bear in mind that any setting chosen here will affect the whole of the selected part from start to finish, but will not alter the MIDI playback.

There are many generic quantisation settings available, most of which you will recognise from previously working with MIDI. However, there are extras specifically for score writing that allow for two types of quantisation at the same time, such as '8,12' and '16,24'. For example, using 8,12 stops the score displaying anything smaller than eighth notes (quavers), while also allowing eighth-note triplets. Similarly, 16,24 allows nothing smaller than 16th notes, while also still showing 16th-note triplets. These settings take a bit of working with to be sure of the correct one, so experimentation is the best way forward; you're looking for the neatest result, and the setting that works for one part may not work for another, depending on its content.

Another method for neatening your MIDI parts involves the quantise tool from the Score window toolbox. (This can be useful if you only have a few notes obviously out of place.) Select the tool, click and hold individual notes that are wrongly represented, and then apply the quantise value you require. This can be hit and miss, depending on where you want the note to be positioned and where Logic wants the note to go. Also, this tool does affect the MIDI position, as it works the same way as in the Piano Roll, so be careful if you're not using a copy of the song file.

A quantise setting of16,24 stops the score from displaying anything smaller than 16th notes (semiquavers)while still showing 16th-note triplets.

A setting of 32,24 allows up to 32nd notes (demisemiquavers) to be displayed, while also still showing 16th-note (semiquaver) triplets.

The most reliable way that I have found of getting around quantising issues is to use a combination of the above methods, by working with the Piano Roll (Matrix) open at the same time as the Score window (setting up a new screen set allows for quick access). Now adjust the notes on the Piano Roll, selecting them individually or in groups and dragging them minutely in the direction they need to go, until they are in the correct places on the score. This way you have the visual grid of the Piano Roll to refer to, and by using Ctrl+shift+drag to move notes you can see that the tiniest movement can have the desired effect on the part.

Although the editing process I've just described definitely affects the MIDI playback, it does give you a good chance of getting just what you need on the printed part — this being the main reason I use a copy of the song file to do the score work on. It can also mean that you end up using much smaller overall score-quantise values without compromising readability for the musicians and, as a result, can speed up your preparation. Additionally, it's possible to lengthen and shorten notes within the Piano Roll more easily than you can in the Score window.

How Are We Doing?

Once you have been through all four of the brass regions using some or all of the techniques above, you should have a fairly accurate set of parts ready to print for your blowers. At this point you haven't added any 'markings', such as those for dynamics, articulation and speeds, but as mentioned earlier you can talk this through with your guys when they arrive. I suggest you give them their parts, play them the original track so that they can follow the music, and then discuss with them the way you want things played.

There are further formatting options available to embellish your newly created parts, such as Key Signatures, Time Signatures and Transposing, and I'll take you through these in greater detail next month. SOS

Interpretation, Syncopation, No Overlap ... What?

At the left side of the score window you will see three tick boxes: Interpretation, Syncopation and No Overlap. These are a very useful set of general formatting tools that visually affect the whole of the part that you are working on.

Interpretation tries to show the part in its simplest form by lengthening note values (visually — not the MIDI data itself) to reduce the number of little rests that appear between notes that you may have played slightly short. This makes the part clearer and smoother to read. The two examples below show a score with Interpretation both ticked and un-ticked, the un-ticked version shown above and the smoother, ticked version shown below.

Syncopation is a term for notes that are played away from the main beats of the bar, or off-beat. Computers like to see things their way and humans theirs, so the Syncopation box tells Logic to show the syncopated sections in a more easily read, less cumbersome manner. Sometimes, however, it produces syncopated passages that, though technically correct, don't quite show up the way humans would write them using the 'rules of theory of music'. So when you use this setting, be prepared for neither ticked nor un-ticked to be completely correct for your musicians. I find that musicians are quite used to this, as computer-printed music is common these days, and they adapt pretty quickly.

No Overlap does as it says, by not allowing the score to show overhanging notes that may have been played in the MIDI track. Often, to get a solo line sounding really smooth, you will have let notes hang on longer than they should, overlapping in order to get the effect of a smooth transition. The two examples to the right show this, with the first example containing adjoining notes that will appear on the score without the use of No Overlap. The score would view this as a chord rather than a solo line. Clicking the No Overlap box should help to rectify this for you, without your having to go through the MIDI data and shorten all the troublesome notes.

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Beats Working

Creating A Rhythm Track In Sonar

Each TTS1 sound can be edited to some degree. In the case of drums, you can edit each drum sound individually.

A walk through the process of composing and processing a drum track highlights how to use various Sonar features and useful shortcuts – some of which you may not have come across before.

Craig Anderton

This column usually focuses on a particular subject in depth. That's great for building up a toolbox of techniques, but this month I want to try something a little different: covering the process of working on a particular project. In this case, we'll take a look at how to create a beat in Sonar, then we'll use that as the foundation for a rhythm track designed for an electro/dance-style piece of music.

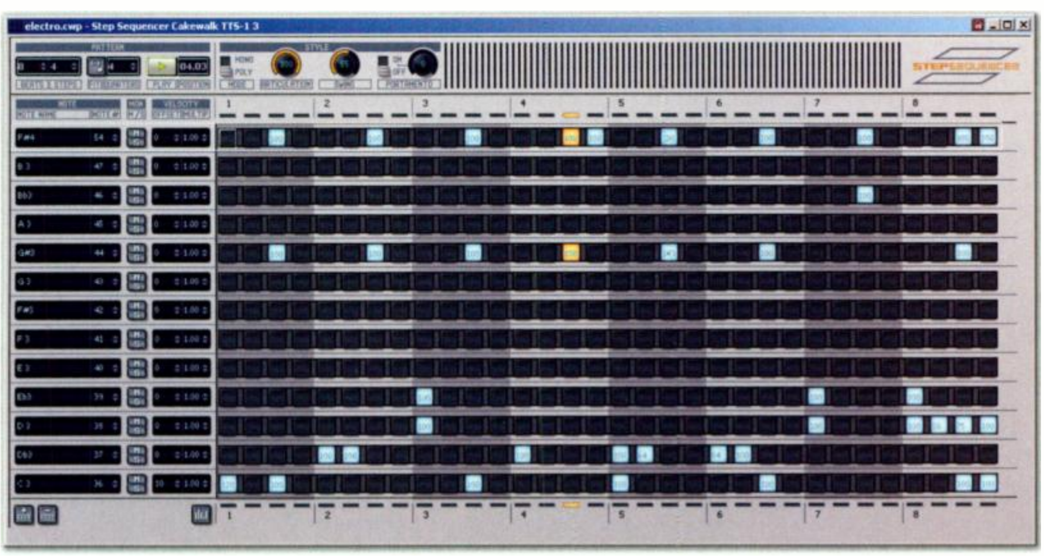
We Got The Beat

The foundation of this genre is a strong rhythm section, typically with analogue-sounding drums. For those types of sounds, you needn't look any further than Sonar's TTS1 soft synth, which has the excellent 'Analog Set' analogue/Roland TR-type drum sounds. So, after opening a new project, let's get started.

1. Go Insert / Soft Synths / TTS1.
2. Choose your desired soft synth options in the window that appears. I make sure that Synth Track Folder is ticked, so the MIDI and instrument tracks end up in a folder, and also I tick Synth Property Page, so the instrument's GUI opens up on insertion.
3. We'll put the drums on MIDI channel 10 (track 10 on TTS1), just to be in step with the General MIDI spec. At TTS1, click on the track name for channel 10, then go Preset / Drum Set / 026 Analog Set.
4. Now let's create a drum pattern. Turn your attention to the TTS1 instrument MIDI track and set the MIDI channel output to 10.
5. You can use any method of note entry you like for creating the part, but let's use the Step Sequencer for now. Click on the MIDI track so that it's selected, then go Views / Step Sequencer.
6. Set the number of beats and steps for the Step Sequencer; 8 and 4 are good starting points, respectively.

Now we're ready to create a drum pattern. If you click on the note names toward the Step Sequencer's left, you can audition the various drum sounds. Let's start with C3, the kick. Click on the Step Sequencer Play button, then on the steps (the squares) in the C3 row where you want a kick-drum sound. If you click in the wrong step, right-click on it to clear it.

You'll find suitable backbeat/snare-type sounds on D3, E♭3 (claps) and E3. You can always click all three of these on the same beat if you want a really big sound. There's a closed high-hat on G#3 and an open high-hat on B♭3. These are good for



The completed drum pattern now resides in Sonar's step sequencer. The gold steps indicate ones that are playing back.

adding an off-beat effect. A 'low clave' type of sound suitable for accents can be found on D♭3. Don't forget that you can add as many rows as you like to accommodate more sounds — just right-click to the right of any row and select Insert Row. You can drag a row into any position. For example, in the Step Sequencer screen shot (below left), I added another row and dragged it to the top to add the tambourine sound at F#4. This is also how you cut, copy and paste row steps, as well as shifting steps left or right if you want to try out some variations.

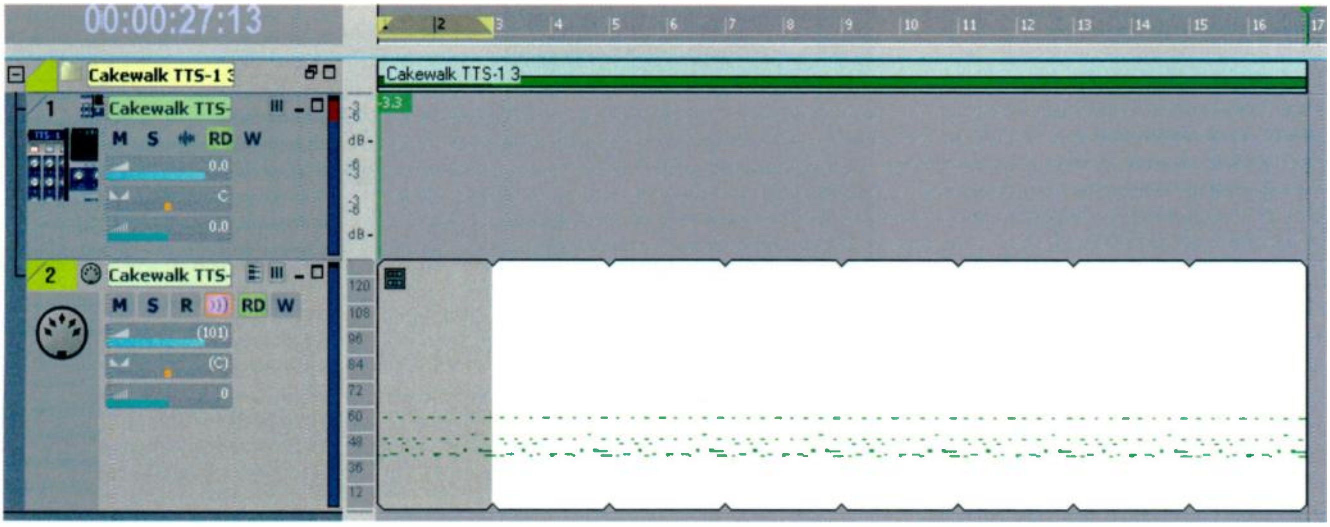
When creating the beat that kicks off the creative process for a tune, I try to produce one that's as complete as possible — in other words, the beat that will be used at the peak of the song. I find it more natural to simplify the beat for other sections, by stripping out parts, than to start with a simple beat and embellish it later on.

Create a beat using whatever sounds you like, then we'll move on to tweaking. But first, hit save, or go Options / Global / Auto-Save and Versioning, and set Sonar to auto-save every few minutes or after a particular number of changes. You can also tick the Enable Versioning box if you want to save multiple consecutive versions of the song, as opposed to just overwriting your existing version with each save.

Using the TTS1's chorus and reverb can add more character to sounds, particularly because these effects sound different compared to the other ones bundled with Sonar.

Tweaq The Seq

Once you've created the beat, it's time for fine-tuning. In the step sequencer, you can fine-tune velocity values on a step in three ways: double-click on a step and enter a new velocity value in the numerical field that appears; click and edit using the mouse scroll-wheel; or click on a step while holding Shift, then drag up or down to increase or decrease velocity respectively. For individual rows, you can add a Velocity Offset (for example, add 27 so that the default values of ▶



The original step sequence filled only the first two measures. Here, the Groove Clip it generated has been rolled out to fill 16 measures.

100 turn into 127) or Multiplier (for example, multiply all velocity values by 1.2) in the fields to the immediate left of the note grid. Finally, a little bit of swing can make a pattern more appealing, at least to my ears. In the group of Style controls toward the top of the Step Sequencer, set Swing to around 55.

Don't overlook the fact that you can edit the TTS1 drum sounds themselves. In TTS1, click on the small square at the top of channel 10 (or wherever you placed the drums). This opens up an edit window for that channel, where, towards the centre of the window, you can select the sound to be edited and adjust Level, Pan, Coarse and Fine Tune, Reverb Level and Chorus Level (see screenshot at the start of the article).

Select the sound to edit with the big right/left arrows, or if you click on the MIDI Edit button the sound will follow any incoming MIDI note. (Curiously, notes are indicated as an octave lower than on the step sequencer. For example, C3 in the Step Sequencer shows as C2 in the edit window.) Towards the bottom of the window you'll find Bass, Mid and Treble controls, along with a Filter that can add resonant effects.

Remember to click the On/Off button to the right of the Tone controls to make these effects active. Finally, above the window's level fader (any level changes are reflected on TTS1's main mixer view), there's a master pan control, as well as master send controls for the Chorus and Reverb.

You might be wondering where you edit the Chorus and Reverb: go to the TTS1 main mixer view, in the upper right, and click on Effect. This brings up a window offering Chorus and Reverb options, including chorus and reverb types, reverb decay time and several chorus-related parameters (see screenshot on previous page).

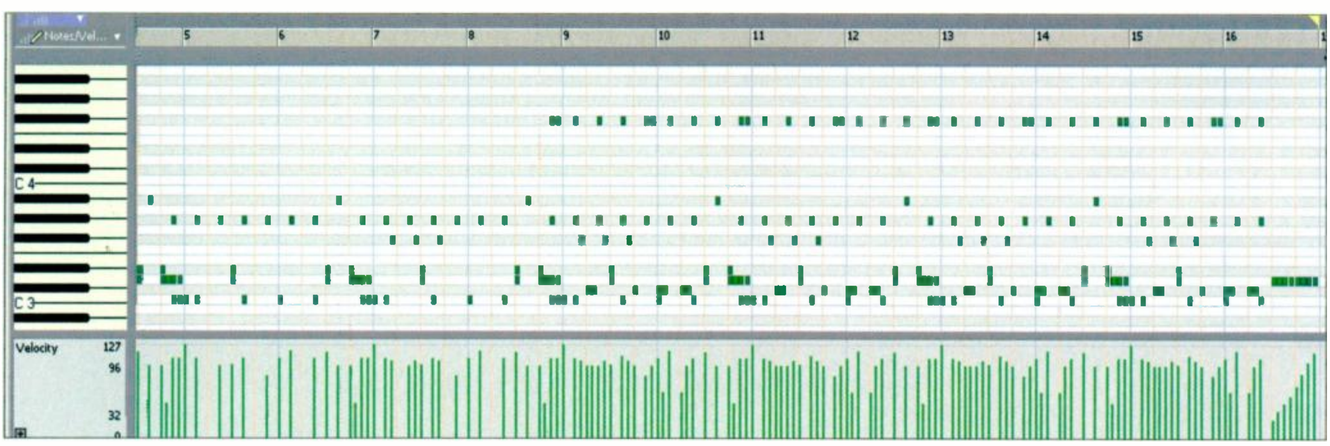
For electro-type drum patterns, try selecting Small Room for the reverb type, and set the time parameter to around 45-60. Turn up the master send control, either in the instrument edit window or on the main mixer view, and the sound will blossom into a bigger, deeper one with more character.

While chorus may not seem all that useful for beats, you can get a very electronic, resonant-sounding effect by selecting Flanger as the chorus type, turning Rate and Depth to zero and setting Feedback above 100. This is where the individual send controls for the different drums really come in handy, as you probably wouldn't want to add this resonant effect to the entire drum part. I use it most often on the kick, and on 'natural' sounds (like tambourine) to make them sound more electronic. Note that it's also possible to send the chorus through the reverb via the Rev Send control, and this can add yet another twist to the sound.

From Beat To Rhythm Track

So far, we've used the step sequencer to layer various sounds and create a two-bar pattern, tweaked the drum parameters for the best sound, and maybe even added some effects within TTS1. Now it's time to take that beat and expand on it.

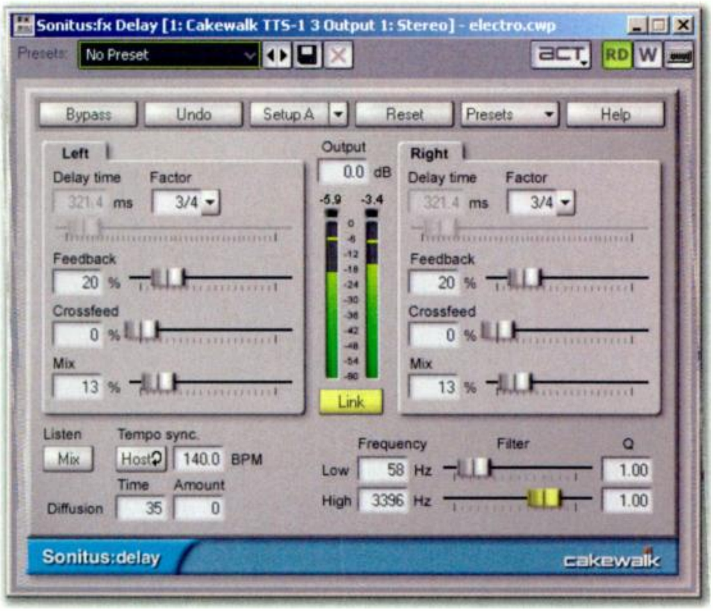
Close the Step Sequencer, and focus on the track view. The Step Sequencer has created a MIDI Groove Clip (as identified by the four rounded corners), which means you can 'roll it out' by clicking on the right edge and dragging it to the right to create multiple iterations of the clip (see screenshot above).



The MIDI Groove Clip has now been converted to a standard MIDI clip, and edited to create a more interesting part that builds over 16 measures.

CREATING A DRUM TRACK

Typical settings for a tempo delay effect. Note how the highs and extreme lows are rolled off a bit to avoid having the echoes 'step on' the main notes.



For example, suppose we want to work on the first 16 measures of the piece; the screenshot shows the original, two-bar clip (grey) rolled out to last 16 measures.

At this point, if you double-click on this clip it will simply re-open the step sequencer, and any change you make to the sequence will be reflected in each iteration of the clip. But what we want is to be able to work on this clip to build the song, by simplifying it towards the beginning and letting it build over time. To 'detach' it from the step sequencer so that we can do this, right-click on the clip and select Bounce to Clip from the context menu, or click on the clip to select it and go Edit / Bounce to Clip. The clip now turns into a standard MIDI clip, and if you double-click on it, the Piano Roll view will appear for editing the clip (assuming you haven't customised this particular shortcut).

As a side note, suppose that after creating this clip you decide that you'd really rather work with it as a 16-measure step sequence in the Step Sequencer. No problem: right-click on the clip, and select Convert MIDI Clip(s) to Step Sequencer. If you don't see this in the context menu, click on the row of arrows along the bottom to open up a 'side' menu with other options. You'll be asked what step resolution you want to use, which will probably be 16th notes if that was your original Step Sequencer resolution.

For now, we'll assume you want to treat it as a MIDI clip, so let's make it easy to edit. First off, it's useful to shorten the note durations a bit. Why? Suppose the Step Sequencer generates four consecutive 16th notes. They'll each be exactly one 16th note in length, so you won't really be able to see the note start and end if you're zoomed out. So select all the notes in the track, go Process / Length, tick only the 'Duration' tick-box, enter 70 in the Percent field, then click on OK. This adds a little space between consecutive notes, making it easier to see where they begin and end.

Now start carving. In this example, I got rid of the tambourine for most of the first eight measures, leaving in a short pick-up going into measure nine. I also deleted the low clave sound for these measures. Thus, for the first eight measures I had only kick, snare and open/closed hi-hat, with the second eight measures picking up in complexity. Then, for the last half of the 16th measure, I deleted everything except for a series of snare-drum hits designed to add a big variation before the next part of the song. The screenshot at the bottom of the previous page shows the result. Note that the part is sparser at the beginning, gets more complex, then ends with the snare roll (with increasing velocities).

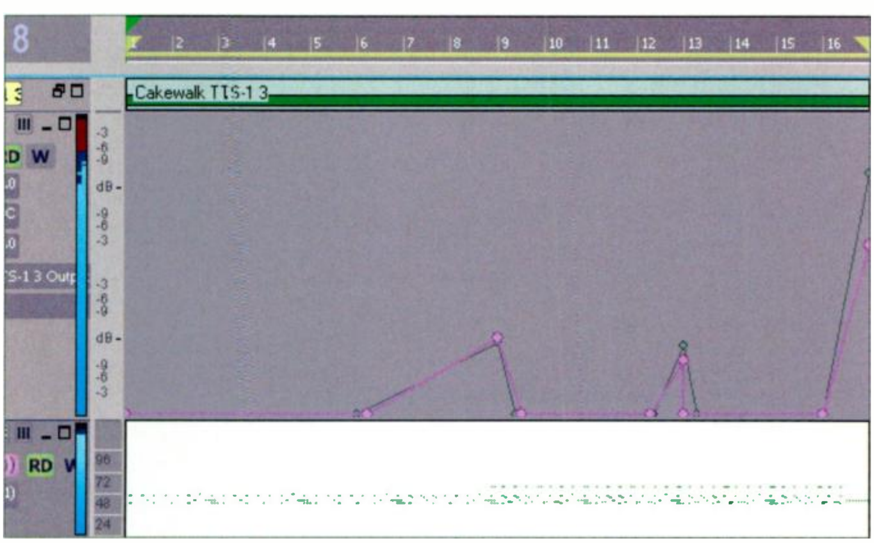
Send In The Audio

Now we're getting somewhere, having taken our simple beat and built it into a much longer part, with some editing to add more interest. However, remember that in Sonar a virtual instrument feeds the equivalent of an audio track — so we can put effects in the audio track's effects bin and further alter the sound. What's more, thanks to automation, we can create variations in the effects.

As an example, one of my favorite effects is synchronised delay, which the Sonitus Delay (bundled with Sonar) does very well. The 3/4 factor setting works well for dance tracks; the screen above shows typical parameter settings to add a rhythmic delay. However, having this delay on constantly can get irritating, so let's automate the delay mix to bring it in and out at strategic times.

1. Right-click in the effects bin for the track where you want to add delay, and go Audio FX / Sonitus:fx / Delay.
2. Right-click in the audio track containing the Delay plug-in.
3. From the context menu, go Envelopes / Create Track Envelope / Delay.
4. A window appears with all available delay parameters for automation. Tick 'MixL' and 'MixR'. If you like, you can choose a colour for the automation curve.
5. Click on OK.
6. Lines will now appear in the track. Double-click on the line to create a node that can be dragged anywhere you want. Drag up to increase the level of delay in the mix and drag down to reduce.

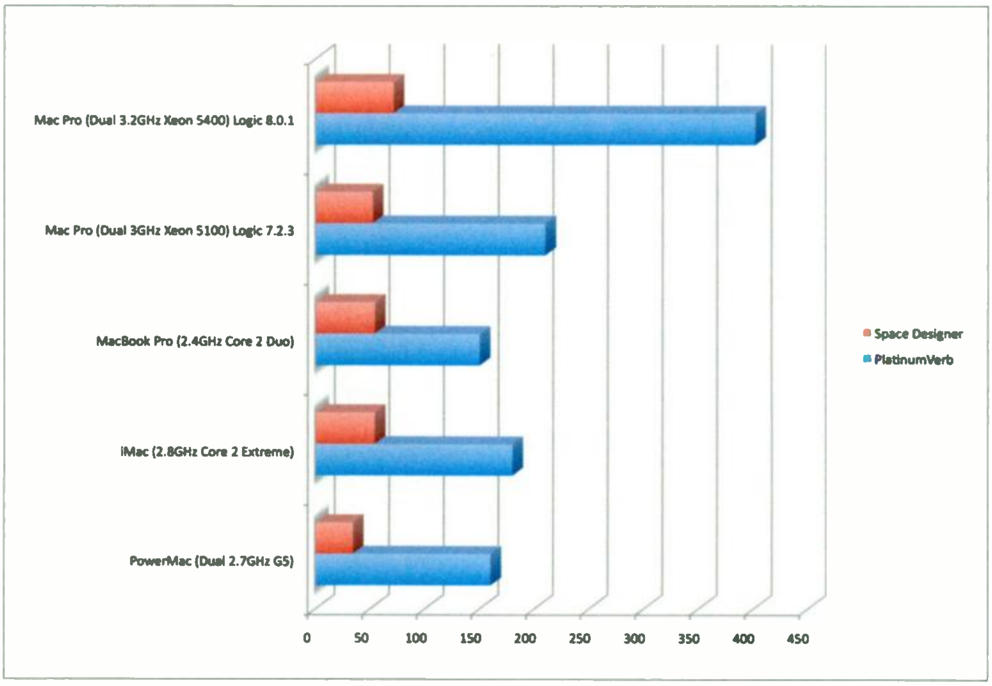
At this point, we've gone from a simple two-measure drum pattern to the start of a drum track. We can copy what we have and use it elsewhere in the song or, better still, come up with variations on a theme. And then comes the bass... but that will have to wait for another time! SOS



The two lines in the audio track are controlling the left and right mix levels for the delay plug-in that's processing the drum track. The level of the delay increases just before going into measure nine, and again towards the end. There's also a slight automation 'bump' towards the middle.

The current top-end Mac Pro offers so much power that some audio software is playing catch-up just to be able to make use of it all. But how much power? We run some tests to find out.

Mark Wherry



Here you can see how the new Mac Pro performs relative to other Mac systems.

It's been almost two years since Apple replaced the PowerMac G5 with the Intel-based Mac Pro, and as Apple's high-end desktop system the Mac Pro continues the PowerMac's legacy as the system of choice for musicians and audio engineers requiring the ultimate performance. While its chassis remains the same as the PowerMac G5 when introduced in 2003 — disappointingly unsuitable for easily mounting in a rack — Apple continue to provide the latest components on the inside, making the Mac Pro one of the most capable workstations available, no matter whether you run Mac OS X or Windows as your operating system.

To 3.2GHz & Beyond...

We briefly covered the introduction of Apple's latest range of Mac Pros, which make use of Intel's latest Xeon processors, in March's Apple Notes (www.soundonsound.com/sos/mar08/articles/applenotes_0308.htm). Available with clock speeds of up to 3.2GHz, among the highlights of these new chips is that each processor has a 12MB Level-2 cache and operates on a 1600MHz system bus, which ought to provide fairly good performance for power-hungry music and audio software. To find out for sure, this month I got my hands on a top-of-the range, dual quad-core 3.2GHz Mac Pro with 16GB memory, Apple's optional SAS RAID card, and an Nvidia GeForce 8800 GT graphics card with 512MB memory.

To begin with, having 16GB of memory installed in the Mac Pro is probably overkill for current music and audio applications, all of which are still 32-bit, meaning that the maximum amount of memory accessible to each is around 3GB. Even taking into account the fact that Logic Pro 8's EXS24 sampler can now address more than this allocation, for pre-loading samples that are to be streamed from disk using the plug-in's Virtual Memory feature, 8GB would probably be a more sensible amount in terms of price and performance.

If you use your Mac for other, non-musical tasks, such as running virtual machines or a number of other, memory-hungry applications simultaneously, more RAM is always a bonus. And if you really want a huge amount of memory, it's now possible to install up to 32GB, which will admittedly add £5760 to the cost of your Mac Pro from the Apple Store, which comes with 2GB as standard.

Testing Times

To test the performance of the new 3.2GHz processors, I installed Logic Pro 8.0.1 (the Mac was running OS 10.5.2) and ran the usual assortment of reverb and instrument performance tests using the built-in audio hardware with a buffer size of 512 samples.

Starting with PlatinumVerb, I was able to run 402 simultaneous instances, which is quite an improvement over the 210 instances possible with the original 5100-series quad-core 3.0GHz Mac Pro reviewed back in December 2006 (www.soundonsound.com/sos/dec06/articles/macpro.htm). Using the stereo version of Space Designer, I managed to run 71 instances, compared to 52 instances on the earlier 3GHz Mac Pro.

Moving along to EXS24, using the old favourite VSL Harp instrument, with the Original Storage setting and the filter and Virtual Memory disabled, I was able to play back 2176 stereo voices simultaneously (compared to 1856 voices on the earlier Mac Pro). However, enabling 32-bit storage caused this number to rise to 3840 voices, which isn't bad (over the earlier 3200 voices). However, it's worth remembering that 32-bit storage requires more memory than storing the samples in the 'original' 16- or 24-bit format, and to achieve this level of performance you have to disable Virtual Memory, which, as mentioned above, has to be enabled for the EXS24 to see more of your Mac's memory.

Rather than only focusing on Logic Pro, I also looked at the performance of Steinberg's Nuendo 4.1, since in the original Mac Pro review we also tested the performance of RoomWorks in Cubase 4. Using the RoomWorks plug-in, I was able to run 119 instances simultaneously, versus the 66 instances possible on the original Mac Pro.

Multicore Confusion

These numbers seem very impressive, but it's interesting to note that while most of the time computer hardware has to catch up to the demands of what a user wants to achieve with software, we're now reaching a point where the software's having to catch up with the hardware. Let me elaborate.

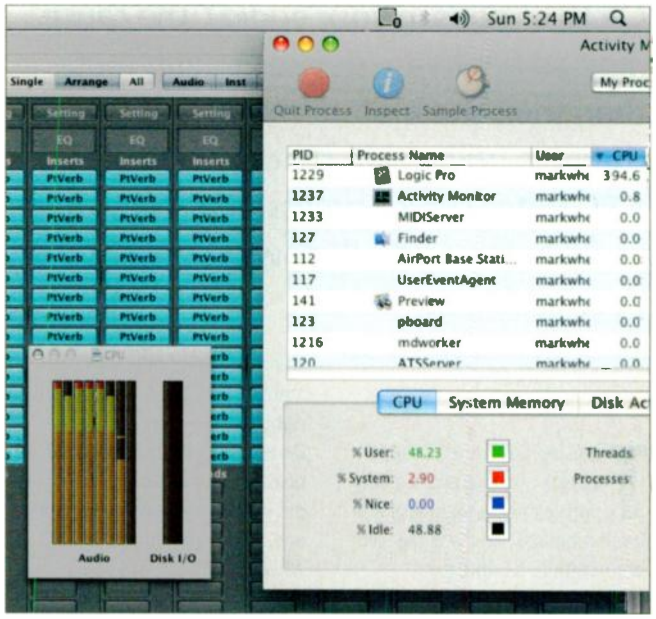
You might have noticed that I've left out a set of numbers normally included with the performance benchmarks: the Process and User CPU usage percentages reported by the Activity Monitor utility. The Process percentage indicates how much of the available CPU resources are taken up by a given application, such as Logic Pro or Nuendo. With this value, 100 percent equals one processing core's worth of resources, so an application running on an eight-core computer could theoretically use a maximum of 800 percent. The User CPU percentage reports how much of the CPU resource is being utilised by Processes (or applications) that have been launched by the user. With this figure, 100 percent represents the total amount of CPU resources available.

Going back to the PlatinumVerb test, 402 instances ran with Logic Pro, using 394 percent CPU and 48 percent User, while in the RoomWorks test 119 instances ran with Nuendo, using 633 percent CPU and 78 percent User. If you compare these figures, you'll notice that Nuendo is clearly utilising more of the available resources than Logic Pro. In a perfect world, you would hope for a linear improvement in resources, where two cores would run twice as many tasks as a single core, four cores would run double the number of tasks as two cores, and so on. But it isn't a perfect world, and the reality is that there's both a limit to how tasks can be split across multiple cores, depending on how complex the audio routing is for your Project, and an overhead in the computer and application having to schedule tasks for multiple cores.

Taking these factors into consideration, I think Nuendo does a pretty good job. However, as has been mentioned many times in the past, it seems as though Logic's audio engine would benefit from some improvement to its handling of multiple cores. Which is, of course, easier said than done.

Coming Soon

Aside from the processor and memory improvements in the Mac Pro, the storage system has also been improved, such that you can now order it with Apple's RAID card, which was previously only available with an Xserve. Using this card, you can now configure a disk array in your Mac Pro that works at a hardware level, for better performance compared to using OS X's built-in software RAID functionality, and supports SAS (Serial Attached SCSI) drives instead of SATA, which also offer improved performance. In next month's Apple Notes we'll be taking a closer look at the new storage options available in the Mac Pro and asking whether the additional cost of the RAID card and SAS drives is worth the extra performance yield when working with audio. SOS



Activity Monitor is a useful way of checking how much of your Mac's resources your applications are making use of, even when such applications report a system overload. Logic Pro's use of multiple cores is somewhat quirky, as you can see here. While all eight cores are displayed, notice how in this situation the eighth core itself is apparently completely free, even though the Project is rather busy.

Apple Launch Final Cut Server

As with this year's NAMM show, Apple declined to exhibit at the 2008 NAB (National Association of Broadcasters) show, held annually in Las Vegas for the broadcast and video industry. In recent years Apple have maintained a significant presence at the show, using it as a platform to announce products such as Final Cut Studio and the 17-inch MacBook Pro. But this year the company feel there are better ways to reach customers, with initiatives such as the Final Cut World Tour, a series of seminars held in various cities around the world to demonstrate the potential of the Final Cut Studio workflow. See www.apple.com/finalcutstudio/tour for details (but note that London wasn't on the itinerary at the time of writing, although it may have been added later).

Despite not exhibiting at NAB, Apple did release Final Cut Server, originally announced at last year's NAB show, at the same time as this year's exhibition. As the name suggests, Final Cut Server is a server software application for Final Cut Studio users that facilitates both media asset management and workflow automation amongst multiple users. What this means, for example, is that if you have an editor working on a project in Final Cut Pro, Final Cut Server makes it easy for him to send a clip from that project to an artist working in Motion (or perhaps an audio engineer working in Soundtrack Pro), and then have the Final Cut project automatically updated when the artist is finished.

Final Cut Server can work over a SAN (Storage Area Network) environment, such as Apple's own Xsan software, where all of your video assets stay on the SAN and play back directly on your desktop system. Alternatively, you can opt to have a version of the project copied directly to your computer and then copy any changes back to the server later.

I managed to see a demonstration of Final Cut Server this month, and it seems like a promising system, adding functionality that was sorely missed in the Final Cut ecosystem, with Apple once again doing a great job of putting a user-friendly interface on a potentially complex application. What's particularly interesting from a technical perspective is that the client-side application is written using Java (as opposed to Apple's own Cocoa framework), since it can also run on Windows-based systems, and this may be the best-looking Java application ever written! While the client is cross-platform and effectively freely distributable, the server application that manages data and provides the core functionality is Mac only and is available in two versions: a 10-client version for £699 and an unlimited client version for £1399.

At the moment, Final Cut Server is obviously aimed at Final Cut Studio users, and although the system can work with any type of file, including audio files or any other type used by music or audio software, only Final Cut Pro files are handled natively (in the sense that the system can know exactly all of the assets used by a given project). It's also worth bearing in mind that such asset-management systems are designed for situations where you have multiple people working on different aspects of the same project. While this type of workflow is fairly common in video post-production, it doesn't happen in quite the same way in audio, even for larger facilities. That said, it will be interesting to see how Apple develop the software in the future, especially as Soundtrack Pro continues to address the needs of the audio post-production market.

Clicks or pops in your audio? The cause may be hardware devices taking more than their share of interrupt time. Now there's a utility that checks for this, and a PC Notes survey amassing results that could help you track down the source of your problems.

Martin Walker

Tracking down the possible causes of audio clicks and pops can be a thankless and frustrating task, so I was pleased last month to be able to recommend an easy-to-use utility for highlighting the presence of any rogue hardware device that occasionally takes more than its fair share of interrupt time (a classic cause of audio interruptions).

Thesycon's DPC Latency Checker (www.thesycon.de/eng/latency_check.shtml) has subsequently proved its worth many times over, especially once I'd started a 'DPC Latency Survey' on the *SOS* forums (www.soundonsound.com/forum/showflat.php?Cat=&Number=58 8140) and encouraged readers to submit their findings. In just three weeks the survey attracted over 100 posts and 4000 views, and there's already a lot to take in. Let's summarise the results so far.

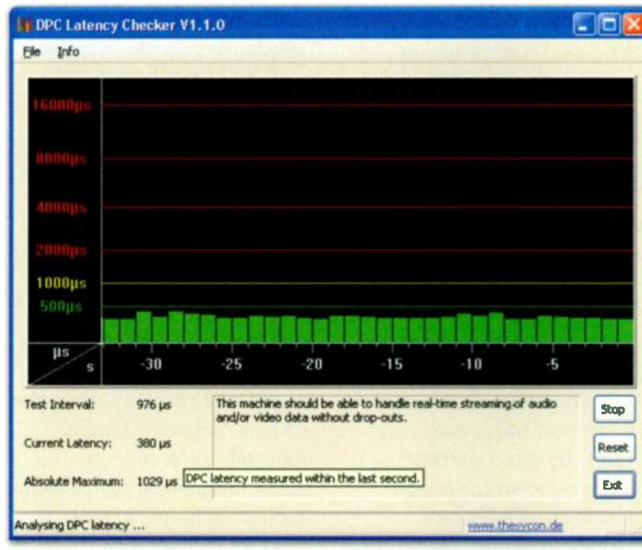
DPC Latency Survey Results

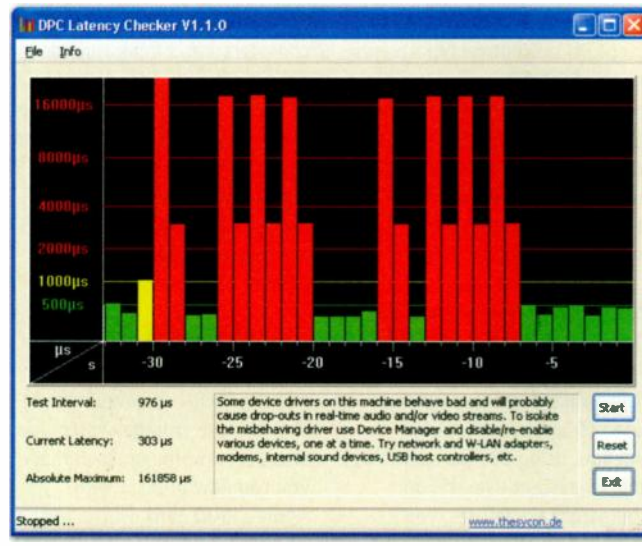
DPC Latency Checker takes one measurement each second, providing a readout of current latency (which, on most PCs, is a relatively constant figure) plus the absolute maximum value recorded since you first launched the utility, which may occasionally spike to a considerably higher value on some systems, because of one specific hardware device.

The reason for such spikes needs to be tracked down to avoid possible audio interruptions (especially if you're using small audio buffer sizes). I've found the most effective approach after discovering spikes is using Device Manager in real time: select a possible rogue device in its list, right-click on it and select the Disable option. If the spikes are unaltered, you then re-enable the device and move onto the next one, until you find the culprit. When you find it, possible cures include updating its drivers, disabling the device each time you want to make some music, or creating a dual-boot system with it permanently disabled.

To get the ball rolling, I posted some baseline figures from my own well-behaved Intel Conroe E6600 2.4GHz dual-core PC running Windows XP with Service Pack 2, which typically measured 50 microseconds, with an occasional peak of up to 100 microseconds on my Internet-enabled partition. There was a smaller typical value of 24 microseconds, with an occasional peak up to 65 microseconds, for my music-only partition.

A hardware device needs to hang on to the interrupt for longer than 500 microseconds to enter the borderline area for audio interruptions, but these results do demonstrate that a stripped down, music-only PC is more likely to manage small buffer sizes (and thus lower audio latency) without audio glitching.





These DPC Latency results from an *SOS* forum user illustrate how enabling Wi-fi on a laptop PC can cripple audio performance.

Network & Wireless Spikes

Quite a few musicians have already discovered that disabling network adaptors (aka network cards, LAN adaptors or NICs), and particularly wireless (WLAN) versions, may cure audio clicks and pops, but DPC Latency Checker makes it far easier to spot possible symptoms early on. For instance, in this survey 'Phat Riffioso' discovered a regular peak of 405 microseconds every 15 seconds on his Asus P5WDH motherboard, which he traced to polling of an unused Ethernet port to check if any device had been plugged into it. The cure was either to disable the Ethernet port in Device Manager, or connect a device to it. 'Timo' also reduced his results from 100-440 typical and 550 maximum to 14-25 typical and 100 maximum just by

BIOS Beware!

Following last month's revelation in PC Notes that Intel BIOS updates could significantly improve audio performance at low latency with quad-core and octo-core PCs, the *SOS* forum's 'HL' reported that his Gigabyte P35DS4 v1.1 motherboard had lots of audio timing problems and stuck notes running Cubase with BIOS versions F7 to F11, and DPC Latency Checker reported regular peaks of at least 695 microseconds. Downgrading to BIOS version F4 or F5 resulted in typical values of just 30 to 40 microseconds.

Over the years, my advice on BIOS updates has always been to leave well alone unless the update specifically solves a problem you've been having, because if you ever had a power cut before the update was complete your PC might not be able to boot up at all afterwards. This new evidence provides a further reason to be cautious, so I would now, in addition, recommend that you run DPC Latency Checker before any BIOS update and again immediately afterwards. Don't be afraid to downgrade a BIOS if using the newer one results in latency spikes that weren't there before

disabling his onboard Realtek RTL8169/8110 Gigabit Ethernet, using Device Manager.

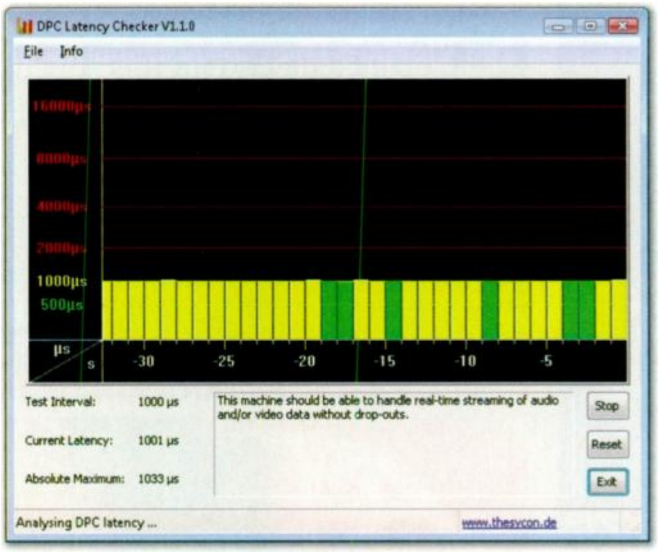
Another interesting finding from Timo was that moving his wireless mouse doubled his lower results to 30-130 microseconds typical and 226 maximum. Back in PC Notes August 2006 I reported on the Cursor XP mouse-pointer enhancement utility from Stardock (www.stardock.com), which managed to stop one musician's audio every time he moved his mouse, and started it again immediately he stopped moving it. So if you're replacing a hard-wired keyboard or mouse with a wireless model, or installing custom mouse or keyboard drivers, I now recommend using DPC Latency Checker before and after to check for spikes, as well as making an image file of your Windows partition so that you can easily backtrack if necessary.

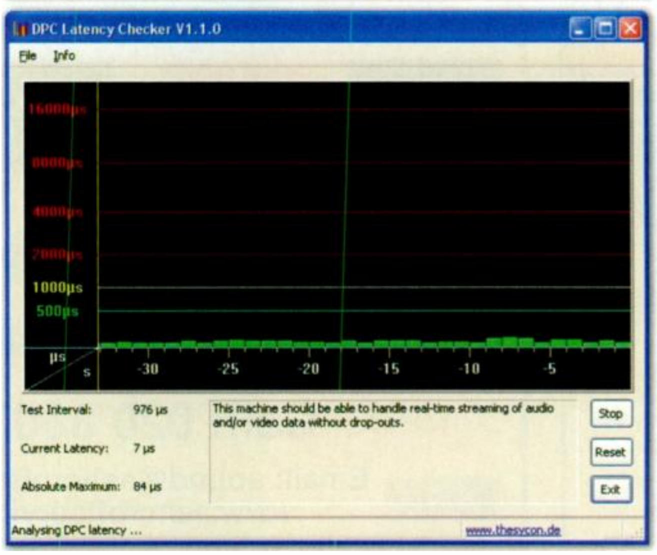
Nobbled Notebooks

'Lachris' reported high average values of 200 microseconds on his Dell Precision M6300 notebook, with peaks of up to 2000 every 15 seconds, resulting in regular audio drop-outs, even at low CPU loads and with high buffer-size settings. These figures were subsequently confirmed by Dell on this model and their Latitude D630, and so far the only cure seems to be to buy a different laptop!

'Nuno' managed to get some good figures for his Dell notebook, but only after installing and configuring the third-party i8kfanGUI utility (www.diefer.de/i8kfan/index.html), which lets owners of certain Dell Inspiron, Latitude, Precision and Smartstep notebook models take manual control of their cooling fans to prevent real-time changes that cause audio drop-outs. Changes in fan rotation speed or CPU clock speed can prove an audio nightmare with some computer models (mostly notebooks). He also reported bad audio glitching on his particular notebook after installing M-Audio's Midisport 8x8 MIDI interface drivers, which was only resolved after a reformat and clean install of Windows.

Once again, the moral is that before you install any new hardware drivers you should make a disk image file, run DPC Latency Checker for a few seconds to gauge its typical and peak results, then install your new hardware and check that the results haven't got significantly worse. If they have, and you're sure you have the latest drivers, at least you can restore the image.





Another forum user posted these figures showing the huge difference in DPC Latency between his Dell E520 desktop PC running Windows Vista and running XP.

Hasta La Vista

Continuing with Dell-related submissions, 'danf' posted the results from his Dell XPS M1330 notebook running Windows Vista (typical 100/200, but occasional spikes to 1000+). He managed to reduce these figures to between 60 and 70, with occasional spikes to 220, simply by installing Windows XP instead. Danny Bullo got an even bigger improvement with his HP Pavilion dv9500t laptop, which gave spikes of up to 3000 microseconds with Vista, but a very low 50-microsecond peak with XP! 'Muied Lumens' then switched to XP after a year of tweaking Vista, reducing almost continuous 1000-microsecond readings to a mere 12-15, with peaks of just 100 microseconds.

While we can't actually conclude that Windows Vista itself is to blame for all these high readings (hardware drivers and, in particular, graphics-device drivers are currently under suspicion) the fact remains that by switching to Windows XP these three musicians solved their audio problems. Perhaps Vista performance will simply leap ahead when more optimised hardware drivers have been developed for certain rogue devices.

DPC Latency Checker may also help us identify problems caused by rogue Firewire controller chips and badly written audio interface drivers, so please report anything unusual you find! SOS

US Court Allows Vista Lawsuit

A class-action lawsuit against Microsoft over the sale of PCs bearing 'Vista-capable' stickers has finally been approved to go forward by a US District Court judge, after customers were mistakenly encouraged to believe that PCs they bought during the Christmas 2006 holiday period could be easily upgraded to Vista when it shipped a month later. In fact, these PCs would only perform well with Vista Home Basic, which lacks many features, including the Aero Glass interface, Flip3D window switcher and Media Center.

However, of more general interest is a 158-page bundle of associated internal Microsoft emails that the same judge has recently been ordered unsealed as part of this case. These include reports that almost 30 percent of all early Microsoft Vista crashes logged by Microsoft were caused by Nvidia drivers. Microsoft themselves were in second place with nearly 18 percent. Most of these driver issues have since been solved, but this information once again underlines how driver performance can bring an operating system to its knees.

Other sobering figures for Vista can be found among the results of a recently published report from Forrester Research (www.forrester.com) covering 50,000 users across more than 2300 large enterprises, which indicate that during 2007 Windows XP was still being used by around 90 percent of businesses, with just 6.3 percent of users moving over to Vista, while Mac users have risen from 1.2 to 4.2 percent during the same period. With Windows 7 rumoured to have a release date in the second half of 2009, many businesses may already be considering bypassing Vista altogether.

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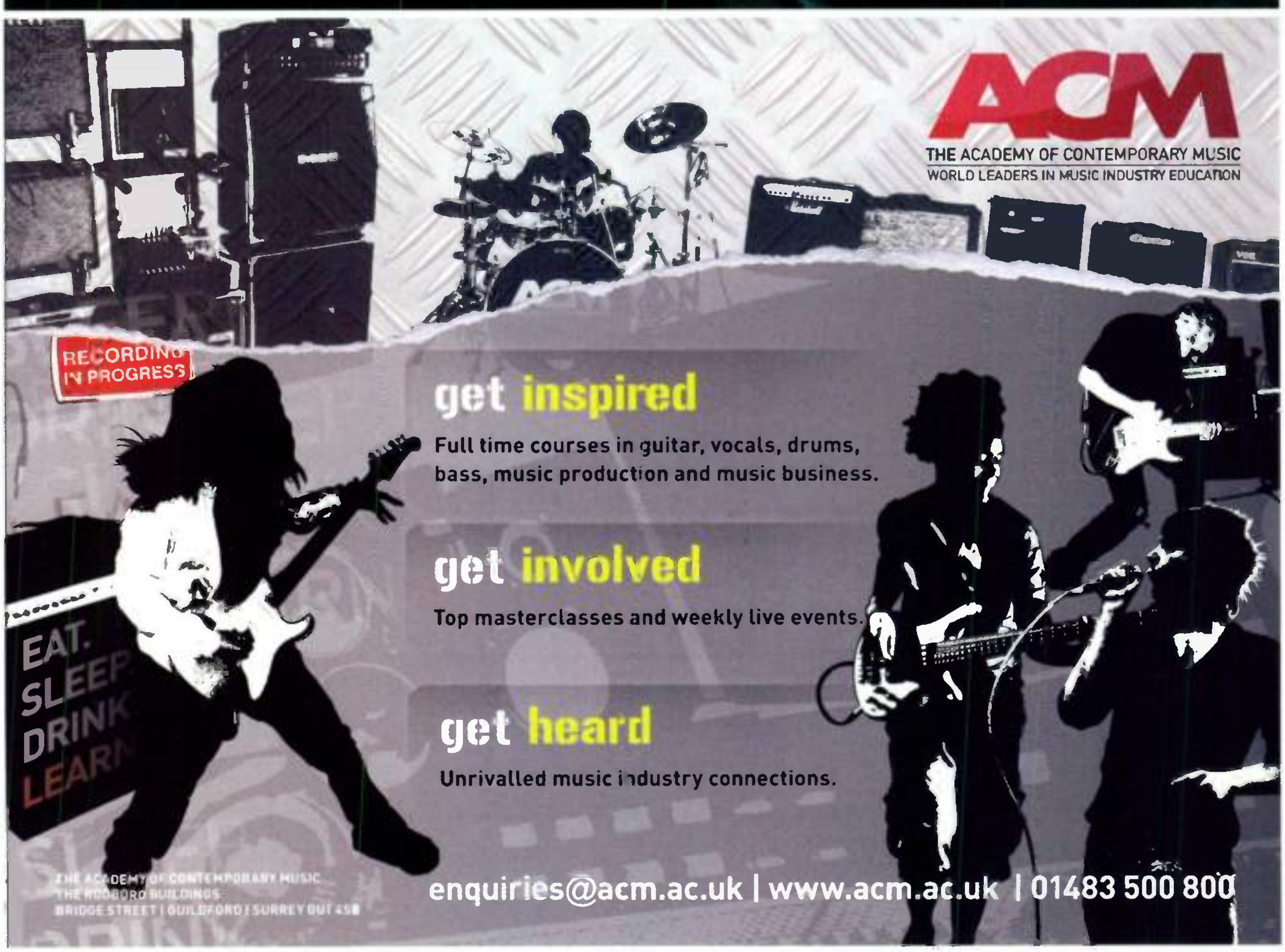
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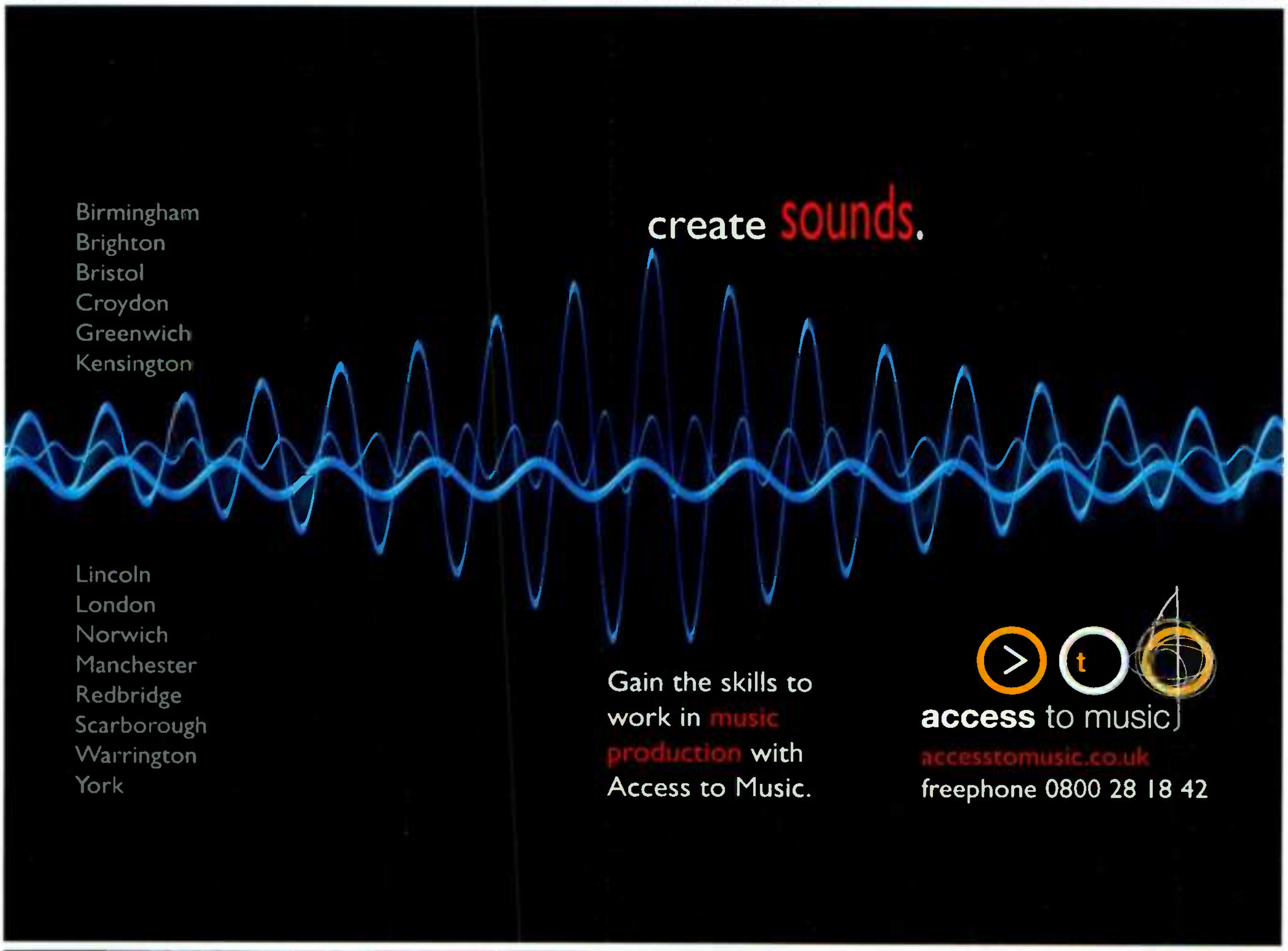

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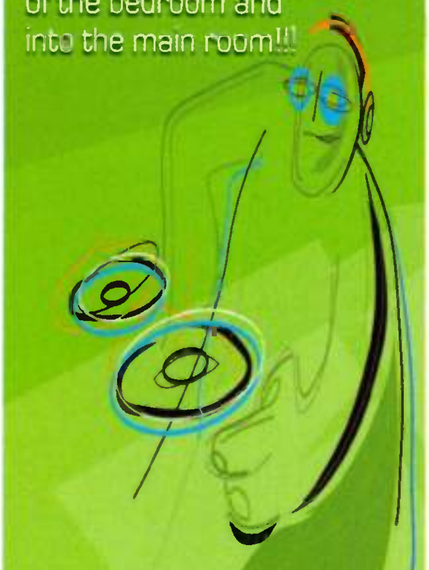

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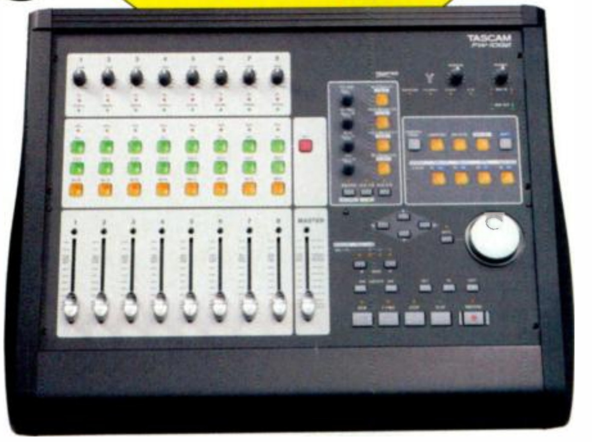

Thanks to Tascam for supplying these great prizes.
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As one of the best-known brands in the audio equipment industry, Tascam have become synonymous with music recording, and their gear is in use by home recordists and studio professionals worldwide. For many people, the Tascam Portastudio, launched in 1979, was the chosen tool for their first ventures into recording, and derivations of it are still available to this day. In professional circles, the company's open-reel and digital tape multitrackers have been faithfully capturing every second of audio for years, and remain in widespread use.

These days, with DAW systems in use for so much recording, Tascam have refocused their product range, launching a new wave of devices aimed at the computer-based music producer. These include a selection of audio interfaces, some of which are prizes in this month's competition.

We've got four separate Tascam audio interfaces to give away. First up, there's the US122L, a two-channel USB 2 device that usually retails for £119. Despite its diminutive size, it has a pair of phantom-powered microphone preamps and MIDI input and output, and can record audio at rates up to 24-bit/96kHz.

The second prize is the US144, which is an upgraded version of the US122L, with which it shares more than a few resemblances. In addition to the feature set of the 122, the 144 has a stereo digital S/PDIF bus, enabling a total of four channels of simultaneous I/O.

The third prize is Tascam's latest USB interface, the US1641, a rackmountable 16-input, four-output device that's reviewed on page 128 of this issue of *SOS*. Interestingly, and unlike many of its competitors, it has 14 analogue inputs, meaning that it doesn't have to rely on an additional A-D converter to get more than eight analogue channels into the connected DAW.

The fourth and final prize in this month's competition is the FW1082, a Firewire interface that's also got control-surface functionality built in. Nine touch-sensitive motorised faders, as well as mute, solo and select buttons, enable hands-on control over DAW software. A jog and shuttle wheel sits above the transport controls, alongside quick keys that can open control panels in software at the touch of a button. The FW1082 has eight analogue inputs (four of which have mic preamps) and two analogue outputs, as well as S/PDIF I/O and twin MIDI Ins and Outs.

If you would like to be in with a chance of winning one of these prizes, fill out the form at the bottom of this page and post it to the address on the coupon, or enter via the *SOS* web site. Please make sure you answer all the questions and complete the tie-breaker. We also require your address, including postcode, and your daytime telephone number, so that we can contact you if you win. The closing date for entries is **June 30th, 2008.** SOS

questions

When was the Tascam Portastudio first launched?
a. 1960 ☐
b. 1979 ☐
c. 2001 ☐
d. 2008 ☐

What type of connection does the Tascam US122L use to interface with a computer?
a. Firewire ☐
b. USB ☐
c. Velcro ☐
d. A proprietary PCI-based system ☐

In total, how many prizes are up for grabs this month?
a. One ☐
b. Two ☐
c. Three ☐
d. Four ☐

Tascam tie-breaker

You are in a room. On a table at the other side of the room are some lovely Tascam audio interfaces. Between you and the interfaces is a bear. The bear is cross. What do you do? Answers in 30 words or fewer, please.

....

....

....

....

Name ....

Address ....

....

Daytime tel. no: ....

Email: ....

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

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

the small print

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

Did eliminating noise throw out the baby with the bath water?

Tom Flint

If you regularly read product reviews, then the chances are that you can often be put off making purchases by certain key phrases. For me, when I read that a product has a 'clean sound' I frequently feel less enthusiastic about it — even though the reference is usually intended as praise. It's not that I'm a lo-fi nut, because I do covet high-quality gear. However, in my book, clean doesn't necessarily mean good.

In fact, in many respects, the less noisy and flawed my recordings have become, the worse they sound! Many people have commented that my recent tracks are very clean and that everything is well recorded. They may well be right, but those attributes are also problematic.

In *SOS* March 2007, Giles Martin commented that "Tape seems to join everything together in the way that digital gives you separation". When I first began tape-bouncing I longed for less hiss and less quality loss. The same was true when I got my first four-track cassette and eight-track reel-to-reel machines. Now, though, I find myself wanting more noise, more hiss, more distortion and more tape limiting, and I don't always want to hear all the instruments clearly either! I want things to bond together. Absolute clarity is not the answer, but neither is it restorative to bung on a load of distortion or tape-emulation plug-ins afterwards.

I refer you to the *SOS* interview with producer Joe Boyd (www.soundonsound.com/sos/jun06/articles/joeboyd.htm), in which he states: "To me, still, if I listen back to the records of the '60s, [*The Doors*'] *Strange Days* is the best-sounding record of the '60s. It's just an amazing job of recording". I'd always loved *Strange Days*, but hadn't singled it out in that way. However, after listening to it again, I began to realise what Joe was getting at.

The instruments on *Strange Days* are clear, yet they blend into the overall ambience, and even the sounds that seem really loud and powerful are actually quite soft. The album was obviously recorded extremely well, using some great equipment and brilliant musicians, and in lovely-sounding acoustic spaces. Nevertheless, it was made in an age when the technology set limitations that seem almost intolerable today.

Back then, multitracking was in its infancy, and many tracks were largely recorded live using careful mic placement. Distortion was inherent in the valves of overdriven amps and mics. There would have been oodles of spill, and plenty of un-editable hiss generated by all those valve amps, mics and early recording desks. Vocal effects came from dirty-sounding EMT plate reverbs. There was also the considerable inefficiency of tape, distorting and compressing audio, requiring noise-reduction systems that audibly affected the sound of the material.

It must have been a real pain in the arse fixing all of that temperamental equipment, positioning everything to minimise noise and spill, and then having to add copious amounts of high-end after each bounce, to regain clarity. You can't blame anyone for wanting more editing options (the lack of which made it necessary to get everything right in the room), more stability and clarity, or an undo button! Still, the equipment offered a winning combination of high quality coupled with bags of ear-friendly noise, hiss and distortion. This helped everything bond together, resulting in recordings that demonstrate a natural depth and homogeneity that is almost impossible to fake.

You don't have to look far to find examples of top professionals who still record bands live in a room, or enjoy the benefits of imperfect gear. I immediately think of Andy Green, producer of Keane's *Hopes and Fears*, who recorded their drums to analogue two-inch tape with Dolby SR engaged. Then there's the Nimrod team. They recorded much of the music for *24: The Game* using an antique EMT140 plate reverb and, for orchestral ambience, hired Abbey Road studios to record in.

As for me, in my humble home studio, I'm certainly after high quality — but absolute clarity? No! I've far too much of that already. SOS

About The Author

When he isn't writing for music technology publications, Tom Flint co-ordinates the East Anglian Mountain Rescue Team from his gothic castle on the edge of a precipice in Norwich.

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.

Next Month in *Sound On Sound...*

Classic Tracks

Devo's 1980 hit single 'Whip It' has been the blueprint for countless electro tracks. Producer Robert Margouleff tells us how it was done in next month's Classic Tracks.

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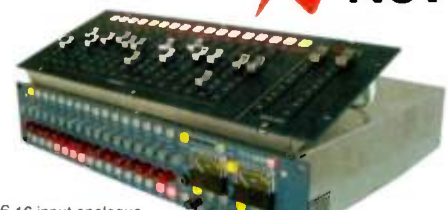

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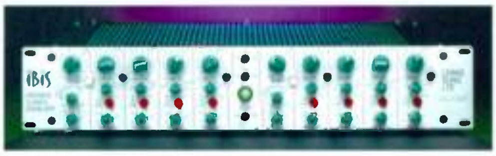

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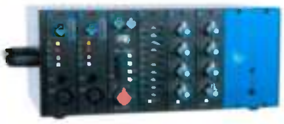

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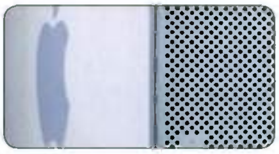
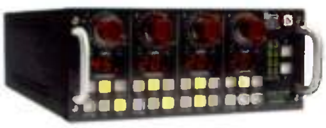


Neve 4081



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