SOUND ON SOUND
The World's Best Music Recording Magazine

August 2005 In This Issue

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controller keyboards for some time, and have recently moved seriously into Firewire interfacing. The new Ozonic seeks to combine both capabilities in one product — and all for less money than you'd think...

Mackie *Tracktion* 2

MIDI + Audio Sequencer [Windows/Mac OSX]

Tracktion's designers have given the program a thorough overhaul and added some significant new features. But have they succeeded in retaining its freshness and ease of use?

Mackie X200

Digital Production Console

Not content with acting as a high-spec control surface and Firewire audio interface for your studio computer, this new high resolution console seems determined to outshine it with its slick touchscreen graphical interface and onboard VST plug-in hosting.

Miglia Harmony Audio

Firewire Audio Interface [Mac OS X/Windows]

A two-in, eight-out interface is the first audio product from British Firewire specialists Miglia Technology.

MXL V6 • 990 • 992

Condenser Microphones

Three new large-diaphragm mics from MXL include a model which attempts to recreate the sound of valves using solid-state technology.

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Plug-in Folder

Mini-reviews of hottest new Plug-ins

Roland FANXUP1 & Fantom Xr

OS Upgrade + v2 Editor For Fantom X

The Fantom X is Roland's best-ever workstation, but it has suffered from one or two annoying omissions, such as the ability to import Roland's own sample format. We explore the Fantom Xr rack module and ask if the v2 OS and editing software provide the solutions?

Roland VS8F3

Plug-in DSP Card For VS-series Multitrackers

This new card narrows the gap between Roland's VS-series machines and computer recording systems by allowing the use of third-party plug-ins within the multitracker environment. We take a look at the card, its bundled plug-ins, and the first of the brand-name offerings from Universal Audio.

Sample Libraries: On Test

Sample Shop

We check out the latest sample libraries on the block...

drummer whose studio requires a monitoring and acoustics makeover...

Technique

Logic v7.1 Stability Issues

Logic Notes

We take a look at some of the stability problems still lurking in v7.1 under Tiger, and also reveal a couple of extra plug-ins tucked away in the new OS upgrade.

CLASSIC TRACKS: The Bee Gees *Stayin' Alive*

Producers: The Bee Gees, Albhy Galuten, Karl Richardson

Disco was an American phenomenon, but its greatest hits were recorded in France by an English band who were trying to play R&B...

Comparing Audio Restoration Plug-ins In *Pro Tools*

Workshop

We line up three of the most popular audio restoration packages available for *Pro Tools*. Which gives the most natural results, which is easiest to use, and how do you get the best from them?

Digital Performer Notes

News & Rumours

More news and ideas from the *DP* scene — as well as some good old-fashioned rumour-mongering...

M for Mobile...

PC Notes

The Pentium M processor has various desirable qualities, not least its very low power requirement, which leads to a reduced need for cooling and a quieter life for musicians. However, it has always been confined to use in laptops, with corresponding limitations — until recently.

Macs With Intel Processors?

Apple Notes

Apple users are used to transitions, having moved from 68k-based Macs to Power PC processors, and the classic Mac OS 9 to Mac OS X. Now it's time for the third and most shocking transition of all: the move to Macs with Intel processors.

Making The Most Of EQ In *Sonar* v4

Workshop

When is an EQ not just an EQ? When it's the *Sonar* v4 Producer Edition Sonitus EQ and you use it for some of the amazing processing tricks we're about to explain!

Practical Groove Quantisation in *Digital Performer*

instruments. But sampling itself has become something of a lost art. In the first part of a short series on rediscovering this skill, we look back at how the technique and the technology developed.

Update: *Cubase SX/SL* v3.1

Cubase Notes

We investigate the major and minor new features on offer in Cubase SX/SL v3.1.

Using The Malström Synth In *Reason*

Workshop

The Malström device can certainly be used as a conventional synth, but behind its slightly bizarre green front panel there's a powerful graintable sound engine begging to be exploited to the full. We show you how...

In this article:

AAS String Studio

Modelled String Instrument

Published in SOS August 2005

Print article : Close window

Reviews : Software

String Studio uses Applied Acoustics' physical modelling expertise to (in theory) generate the sound of any string instrument. Good or bad vibrations? We find out...

Martin Walker

Imagine a magic guitar which can be restrung with anything from two to 64 strings and played with a plectrum, bow, or bouncing hammers at any position along the strings. You can pitch the strings with fingers or add frets, damp them, or shrink or expand the soundboard. You can listen to it acoustically or via a modelled magnetic pickup, and it has built-in multi-effects. You need imagine no longer, since I've been playing this instrument a lot over the last few weeks — it's *String Studio*, from Applied Acoustic Systems.

Panel B contains the various modules that make up this unique physical string model and allow you to interact with its various parameters.

When you've experienced the many and varied sounds that it encompasses, 'String Studio' does seem the perfect moniker. However, it's led some musicians to assume that this product was designed to create hyper-realistic orchestral strings to compete with the various vast multisampled libraries now available. In fact, it's a far more versatile synthesizer that physically models a wide range of sounds generated by strings, such as the hammered strings of acoustic pianos, bouncing hammers of instruments like the dulcimer, the bowed strings of the violin family, and the plucked strings of guitar and pizzicato violin. And that's not all — this new AAS engine is also capable of a wide range of ethereal pads, unusual textures, and out-and-out weirdness.

AAS *String Studio* £110

pros

- Uses a completely new physical model capable of a huge range of superb plucked, bowed, and hammered string sounds.
- Every parameter can be easily mapped to MIDI controllers for expressive real-time performances.
- Huge high-quality bundled library.

cons

- Some of the graphical controls are small and fiddly to use.
- Getting the best-sounding results requires some effort.

summary

If you like stringed instruments, and have a MIDI keyboard, you simply have to try *String Studio*.

information

£ £109.99 including VAT.

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W www.scvlondon.co.uk

W www.applied-acoustics.com

Test Spec

PC REVIEW SYSTEM

- 2.8GHz Pentium 4C PC with

Hyperthreading, an Asus P4P800 Deluxe motherboard and an Intel 865PE chipset, running an 800MHz Front Side Buss, 1GB of DDR400 RAM, and Windows XP (SP2).

- Steinberg *Cubase SX* v3.0.2, Cakewalk *Sonar* v4.
- AAS *String Studio* version reviewed: v1.0.

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Overview

Applied Acoustics Systems have championed physical modelling for some years now, ever since the first release of their *Tassman* software synth in 2000, but with *String Studio*, they've unveiled a completely new acoustic 'engine'. As with their *Lounge Lizard* electric piano, processor overhead remains comparatively low compared to modular designs like *Tassman*, and there's plenty of opportunity to allow the user to interact with the engine in real time using MIDI controllers.

String Studio will run on Mac OS X 10.2 or later with a minimum 733MHz G4 processor, while PC users can use Windows 98SE, 2000, or XP with an 800MHz Pentium III or faster. As usual, if you want to run your sequencer and other software synths alongside, the more powerful your processor is, the better. Installation is easy, but unfortunately you have to enter your serial number and go through the now familiar challenge/response routine on the AAS web site before you can play a single note. *String Studio* can be run in stand-alone mode, or as a plug-in in VSTi, DXi, Audio Units, and RTAS formats inside a suitable host application.

The main display will be familiar to existing AAS customers. There's a strip across the top with options for loading/saving and importing/exporting of presets and preferences, editing functions, and MIDI Link options (more on these later), while the Toolbar beneath it shows the current preset name and MIDI link, maximum polyphony, active MIDI channel, the value of the currently selected control, plus a CPU meter and MIDI activity 'LED'.

The main area is split between the Preset Browser (which can be hidden if preferred) and the main control area. *String Studio* has a lot of controls in this area, but AAS have managed to pack the majority into two very manageable areas labelled Panel A and B, which you switch between by clicking on one of the three illuminated buttons on the main title bar (the third is Compare, which lets you make comparisons to the original sound when editing presets, and there's also a master level control with stereo level meters). Panel A is the default view, and houses two horizontal rows of modules, mostly featuring fairly traditional-looking rotary knobs (see the box on the right for more details on this panel).

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The Bundled Library

In the final couple of months before shipping *String Studio*, a huge amount of effort has gone into the bundled sound library, and it shows. Compared with the embryonic sounds that some *SOS* staff heard at trade shows, the vastly improved presets the software now contains reflect both further refinements in the PM engine, and the increasing adeptness of the designers in using the engine parameters to recreate existing acoustic instruments — and to dream up new ones.

The presets have been carefully organised into folders to make finding a suitable sound easier. As always with AAS products, the Imports folder is at the top, so you can add downloaded presets to the library, while the instruments are contained within a further 17 folders. The first of these, Guided Tour, contains 57 highlights from the other categories, while the others include guitars, basses, clavinets, pianos, harps, and pad and bowed sounds, as well as sound effects. I estimate that there must be about 500 presets in all, covering a lot of sonic ground.

Many of the guitars and basses are truly excellent, and *String Studio* excels at clavinet, clavichord, and harpsichord sounds, complete with the characteristic mute 'choking' as the notes are released. Not surprisingly for an Applied Acoustics product, there are quite a few electric pianos of note, although *Lounge Lizard* is the real expert in this department, but the acoustic pianos are rather weak. However, some of the bowed strings are superb, as are the selection of ethnic instruments such as the Japanese koto and the Middle Eastern dulcimer. The synths are a mixed bunch, but some of the pads and ambient sounds are wonderful. Overall, there are loads of real gems on offer, some highly expressive.

You can select any *String Studio* preset by double-clicking on its name in the browser, but since there are far more on offer than could be chosen from the 128 available MIDI program changes, AAS also provide a Program Change Map to which you can allocate your choice of presets and then switch between them remotely via MIDI program change commands from a keyboard or sequencer.

Here are a few of my favourites, some of which you can hear on this month's free covermount *SOS* DVD.

- 'Jazz Guitar': A touch-sensitive clean sound that's ideal for jazz, country, and 'My Sweet Lord' impersonations.
- 'Hold and Hammer Lead': A screaming lead with hammer-on harmonics, offering a wide expressive range.
- 'Mute Bass': A very expressive instrument, offering a range of tones from round and smooth to hard and 'clicky'.
- 'Pop-Rock Clavinet': If you're feeling funky, a quick burst on this will surely satisfy you.
- 'Violin 4': You can really hear the bite of the bow being scraped across the strings here, and each note sounds slightly different, just like on a real instrument.
- 'Dulci Delicate': An authentic-sounding hammered dulcimer, with distant bouncing repeats.
- 'Soft Pad 1': An unusual cross between a brass sound and that of a bowed string, with rosin scrapes on top notes.
- 'Cosmos Soundscape': A unique and haunting combination of bowed string and resonant enveloped filter.

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String Things

The more interesting low-level design of your instruments goes on in Panel B, where there are six main modules. Some of their controls are multi-functional and context-sensitive — in other words, the labels below some of the knobs change as you go along, but thankfully this feels quite natural. The most important module is the virtual String, which is permanently 'in circuit'. Its Damp and Decay controls let you adjust the high-frequency content and decay time of the string respectively, to mimic different materials such as nylon or metal, while the Inharm control lets you detune the upper partials, as though you were changing the 'width' of the string.

Additional grey rotary knobs beneath the main controls let you alter the amounts of various modulation signals related to the main controls, and each knob may control and display the values of up to three sources, selected by clicking on one of the green routing 'LEDs' to their right. The modulation signals can also be inverted by clicking on the LED above each knob. For the String Module, the sources are Keyboard (which alters the Damp and Decay values so that the high notes have a shorter decays and more damping), and Ratio, which allows you to adjust decay times for note on and note off, to reproduce the action of dampers on a string.

The string is set into motion by the 'Excitator' module to its left, and this is where you can choose a plectrum, hammer, bow, or bouncing hammer (like those used to play a dulcimer, for instance). This is also where you use the Geometry controls to determine at what position you'll 'play' the string, where the string will be damped (if anywhere), and whether or not its sound will be heard naturally or via the modelled magnetic pickup. The latter is movable along the string so you can achieve bridge or neck sounds.

The remainder of the modules are all optional, and are activated by clicking on the LEDs alongside their title. Beneath the string module is Termination, which models the finger or fret pitching contributions, while above it is the Damper, which adds features for 'stopping' the sound, like virtual felt dampers (as on a piano) or a performer's finger (as on a guitar). The string signal can then pass through the Filter section, where you can add modulation courtesy of the ADSR filter envelope, or put it through a wide variety of more traditional 'analogue' swept filter treatments from subtle to extreme. There are low-pass, band-pass, notch, and high-pass options, a very handy three-peak formant filter for throaty vocal-like sounds, plus a dedicated LFO with a variety of waveforms and sync options.

Next in the signal chain is the Body module, where you can mix in the unique resonances of a range of soundboards ranging from that of a tiny violin to a huge piano, and fine-tune the resonance mix and decay to simulate different materials. This module can add a lot of character. If you want to add a little grit, you can also switch in the Distortion module, which contains a choice of algorithms from mellow to metal. The final module in Panel A is EQ, where you can fine-tune your instruments using a three-band design with low- and high-shelving sections, and a parametric mid section.

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Panel A

Panel A contains the more traditional controls for fine-tuning your sounds, including multi-effects.

There are quite a few controls in the seven modules of Panel A, but it doesn't take too long to get the hang of them all. The Vibrato module provides pitch modulation via a dedicated LFO with Rate and Amount controls. Vibrato can be delayed and faded in using two further controls, or added in real time with the mod wheel. However, the highlight of this section for me is the Error control, which lets you add a varying amount of randomness to the Vibrato controls for each voice. It's great for adding slightly differing vibrato to each 'player' in a string section to enrich the ensemble effect, for instance!

The Keyboard module contains a set of master controls covering overall tuning in octaves, semitones, and Hertz, while the Stretch and Error knobs allow you to create the 'stretched octave' tunings of the acoustic piano and add an element of slightly random tuning to every note of your orchestral strings, for instance. For thicker sounds, you can switch on double or quadruple Unison modes at the expense of similarly increased CPU overhead, or to quickly convert a six-string guitar into a 12-string version. Associated Detune and Delay controls add 'jangle' and 'spread' respectively. Finally, you can switch from polyphonic to monophonic mode for sounds like solo guitar, and select from low-, high-, or last-note priority to suit your playing style. The separate Portamento module lets you add pitch slide between the notes in both monophonic and polyphonic modes, and includes a Legato mode so that it only triggers when one note is pressed before the previous one is released.

The optional Arpeggiator module may seem an odd inclusion for a stringed instrument, but apart from its synth runs, you can also press it into service for harp glissandos. It provides a Range in octaves, a selection of four note-order options, a direction with the Span options (upwards, downwards, or both), and a latch mode that keeps playing the current arpeggio continuously until you play some different notes. However, its real strength is the Pattern display, where you can individually activate or deactivate any combination of steps in a length of anywhere between one and 16 steps to produce rhythmic patterns. Your modified patterns are saved with the associated presets. The overall Rate is either set by a rotary knob, or can be locked to a variety of synchronised options tied to the master internal clock of the Clock module. This is a clever design in itself, where Tempo can be typed or tapped in, while an External option lets you lock *String Studio* to the tempo of your host application.

The Output Effects comprise chorus, delay, and reverb, each with a separate Bypass option. All three provide a good selection of alternative algorithms including mono and stereo chorus and flanging, digital, ping-pong, slapback, and tape delays, and drum room, club, and various room and hall reverbs. The order of the chorus and delay effects can be swapped, and their tempo set manually or locked to a wide variety of sync options. Overall, I found the effects very useful in adding the final touches to instrument sounds, from added richness to swirling stereo movement, and short slapback to moving ping-pong delays, and the reverb is versatile and has relatively smooth tails, although you might want to replace it if you have more advanced plug-in alternatives. The final section is the Recorder module, where you can save the audio output of *String Studio* as a WAV or AIFF file.

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In Use

Playing this synth is a joy, and it's easy enough to start creating your own presets by modifying existing ones. However, be prepared for a little initial confusion as you explore the various parameters — this is, after all, a totally unique synth engine, so quite a few of the controls will be unfamiliar. Thankfully most of them make sense fairly quickly once you try them, such as the Stiffness and Protusion of the plectrum, the Force and Friction of the bow, and the Mass and Stiffness of the hammers.

I soon had some lovely acoustic and electric guitar sounds with lots of character and expression, plus a bowed string that resembled the sound of a string quartet, and various ethnic instruments reminiscent of the Japanese koto. However, further exploration proved that the engine could be pushed into unexpected directions — the Bow Excitator, in particular, is capable of creating some wonderful ethereal pads, even before you add EQ and effects, and once you add this final polish you can end up with sounds that I very much doubt you'd achieve elsewhere.

Since this synthesis engine calculates all the sounds in real time as its various modules interact, I found that each note could sound slightly different depending on my performance, just like a real instrument, even before I linked MIDI controllers into the picture to provide even more expression. However, physical modelling is always likely to be more computationally intensive than sample-based synths, so it's important to keep an eye on your maximum polyphony.

You can select the number of voices from a choice of two, four, eight, 12, 16, 20, 24, 28, or 32, although the number of simultaneous notes available is also affected by the Mono/Poly mode and whether Unison mode is active. It's a shame that there's no six-voice setting, since this would be the perfect option for acoustic and electric guitars to avoid wasting CPU power. However, like *Lounge Lizard*, *String Studio* has dynamic voice allocation, which means that your CPU need only run those notes that are actually playing. Typically, the overhead for my PC's Pentium 4 2.8GHz processor was 30 percent for four voices, and 50 percent for eight.

If you've not already guessed by this point, I loved *String Studio*, but although many of the sounds are great fun to play, there are a few parts of the user experience that could definitely be improved. Some of the controls are really tiny (particularly the LEDs next to many controls), making some mouse maneouvres a frustrating experience, and given the amount of virtual wood grain and brushed metalwork in the graphical user interface, perhaps all of the controls could be made larger in a future update. The preset browser also has a fixed width sufficient to view the folder names, but once you open a folder you need to indulge in a lot of sideways scrolling to see the contents. Finally, there's insufficient space to display the full preset and MIDI link names on the menu bar, making it difficult to remember which you've selected, and the MIDI activity indicator disappears completely if you hide the Browser window.

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MIDI


The bundled library is nicely organised, while the various MIDI Link controls let you tap *String Studio's* expressive potential.

All parameters in the VSTi and DXi versions of *String Studio* can be automated from their host application by directly moving the controls and recording their movements. However, keyboard players will appreciate the ability you have to change any parameter value in real time using your choice of MIDI controller — you just right-click on the control, select the 'Learn MIDI Link' function (shown above), and then move the controller that you wish to use.

Refinements include the ability to define the upper and lower parameter values linked to the controller's movements — and these can also be reversed, so that parameter values are reduced as the controller value increases. Since multiple parameters can be linked to a single controller, you can make several *String Studio* controls vary by differing amounts and in different directions simultaneously by (for instance) moving the mod wheel to achieve more complex interactions. Or you could set up multiple controllers, each controlling an individual parameter.

MIDI links to use with different instruments can be saved, imported, and exported just like the presets, and you can define one to be the default. These really are the key to being more expressive with *String Studio* — you could, for example, change your virtual finger-picking position using the mod wheel linked to the Excitator position, switch from neck to bridge pickup sounds via the pickup position parameter, or articulate bowed swells using overall level.

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Conclusions

In a perfect world, we'd use a talented guitarist or string player when we wanted these instruments in our songs, but few of us are lucky enough to have such performers on tap all the time. With practice, *String Studio* can provide some incredibly realistic virtual versions when we have to play in the parts via MIDI, but for me its most exciting aspect is the ability to create new monophonic and polyphonic stringed instruments that sound believable, react to your playing technique like acoustic equivalents, and can be pushed in directions that no other synth can emulate.

You could download the demo version, skim through its presets and not be particularly impressed, but once you start playing this instrument seriously and experience the realistic way it responds to your performance (and particularly once you involve MIDI controllers), you'll be hooked. This is a unique synth that sounds like nothing else on the market. SOS

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SOUND ON SOUND
The World's Best Music Recording Magazine

Apogee Big Ben

Master Digital Clock

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Reviews : Accessory

Apogee's new master clock includes their latest clock-regeneration technology, which can synchronise to even very poor-quality clock sources.

Hugh Robjohns

Photos: Mark Ewing

The vast majority of home studios these days will undoubtedly have some digital equipment, and many already have quite complex configurations of hardware and computer-based digital audio equipment. As the level of complexity rises, the role of a high-quality, central master clock becomes increasingly important, both to keep everything in synchronisation and to act as a central clock distribution point. Daisy-chaining clocks between equipment can work in small systems, but propagation delays can become a serious problem in larger installations, and a star distribution of stable time-aligned clocks is a much better solution.

Apogee's latest contribution to this area of the market is the 1U rackmounting Big Ben, which is claimed to represent the state of the art in master-clock design, with clock distribution and a range of interesting features built in. The name is derived from the widespread association of Britain's Big Ben with accurate time keeping — even though the name actually refers to the large hour-chime bell in St Stephen's Tower at the Palace of Westminster, rather than to the clock mechanism itself!

Apogee Big Ben £1228

pros

- Grade 1 master clock supporting all rates to 192kHz.
- Up to ten clock outputs.
- Flexible rate variations plus Super Clock and DSD modes.
- Comprehensive external reference options.
- Word-clock termination checking.
- Digital audio reference distribution and format conversion.
- Sure Lock maintains clock if external reference is lost.
- Video-related pull-up/down modes and VSO.

cons

- No conversion between single-wire and double-wire AES modes.
- Potential sync glitches when recovering from Sure Lock.
- Sure Lock bug in review sample.

summary

We have come to expect nothing less than excellence in clocking and jitter handling from Apogee, and the Big Ben is no disappointment. As a master clock it excels in every way, but the unique additional functionality of audio signal distribution and format conversion will prove very beneficial to many.

top

Grade 1 Master Clock

The Big Ben is fundamentally a Grade 1 master clock generator able to operate

information

£1227.88 including VAT.

Sonic Distribution +44 (0) 1525 840400.

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www.sonic-distribution.com

at any of the standard rates from 44.1kHz to 192kHz. It provides six word-clock outlets, two AES outputs, a coaxial S/PDIF socket and a configurable optical port for S/PDIF or ADAT interfaces. Two of the word-clock outputs can be configured to provide DSD, Super Clock, or various other multiple-rate references. There is also an option-card slot which can be used to extend the unit's compatibility with future interface standards. Firewire compatibility is already available, and a video generator card is expected shortly.

Many master clocks are designed to run independently, but the Big Ben can be slaved to an incoming clock reference signal (including video), if required. Apogee claim that any jitter on an external reference clock is removed to such an extent that it is usually impossible to tell the difference between Big Ben's operation slaved to an external clock or running from its internal clock generator. An extension of this de-jittering functionality is that, when using a reference input carrying audio (in other words AES, S/PDIF, ADAT, or Firewire), the data is automatically de-jittered before distribution to all the other audio-capable clock outputs.

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Connectivity

The Big Ben carries a lot of socketry on the rear panel to enable easy connection with a wide variety of equipment, although, unlike many master clocks, there are no front-panel interfaces to make it easier to accommodate temporary equipment. Six BNC output sockets to the right-hand side of the back plate carry separately buffered standard word-clock outputs. The first four operate at the unit's selected clock frequency, while the last two can be configured independently to output clocks at some multiple of the clock rate (quadruple, double, half, or quarter the selected sample rate, plus Super Clock or DSD references). To the left of these outputs is a seventh BNC socket which accepts an external word clock or a video reference input signal. The socket is permanently terminated with 75(omega), and the input format is determined by a front-panel menu selection.

Moving further left, there is blanking panel for the slot-in option card. This is a nice idea and in theory enables the Big Ben to provide reference clock signals for any future format. The X-Firewire card was the first to become available, potentially enabling keyboards, synth modules, and other Firewire-based devices to provide stable clock references. A forthcoming X-Video card will allow Big Ben to function as a master video-sync generator in sync with its digital word-clock outputs.

The remaining interface sockets can all carry digital audio in various formats, and are provided as both inputs and corresponding outputs. S/PDIF is catered for with both coaxial and optical connections for input and output, but the Toslink connectors can also be configured for the ADAT format. The AES interface is equipped with dual XLRs for both inputs and outputs, allowing a choice of AES references as well as accommodating a high-sample-rate stereo signal in the double-wire mode.

It is unusual for a reference clock to distribute audio along with the embedded clocks (because of concerns that the audio data can increase cable-induced jitter artefacts), but that is precisely what Big Ben does by default. If the idea of distributing audio with your embedded clocks worries you, there is an internal jumper option to defeat the audio distribution mode and send 'digital black' (in other words a mute signal) to all digital audio outputs instead.

However, by default the audio carried on a selected external clock reference is distributed automatically to all the outputs capable of carrying audio, complete with the necessary format conversions. The situation is complicated slightly since the ADAT interface is able to carry up to eight channels while the rest are only stereo, but the channel routing is entirely logical given the inherent restrictions. So, for example, if an eight-channel ADAT signal is chosen as the reference, all eight channels will appear at the optical output (still in ADAT format), the first two channels will also appear on the coaxial S/PDIF output and the first AES output, and channels 3+4 will be presented on the second AES output. Likewise, a stereo input is copied to all the other stereo outputs, and repeated across each pair of available ADAT outputs. The only disappointing limitation is that it is not possible to convert between single-wire and double-wire AES signals — although I doubt that will be a concern to many.

The final back plate connector is the usual IEC mains inlet. The internal power supply is a switched-mode design that accepts 100-240V AC, at 50Hz or 60Hz.

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Controls & Configuration

The Big Ben shares its front-panel styling with other Apogee products: purple and silver background colours with clear black and silver labels. Controls are kept to a minimum with just a quartet of cursor keys and a power/standby button, but the panel also carries a comprehensive set of LEDs to indicate every aspect of the unit's status. An alphanumeric LED display is included to show the current sample rate, various configuration messages, and the lock status. Turning the unit on results in an entertaining light show as every LED is tested (several times!) and the unit's name is flashed on the LED display.

Unusually, the powering arrangements can be configured with a pair of internal jumpers. The default condition is for the front-panel button to toggle the power mode as you would expect (although this is not a traditional mains isolation switch — it is a control input to the power supply). However, this button can be disabled completely if required to prevent the unit being switched off accidentally, and there is also an 'auto-on' option in which the unit will power on automatically whenever a mains supply is connected (but it can still be switched between the On and Standby modes). This is a useful set of options that will appeal particularly to installers using the Big Ben as the reference for a large system.

The four cursor buttons are used to adjust the Big Ben's configuration in a pretty intuitive way, and there is a degree of 'intelligence' such that inappropriate functions and settings are automatically hidden from or added to the available options as the machine's configuration is changed. For example, if an external clock source is selected, the facility to change the sample rate is removed, since that parameter is defined by the external clock source itself.

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The Art Of Clock Regeneration

Any system that has to extract and regenerate a clock signal relies on some form of Phase Locked Loop (PLL). However, extracting a precise clock requires a rather inflexible PLL, unable to track widely varying source clock rates, while a more flexible PLL cannot exclude jitter artefacts very well.

Some systems try to overcome these inherent difficulties by using multiple stages of progressively more precise PLLs, but Apogee have taken a different approach with their latest clock technology, code-named C777. This is an entirely digital process (instead of the analogue or hybrid analogue/digital approaches more usually employed) which uses Direct Digital Synthesis (DDS) technology to generate the required clock frequency — allegedly with immeasurable jitter. When synchronised to an external reference clock, DSP-based adaptive digital filtering is used to condition the source clock, and the combination is claimed to provide the most effective jitter reduction available. In theory, even poor sources with excessive jitter can be used as a master clock, and the de-jittered signal can even be passed on to other equipment.

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Clock Sources

On either side of the sample-rate display are two columns of legends used to indicate the current clock source — and it is a comprehensive list! On the left are options for the internal mode and the external reference inputs: the two AES inputs, the S/PDIF coaxial connector, the optical port, and the option card. On the right are legends for the external word-clock and video-input options, and if Video is selected then the detected video format is also displayed (NTSC, PAL, or B&W). There is a warning in the handbook that it can take up to 30 seconds for the unit to fully stabilise to a video reference, and if the AES mode is configured for double-wire operation, then selecting AES 1 as the reference input automatically illuminates the AES 2 flag as well.

The four-digit sample-rate display shows the selected clock frequency if operating from the internal clock generator, with all the standard rates between 44.1kHz and 192kHz. If the Big Ben is locked to an external source, the display shows the actual (measured) input sample rate. Immediately below the numerical read-out is an indication of the unit's lock status, using a green arrow and a blue LED, along with Lock Wide and Lock Narrow legends. When the unit is solidly locked to the selected reference both the green arrow and the blue LED light steadily, along with the Lock Narrow legend.

There are two situations where the Lock Narrow indicator may go out. One is if an external clock source is jittery or unstable in some way. In this case the blue light goes out and the Lock Narrow legend is replaced with Lock Wide — meaning that the unit is still synchronised to the external reference and the derived clock outputs are stable, but that the external source is dubious in some way and Big Ben is struggling to remove all of the instability or jitter.

The other situation is when the unit loses lock completely — such as if the reference input cable becomes disconnected. In this case the Lock Narrow legend goes out immediately and the green arrow starts blinking — but the blue light remains on to indicate that Apogee's Sure Lock function is active. Essentially this means that the Big Ben is continuing to generate clock outputs at the last valid sample frequency from its internal clock generator. When the external reference is reinstated the internal clock will slew back into synchronisation with the external feed (which is indicated when the arrow stops blinking and the Narrow Lock legends illuminate). However, depending on the attached equipment and the relative local/reference drift, some sync disturbances may occur as the Big Ben re-establishes lock. For situations where it would be better to be aware of a missing reference clock, an internal jumper is provided to defeat the Sure Lock mode if required.

Sure Lock is a good idea and seemed to work well... when it worked. However, I found that the review model appeared to disable the Sure Lock mode automatically! Resetting the unit to the factory default (by holding the Down key while turning the power off and on again) activated the Sure Lock feature which worked as advertised the first time I removed an external reference clock.

However, subsequent removal of the clock resulted in no clock outputs and '0000' displayed on the LED sample-rate counter. Only resetting the unit would temporarily reactivate the Sure Lock mode and I could find no explanation for this odd behaviour in the handbook. After checking with Apogee, it turns out that this is a known bug that strikes at random, and which they are hoping to address in a forthcoming firmware release.

Changing any of the machine's parameters or configurations is a simple case of navigating to the required setting using the Prev/Next buttons, and then changing the value with the Up/Down buttons. Pressing any of these four keys causes the unit to enter Setup mode, indicated by a small LED in the centre of the quartet of buttons. The most recently adjusted parameter then flashes and it can either be adjusted further or a different parameter can be selected and then adjusted as necessary.

Repeated presses of the Prev/Next buttons cycle around the various options. The complete set comprises the clock reference source, the sample rate, the optical interface format, the AES format, the video pull-up/down options, and finally the multiplier rates for the fifth and six word-clock outputs. Not all of these are available at all times, depending on the underlying configuration. The unit drops out of Setup mode automatically two seconds after the last button press and the displayed parameter stops flashing to indicate that the new setting has been activated. All settings are stored in non-volatile memory and are recalled when the unit is powered up. If required, the default factory settings can be restored by holding the Down key depressed while powering the unit off and on, and if the Up key is held when the power is cycled word-clock outputs five and six provide DSD reference clocks.

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SMux Output Options

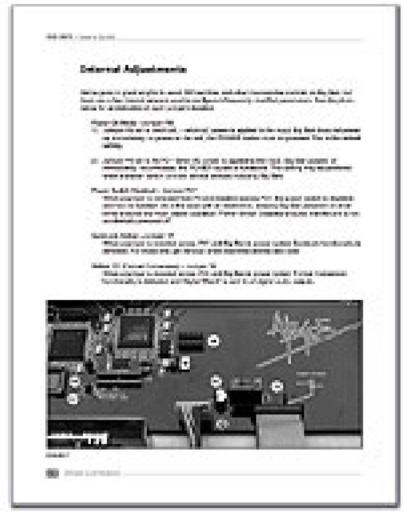
I have already explained the clock-reference and sample-rate options. The following adjustable parameters concern the optical and AES interface formats. The optical port can be switched between S/PDIF and ADAT modes, but the latter is extended with the associated SMux 2 and SMux 4 formats. SMux 2 conveys four channels at 88.2kHz or 96kHz over the standard ADAT interface, while SMux 4 provides two channels at 176.4kHz or 192kHz. The AES modes are simply single-wire or double-wire configurations — the former supporting two channels at all rates between 44.1kHz and 192kHz, and the latter using two AES pairs to convey stereo signals at rates from 88.2kHz to 192kHz (but operating each cable at half the sample rate).

The next set of options affects the pull-up/down multipliers that are often required when working with film or NTSC video — but these are only available when the Big Ben is switched to internal clock or external video references. By default, the LED array shows None, meaning that the sample rate is as shown on the numeric display. However, if any of the pull-up/down options are employed, the appropriate offset is indicated by an associated LED: ±4 percent or ±0.1 percent. The first is used to translate between 24fps (frames per second) film and 25fps PAL video, and the second for working with 29.97fps colour NTSC and either 3:2 converted film or B&W (non-drop frame) NTSC. Clearly, these are specialised functions, and a detailed explanation of their use is beyond the scope (and space) of this review — but their inclusion certainly broadens the usefulness and appeal of the Big Ben as a master clock in video-related audio areas.

Another option in this section is labelled VSO — Variable Speed Override. When operating on the internal clock, entering this mode allows the user to change the sample rate to any desired value. The varispeed display can be given as sample rate, percentage change, or musical cents (the display mode being changed by holding the Prev/Next buttons).

The final parameter sets determine the independent multipliers for the outputs from word-clock outputs five and six. The options are x1, x2, x4, and x256 (Super

Clock), plus /2 and /4 (in other words division by two or four), although not all options are valid with all base sample rates. The x4 and x256 modes are sensibly disabled for sample rates above 88kHz, and the x2 mode for rates above 176kHz. Likewise, the /4 mode is inactive for rates below 176kHz, and the /2 mode for rates below 88kHz.



The user manual for the Big Ben shows a number of internal jumper selections which can be made to customise the operation of the unit.

As mentioned earlier, if the Up button is held while the unit is powered off and on again, outputs five and six are re-configured to supply DSD reference clocks. In this mode output five provides a clock at x64 the base rate (around 3MHz), while output six provides a signal at x128 the base rate (around 6MHz) — and in this case the base rate means the lowest denominator of the selected sample rate (so either 44.1kHz or 48kHz). The front-panel display doesn't highlight this special DSD mode in any way, although if you try to adjust the multiplier value for these two clock outputs you find they are inaccessible. Turning the unit off and on again resumes the standard operating mode.

The final array of LEDs on the right of the front panel comprises pairs of red and green lamps for each of the six word-clock outputs. Green LEDs illuminate when a correct 75(omega) termination is sensed, and red LEDs shine if the detected load is less than 75(omega) (for example an incorrect 50(omega) termination, a double termination, or a cable short-circuit). Unterminated or open-circuit cables are indicated by extinguishing both LEDs. This facility reassures that the system is wired correctly, and is also useful in confirming that clock connections are correctly and precisely terminated — something which is always important, but which becomes critical with elevated clock rates.

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In Use

My own digital studio equipment is generally clocked from an Aardvark Aardsync II, supplemented by a Drawmer M-Clock for the occasions when I need to operate separate sections at different sample rates. Consequently, plumbing in the Big Ben was a straightforward task.

Everything worked exactly as expected, and I found configuring the Big Ben very fast and easy once the functionality had become familiar — although there do seem to be an awful lot of button presses involved when trying to change some settings. In most cases, though, once the system is configured only the sample rate would need to be changed regularly, and since the Setup mode remembers which parameter you last changed, changing the rate again becomes much faster.

I was unable to hear any difference from the connected equipment when switching between the Aardsync II and Big Ben as master references, but in itself that suggests that the Apogee box is at least as good as the Aardvark — one of the best clocks available in my opinion — and quite possibly better. Given the moderate cost of the Big Ben and its far more extensive functionality, this is a very impressive result.

Using the Aardvark as an external reference for the Big Ben, I was able to check that its sample rate display gave accurate values for a range of non-standard reference sample rates. The Aardvark can generate numerous pull-up/down rates, but the Big Ben recognised them all without any problems. It even locked happily to 32kHz despite this being outside its declared operating range.

One aspect I was interested in investigating further was the Big Ben's ability to reduce the jitter of poor-quality sources, so I hooked up a cheap domestic portable CD player, using its S/PDIF output as the embedded clock reference and using an ordinary 1m audio phono cable (not a proper 75(omega) digital cable) to make life as difficult as possible, along with a very simple old stand-alone 'hi-fi' D-A converter without any sophisticated de-jittering functions of its own. It didn't take much critical listening before it became clear that the soundstage was flat and lifeless, and the acoustic ambience did not have the spaciousness that I knew was encoded in the source material. I then re-plugged the audio phono cable carrying the S/PDIF signal to the Big Ben and used a pukka 75(omega) cable from the Big Ben back to the converter. The difference was quite marked, with the stereo image becoming far more solid and three-dimensional, and the subtle ambience cues becoming far more lifelike. An impressive result indeed, if entirely subjective, suggesting that the de-jittering capabilities of the Big Ben are very powerful.

Overall, the Big Ben is an impressive machine with capabilities that reach well beyond those normally associated with master clocks. As a result, it is a very cost-effective and pragmatic device for a wide range of applications, and will serve as a flexible problem-solver in many more. Equally at home in a modest home studio, an audio-video post-production suite, or a high-end digital-audio production or mastering studio, the Big Ben's unique feature set makes it worthy of very serious consideration. SOS

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Apple *Logic Express*
£199

pros

- Incredibly powerful.
- Generous library of Apple Loops, samples and plug-ins as standard.
- Full *Garage Band* compatibility.

cons

- Not much less complicated than *Logic Pro*!

summary

If you feel *Garage Band* is too basic for you but you can't justify buying *Logic Pro*, *Logic Express* will get you most of the way there for surprisingly little money.

information

£ £199 including VAT.
T Sound Technology +44 (0)1462 480000.
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Apple Logic Express

MIDI + Audio Sequencer [Mac OS X]

Published in SOS August 2005

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Reviews : Software

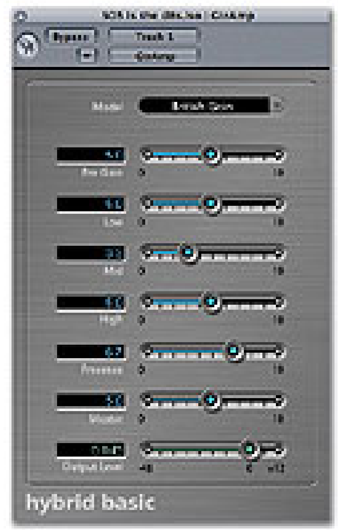
Buying the *Express* version of Apple's *Logic* will save you a cool £500 over the full-blown *Pro* edition — and you'll still get a mighty powerful sequencer...

Paul White

Back in April 2005, Martin Walker wrote a very well reasoned article pointing out that many people don't really need the flagship versions of MIDI + Audio software packages. Most home-studio users could do pretty much everything they need using a 'light' version of their favoured program, saving themselves money and avoiding undue complexity at the same time. For that reason, I thought it was worth throwing the spotlight onto the light version of Apple's *Logic*, which offers particularly good value for money.

The light version of *Logic Pro* is, of course, *Logic Express*, and when you look at it a bit more closely, it isn't actually very light at all, despite being priced at under £200. Indeed, in many respects, *Logic Express* outperforms the original 'full fat' *Logic 6* and includes many of the cutting-edge features of *Logic Pro* 7.1, as well as benefiting from recent additions to *Garage Band*. You get fewer bundled plug-ins than with *Logic Pro* and the software only offers around 10 times as many audio tracks than you're ever likely to need (255) rather than 30 times more than you'll ever need, but you can still run up to 64 software instrument tracks at a time, which is more than any current computer can reasonably support. As track count is largely a function of hardware and drive speed, the real-world maximum track count for *Logic Express* and *Logic Pro* is essentially the same.

Apple have limited the I/O capability of *Logic Express* to 12 simultaneous inputs



and outputs with a maximum sample rate of 96kHz, but again that's more than most project studio owners need. Similarly, the number of busses is restricted to eight, but these can be stereo, and as most people use them simply as effects returns, that shouldn't be too much of a limitation — I don't think I've ever used as many as eight yet. The surround option has also been dropped, as has distributed audio processing. Perhaps this last omission is a mistake, as keeping it in might just have enabled Apple to sell a few more computers to use as processing nodes, but I don't think many *Express* users will miss it. Audio files can be handled in WAV, *SDII* and AIFF formats, and MP3, AAC and Apple Lossless import and export is also supported, but more specialised formats, specifically AAF import/export, Broadcast Wave support and *Final Cut Pro* XML import/export, have been omitted. Quicktime movies can still be run in sync within *Logic Express*, though, so anyone wanting to experiment with sound for picture can do so very easily.

In order to keep the education market enthused, Apple have left in the whole of the very serious scoring section from *Logic Pro*. *Garage Band* songs can now be opened in *Logic Express*, and all the Apple Loop material developed for *Garage Band* works just fine in *Express* too. All the *Logic Pro* mix automation features are included, as is support for control surfaces, the only difference here being that *Logic Express* can only support one 'bank' of control surface — so, while you can use a Mackie Control plus expanders configured as a single control surface, you can't have one working as a mixer and another dedicated to controlling soft synths. With *Logic Pro* 7.1, full plug-in delay compensation was extended to the busses and outputs, so that's now been added to *Express* 7.1. This also works with third-party DSP-powered plug-ins such as those running on Universal Audio and TC Electronics cards.

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Looking Around

As a *Logic Pro* user, I found *Express* instantly familiar and it's only after you've been working with it for a few minutes that you realise anything is different. The first thing I clocked was that the transport window didn't have all the layout and resize options offered by *Pro* and it couldn't be turned into a giant SMPTE display. In fact *Express* has no dealings with SMPTE at all, though it can sync to MTC and supports MIDI Machine Control. The second thing I noticed is that when I clicked to engage the channel EQ, it called up a more basic four-band equaliser with shelving high and low sections augmented by two swept mids.

There's no spectrum analyser and no linear-phase EQ, either, but it's still a very good and flexible EQ section with a friendly graphical interface. One other difference is actually an improvement: in *Logic Express*, the track icon menu still appears as a matrix so you can see all the options at once. *Logic Pro*, by comparison, has switched to a seemingly endless list that is frustrating in the extreme to use.

Although *Logic Express* lacks many of the 'big gun' plug-ins featured in *Logic Pro*, you can do a lot with the equivalents borrowed from *Garage Band*. These include guitar and bass amp modellers and the new *Hybrid* synths.

Another omission I found after wandering around the Environment is that there are no dedicated live input tracks for external synths as there are in *Pro*, and I could find no way to route external signals into the Aux channels. If you want to feed a live instrument into *Logic Express*, you need to route it via a track input set to 'record ready' mode.

The familiar Matrix, Arrange, Track Mixer and Environment pages are there along with Score, Event List, Transform and Hyper Edit. There's no Project Manager, but you can still save songs as Projects to keep everything in one folder, which is a great improvement to *Logic*'s workflow in my view. The most obvious missing items are the 'big gun' plug-ins that come with *Logic Pro* 7.1: the guitar and bass amps, *Match EQ*, *Sculpture*, *Ultrabeat*, *EVB3*, *EVP88* and of course *Space Designer* and *Pitch Correction*, but you still get all those cute 'crop circle' analogue and FM synths, the *ES1* monosynth and the *EXS24P MkII* sample player with a very decent sample library thrown in to boot. New samples can be added if they're in *EXS24* format, but you can't create your own samples or import other sampler formats with this version.

On top of that there's the full set of *Garage Band* instruments, and if you haven't tried these yet, I think you'll be surprised at how good they are. Though simple on the outside, they use the engines of the *Logic Pro* plug-ins, and the tonewheel organ sounds every bit as good as its big brother while being a lot easier to use. *Digital Stepper* is also very versatile and the analogue synth models are as warm and fat as they should be. The *Garage Band* instrument set includes a sampled piano plus some perfectly acceptable drum kit sounds, and with v7.1, these are joined by two new *Hybrid Synth* plug-ins. In all there are now almost 40 effects plug-ins, with reverbs going all the way up to *Platinumverb* (the top of the tree before *Space Designer* came along) and nearly 20 software instruments. Guitar and bass players will also find modelling guitar and bass amps in there, and though they aren't as fancy as ones in *Logic Pro* 7.1, they work well and are easy to use. To complete the package you also get a vast library of Apple Loops (over 1000) so there's really nothing else you need to get started making some very high-quality music other than an audio interface and a MIDI keyboard.

With the update to 7.1, there are further useful additions. Since pitch and time correction were introduced to *Garage Band*, *Logic* has been updated to maintain compatibility, so although there's less functionality than with *Logic Pro*'s *Pitch Correction*, you can still choose a musical scale, then correct your vocals to it using just a single slider. This is a great bonus and almost worth the cost of the entire *Logic Express* package if you're not a great singer. Similarly, audio timing can be improved using a single correction fader, though the quality of the results depends on the type of input source being 'quantised'. In a similar area, there's now a 'Follow Tempo for Recorded Audio' option that applies time and pitch-shift to the file when you change the song's tempo or key. Again the results depend on the type of audio and how far you try to move it from its original pitch or tempo, but used over a sensible range, the results are impressive.

Ease of use has also been improved, so as well as the original 'Wizard'-style function to help you set up new songs, there's now plug and play for audio interfaces. As soon as you connect and switch on an external audio interface while *Logic Express* 7.1 is running, a dialogue box appears and asks if you'd like to use the new device.

Another feature *Logic* users have been asking for for a long time is the ability to drag or copy plug-ins without having to visit the Audio Configurations window. Now you can do it in the Track Mixer or Arrange windows using a hand tool. Very welcome! The key command set has been increased, the Automation Curve Tool from *Pro* has put in an appearance and there's now dedicated support for the Korg Kontrol 49 and Micro Kontrol, the Frontier Design Group's seriously wonderful wireless Tranzport, Tascam US2400 and FW1082 controller/interfaces plus the JL Cooper CS32 and Fadermaster 4/100.

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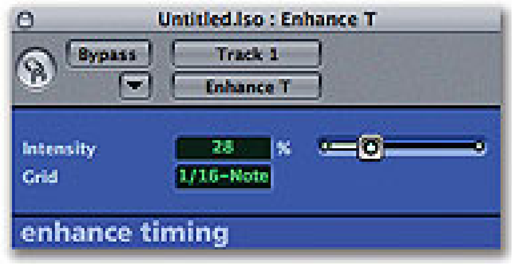
Final Thoughts

The plug-ins that Apple have tended to omit from *Logic Express* are, in the main, also the ones that take a lot of processing power, so if you're not using a G5 machine, *Logic Express* might be a good choice, especially if you don't need some of the more esoteric functions of *Logic Pro* or the ability to sync directly to SMPTE. The Track Freeze feature means that even older machines can be used successfully, as processor-intensive audio or instrument tracks can be temporarily frozen to conserve CPU overhead at the expense of only a little more hard drive activity. In many cases, what's been lost from the *Pro* portfolio has been replaced by the back door courtesy of *Garage Band*, so all the essentials are covered. Another good point is that *Logic Express* doesn't use a hardware dongle, which is a great benefit if you're using a laptop on the move. I won't go into how well the new v7.1 features work as that is covered in our overview of *Logic Pro* 7.1 in *Logic* Notes, but in the short time I've had the upgrade, I haven't found anything I don't like.



Pitch and timing correction are available in *Logic Express*, again thanks to *Garage Band*-style single-slider interfaces.

If there's a criticism, it is at the philosophical level, insomuch as *Logic Express* actually gives you too much, so beginners probably won't find it much easier to learn than *Logic Pro* — even the notorious (but simple once you get to know it) Environment window is there in all its glory. However, the up side to this is that when you eventually feel the need to move from *Logic Express* to *Logic Pro*, the learning curve is almost non-existent, and of course there's always the option of starting out with *Garage Band* and then working up to *Logic Express* when you're ready. Given its low price, great functionality and generous library of plug-ins, samples and loops, *Logic Express* is a fabulous bargain and, teamed with a Mac Mini, could form the basis of a very serious music studio at a seriously attractive price. In essence, you're paying less than a third of the price of *Logic Pro* and getting over 90 percent of the functionality. SOS

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BSP Cubase Keyboard

QWERTY Keyboard

Published in SOS August 2005

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Reviews : Accessory

Martin Walker

Back in May 2003, Paul White tested BSP's *Logic* Keyboard, which featured a custom set of coloured keycaps displaying the keyboard shortcuts defined for *Logic* v5.5. The keycaps could be bought pre-fitted to Apple USB or wireless keyboards, or in a kit form, comprising a set of loose keycaps for you to fit yourself.

Two years on, many more applications now support BSP's keyboards, including Apple *Final Cut Pro*, Adobe *Photoshop* v7, Sony *Vegas*, and Steinberg's *Cubase* or *Nuendo*. Strangely, although *Cubase* and *Nuendo* are cross-platform apps, the keycaps for these two packages are only available for PC, and come pre-fitted to three possible keyboard types — PS/2 Classic, USB Multimedia, or Wireless Multimedia. No DIY kit is available.

I was sent the PS/2 Classic, and although I was already familiar with many of the keyboard shortcuts, I learned many more in a short space of time thanks to this keyboard. All the letters, numbers, function and cursor keys, plus those of the numeric keypad (70 in all) have colourful replacements printed with the normal and Control functions normally allocated to *Cubase* (but not the shifted ones), as well as their original functions. The colours are carefully used to group related functions.

Of course, *Cubase*/*Nuendo* shortcuts are totally customisable, so these custom keyboards will only be useful to you if you haven't altered the default shortcuts. I'm afraid I wasn't very keen on this particular keyboard's action either, finding it rather 'flabby', although of course keyboard feel is a matter of very personal taste.

I would say that this keyboard could prove very popular with *Cubase*/*Nuendo* users but for one thing. PC owners are notoriously price-conscious compared with their Mac brethren, and even at the cheapest £82 street price I found for the

pros

- A great way to boost your *Cubase*/*Nuendo* productivity.

cons

- More expensive than standard PC keyboards.
- Only works with default Steinberg shortcuts.

summary

A great way for PC users to get more out of *Cubase* or *Nuendo* — if they can justify the price.

information

£ PS/2 version £92.83 including VAT.

T Contech Electronics +44 (0)1438 315757.

W www.logickeyboard.com

PS/2 version, I can't help feeling that many people will find it hard to justify spending over £60 more than the price of a typical standard alternative. SOS

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CME UF8

88-note Weighted Controller Keyboard

Published in SOS August 2005

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Reviews : MIDI Controller

The price of controller keyboards has fallen sharply over the past few years, but whoever thought that we'd see an 88-note weighted-action keyboard with aftertouch for under £500? Enter CME's UF8...

Martin Walker

There's certainly no shortage of 88-note weighted controllers available for the pianist on the market today. There are also lots of semi-weighted synth-action controllers with spans of 25, 49, 61, or 76 keys, most with mod and pitch wheels, and some with aftertouch. However, the UF8 keyboard controller from the Beijing-based Central Music Company (CME) is something I've been searching for for several years — a budget MIDI controller with 88 keys *and* aftertouch.

Photos: Mark Ewing

Some musicians assume that 88 weighted keys can only be used for playing piano and other percussive sounds, and that they aren't suitable for synth sounds and orchestral instruments. However, I've never been convinced by this argument, especially since many modern *Gigastudio* libraries require keyspans of 76 or more notes to easily access the keyswitching functions used to move between different articulations. Within reason, weighted keyboards can be extremely useful for playing all instruments, and moreover, not everyone has the space or the money for two controller keyboards. So, why use two keyboards in the studio when you could use one?

CME UF8 £430

pros

- Excellent weighted hammer action.
- Responsive aftertouch.
- Nine assignable sliders and eight assignable rotary controls.
- Breath-controller input.

cons

- Aftertouch and breath control require mains power supply.
- Rotary knob detents can make smooth filter sweeps difficult.
- Tiny power switch.

summary

If you want an 88-key controller with a high-quality weighted hammer action, aftertouch, transport controls, and a set of 17 user-definable controllers, the UF8 is by far the cheapest option. I highly recommend it.

information

£ UF8, £429.99; UF400E, £149.99. Prices include VAT.

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

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W www.cme-pro.com

Test Spec

top

Out Of The East

PC REVIEW SYSTEM

- 2.8GHz Pentium 4C PC with Hyperthreading, an Asus P4P800 Deluxe motherboard and an Intel 865PE chipset, running an 800MHz Front Side Buss, 1GB of DDR400 RAM, and Windows XP with Service Pack 2.
- Steinberg *Cubase SX* v3.0.2.
- NI *Pro 53*.
- CME USB driver v1.04.

Although this new UF range is the first to appear under CME's own name, their patented weighted keyboard action has apparently already appeared in various other companies' products. There are four models in the range. The UF5, UF6, and UF7 have semi-weighted synth-action keyboards of 49, 61, and 76 keys respectively, while the UF8 being reviewed here has a 88-key weighted hammer-action keyboard. Otherwise, they are identical, except that the UF8 is supplied with a sustain pedal — on the others, this is an optional extra.

All four UF keyboards incorporate several unusual features — the aforementioned channel aftertouch, a breath-control input, the option to retrofit an optional Firewire-based audio interface, plus a largely aluminium case compared with the plastic of some competitors' models. Each has eight rotary controllers, nine sliders, transport controls, pitch and mod wheels, plus sustain and pedal controller inputs. They are also excellent value for money — the UF8, with 88 weighted hammer-action keys is just under £430 in the UK, while the remaining semi-weighted models come in at £270 for the 76-key, £230 for the 61-key, and £170 for the 49-key versions.

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A Closer Inspection

The entire range has stylish good looks, and the unusual metallic red end cheeks and rear panel provide lots of character. At 23.5kg the UF8 isn't a lightweight keyboard, although it's still fairly typical of its genre, with M Audio's Keystation Pro 88 at 21.4kg and Oberheim's MC1000 at 20kg. However, those intending to gig on a regular basis should bear this in mind (CME's semi-weighted 76-key version would possibly be more suitable at just 11.8kg).

The indented modulation and centre-sprung pitch-bend wheels both have a smooth positive action and are placed to the left of the keyboard for easiest access, while the transport, controller, and editing controls are ranged in groups from left to right across the top, leaving plenty of space for you to rest a computer keyboard on top (as suggested by the top-panel graphics, which mark out the outline of one on the right). Round the back, there's a standard MIDI Out, the sustain pedal input, footpedal input, breath-control input, a USB port for bi-directional connection to your computer, a power socket, and an On/Off power switch.

This last switch is tiny, and the cause of one of my few hardware grumbles — it's simply hard to find and operate. Power for the keyboard can be supplied via the USB connector, but aftertouch and breath control is only available when you plug in the supplied adaptor and power it from the mains supply. This provides 12V DC at 1.5 Amps, sufficient to power devices like Yamaha's BC3 breath controller. These normally require a 15V supply, which couldn't be easily derived from a USB connection. However, the USB socket still has its uses — the USB drivers provide an additional MIDI input to send SysEx setup files to the keyboard, and are supplied for Mac OS X 10.2/10.3 and Windows 2000/XP.

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Audio Interface Option

The UF400E audio interface is an optional extra that wasn't available at the time of this review, although in fact I can tell you a lot about it, as CME have admitted that it's based on Terratec's Phase 24, which I reviewed in *SOS* May 2005. With a stereo pair of analogue balanced inputs and outputs, plus coaxial S/PDIF In and Out, MIDI In and Out, and a further unbalanced stereo out with level control suitable for a third or fourth line or headphone output, it supports sample rates up to 192kHz. I measured its dynamic range as a good 109dBA at 44.1kHz, and judged its audio quality as excellent for the price, and similar to M Audio's Audiophile 192 and ESI Pro's Julia.

However, the UF400E has some additional features not found on the Phase 24, including a mic preamp, a high-impedance guitar input, additional MIDI In, Out, and Thru sockets, and (yet another CME first) mLAN support, so it can be connected to one of Yamaha's digital mixers. I can't comment on the quality of the mic and guitar preamps, but judging by the parts of this audio package that I have tested, and its projected £150 UK price tag, I think this will be a very popular option.

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Editing

Most people will head for the editing section first, where they'll find a three-digit, seven-segment display, Inc and Dec buttons, and an endless data dial (rotary encoder) with a 'collar' of 12 LEDs around its circumference to indicate its virtual position. By default, the display shows the current Program number, so a quick spin of the dial will select your sounds, and you can change the associated MIDI channel after pressing the Channel button. After a few seconds, the Program button will take priority once more, no matter what other function you have been using. The four velocity responses are displayed as 0, 1, 2, and 3 (described in the manual as Norm, Hard, Soft, and Wide), and Aftertouch can be disabled when not required.

There's a Transpose button for shifting up to ±12 semitones, and an Octave Shift button with a ±3 octave range. I find I rarely need to transpose an 88-note keyboard, but these two functions become a lot more useful when using the Split function to turn those 88 keys into two separate zones with different MIDI channel and voice settings. The split is at F#2 (note 54) by default, but you can push and hold the Split button and press any other note to alter this position.

The Dual mode button lets you send MIDI data on two simultaneous channels to layer sounds, but you can only use it when the Split mode is off. Although the manual doesn't mention this, I found you could also set the Transpose/Octave functions individually for zones and layers — with the Split/Dual functions off, you set the lower zone/layer, and once you activate one of them, the same buttons set the upper zone/layer.

The final editing button is Drawbar, which inverts the action of some faders (more on these in a moment), but there are several additional functions available from two-button combinations, including All Notes Off (panic), GM, GS, and XG modes, a MIDI demo, and several hidden ones that I'll come to later. Usefully the various buttons stay illuminated if they are set to non-default values, to remind you if your current setup is transposed, uses a non-linear response, or is in dual, split, or drawbar modes.

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Controllers

The controller section comprises eight rotary knobs across the top, each with two functions, and nine faders beneath them, with up to three functions, all controlled from the associated knob and fader function buttons. The eight rotary knobs are allocated to the controllers normally associated with cutoff frequency, resonance, envelope attack and release times, pan, reverb and chorus levels, and tempo. The last of these eight knobs starts sending MIDI Timing Clock data as soon as you move it, and is adjustable from 20 to 250bpm. You can stop this by powering down and up, or using the Reset function, although this latter method also resets any custom controller allocations you may have made.

Clicking on the Knob Function button to light its associated LED reallocates these eight knobs to Expression, Breath, Bank MSB, Bank LSB, and four more identical Expression controls, but this time you can define your own choices for any or all of them — to allocate your own choice of controller number, you hold down the Knob Function button, tweak the desired control, and then dial in the new controller number using the endless encoder and/or Inc/Dec buttons.

The rear panel keeps it simple, with just MIDI Out, USB, Sustain pedal and footswitch connectors, and the very small power switch, although there is also the breath-controller jack, seldom seen these days.

When no LED is lit next to the Fader Function button, the faders control Main Volume and the volume of MIDI channels 1 through to 8, while one click of the Function button lights the other LED alongside this button (shown overleaf). This leaves the Main Volume as before, but switches the other eight faders to controlling MIDI volume for channels 9 to 16.

Another click on the Function button lights the lower LED, which defaults to providing Expression control for the first eight channels, but all nine faders can now be reallocated to your own choice of controller numbers for the current MIDI channel. Operating the Drawbar mode mentioned previously forces the faders to a fourth set of fixed controllers from #13 to #20, and inverts their action to act as organ drawbars with suitable instruments.

If you've defined new controller settings for the user-defined rotary and slider knobs, you'll be pleased to hear that they survive power down, although you can at any time return the UF range to its factory default settings. There are also a couple of 'secret' functions not mentioned in the manual that should be extremely popular — pressing Drawbar and Channel dumps every keyboard setting as a SysEx file that you can save using your sequencer or a SysEx utility, while pressing Aftertouch and Dual waits for a SysEx dump in the other direction when using the USB drivers. Together they make it far easier to save and restore various different setups.

The remote sequencer transport controls comprise ReturnToZero, Rewind, Fast Forward, Record, Stop, and Play, and PC templates are provided to set them up for use within *Cubase*, *Nuendo*, *Logic*, *Cakewalk Pro Audio* and *Sonar* on the PC (but there are none yet for Mac users).

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In Use

For most musicians buying an 88-key controller, the feel of the keyboard is arguably more important than any other function. I first tried out the UF8 keyboard at this year's Frankfurt Musikmesse show, which was ideal, as I was able to compare it directly with other weighted hammer-action keyboard controllers from companies including Doepfer, Fatar, M Audio, and Oberheim, as well as a selection of acoustic upright and grand pianos.

I personally found the feel of the UF8 better than or at least as good as any of the other controllers I tried. It has a positive feel with the right amount of initial resistance, give, and rebound. The action is sensibly not graded (ie. heavier in the lower registers, and lighter at the top) because this controller is not intended solely for triggering piano sounds. The action is reasonably light as well, and felt fairly similar in this respect to the majority of 'real' pianos I tried (I've never agreed with those that think that weighted keyboards should come with a free Charles Atlas bodybuilding course). However, opinions on keyboard actions are very personal, and while many other players who have tried the the UFs seem to agree with my view, one did report that the UF8 felt a little 'spongy', so you should try it before you buy it if possible.

Of the four velocity responses, I found the default, Norm, to be by far the most useful, as it provides the widest and most linear range of expression, although I did find it a little difficult to achieve velocity values over 70 using this mode. The Hard curve hikes the velocity values resulting from quiet playing upwards, to make it easier to get to the 'harder' end of the dynamic range, and the Soft curve makes it very easy to generate high velocity values of 70 and over even when playing softly (this is ideal for drum sounds). Wide combines both of these characteristics for aggressive results. However, if none of these responses is to your taste, then you should know that CME are hoping to release their own software utility in the future which will allow you to tweak them.

I also had the chance to try out the semi-weighted synth actions of the smaller UF models, and to me they felt similar to my Korg M1, with a light but positive action, and a sudden 'give', rather than the 'pushing down on a spring' feel of many budget keyboards. I expect them to be very popular for those who require fewer keys.

The aftertouch is excellent on all four models, allowing a smooth and controllable increase from 00 to 7F for expressive results, rather than the 'on/off' effect of many keyboards I've played. When I plugged in my trusty Yamaha BC1 breath controller, it worked perfectly with the UF8, outputting data by default on MIDI controller 2 for adding expression to suitable sounds (although it's easy to reallocate it to any other controller number, such as expression or filter frequency, for other purposes).

With its endless rotary encoder and host of assignable knobs, buttons, and transport controls, to say nothing of its weighted aftertouch-capable keyboard, the CME UF8 is ready to control almost anything in your MIDI studio.

I did initially have a hitch using my VP3 volume pedal (distributed in the UK by BCK). When plugged into the UF8, it only generated MIDI controller values from FF down to about 46. However, after a little Internet research I discovered that there are two pedal-wiring options — the VP4 and many other pedals do work perfectly with the UF8, while those whose pedals have polarity switches can solve this issue in a moment. My problem was cured after five minutes' work with my soldering iron to swap the ring and tip connections on the VP3 output jack.

The UF8's own rotary and slider controllers felt smooth and reliable, and I found the user-definable controller settings quick and easy to reallocate. However, CME should perhaps consider dropping the detents on their rotary pots, since while having a central 'click' is perfect for pan controls and the like, they do make it slightly more difficult to achieve smooth sweeps over the whole range for other parameters.

Sadly, I wasted some hours trying to get the transport controls to work in *Cubase SX* using the supplied template before I discovered on the CME forum that you have to be using the USB drivers (the standard MIDI output sends out different data from these buttons that may work with other software, but doesn't with *Cubase*). Once I'd installed the USB drivers under Windows XP, the two banks of faders worked well with the *Cubase* mixer, and all the buttons finally burst into action, although the Stop button caused the song position pointer to jump backwards. Thankfully I was able to download a revised template posted by a CME forum user that cured this problem.

Overall, while there have been a few teething troubles with both drivers and templates, I've been extremely impressed by how quickly CME are sorting them out and responding to user suggestions for new features and improvements. Even on the day I was due to send in my review, an updated driver cured two problems, so I could delete the paragraph discussing them!

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Conclusions

During my personal search for an 88-key keyboard controller, I've investigated models from Doepfer, Fatar, Korg, Kurzweil, M Audio, Novation, Oberheim, Roland, and Yamaha — there's never been more competition for the market at which the CME range is aimed, and new MIDI controllers are seemingly being released on an almost monthly basis.

If you don't need aftertouch, you may be sorely tempted by M Audio's Keystation Pro 88 (reviewed in *SOS* October 2004) with its greatly increased complement of knobs and sliders, especially at its cheaper street price (around £320 in the UK). However, if, like me, you want aftertouch, the options narrow considerably. Oberheim's MC1000, at about £450, provides the same range of basic features, but only has two real-time slider controllers and no USB support, while Doepfer's LMK2+ has no knob and slider controllers at all, and is built into a sturdy flightcase for the gigging musician, but is considerably more expensive at £750. None of these offer an optional audio interface or a breath-controller input.

Overall, CME's UF8 looks good, feels good, has responsive aftertouch, the versatility of a breath-controller input, and quite enough knobs, sliders, and buttons for most musicians. And at its bargain UK price of £430, the arrival of the CM8 means that my search for the right controller is finally over. SOS

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SOUND ON SOUND
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ESI Pro ESP1010

PCI Audio Interface [PC]

Published in SOS August 2005

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Reviews : Recording System

If you're on a tight budget, ESI's new PCI audio and MIDI interface gives you more ins and outs for less money. But how does it stack up sound-wise against its pricier competitors?

Martin Walker

ESI Professional now market a wide range of audio interfaces that cover Firewire, PCI and USB formats. Their recent Julia soundcard (reviewed in *SOS* January 2005) and RomI/O MIDI interface have proved very popular, but ESI are probably best known for their budget Waveterminal and more upmarket Wami Rack ranges. The latest ESP1010 model being reviewed here falls into the Wami Rack category, and is described as a "high-quality 24-bit 10-in/10-out PCI audio and MIDI interface with 19-inch rackmountable breakout box".

Photos: Mark Ewing

As with most '1010' products, two of the inputs and outputs are of the digital S/PDIF variety (in this case a coaxial in and out and a duplicate optical output), leaving eight analogue inputs and outputs, which are all capable of both balanced and unbalanced operation. There are also two MIDI inputs and outputs. Since this area of the market is already rather crowded it will be interesting to see how ESI manage to make their new offering stand out.

ESI Pro ESP1010 £249

pros

- Superb value for an audio interface with two mic preamps and fully balanced eight-in/eight-out analogue capability.
- Includes a digital monitoring mixer and twin headphone amplifiers.

cons

- Bulky umbilical cable.
- No power indicator on rackmount breakout box.

summary

ESI have managed the impossible with the ESP1010. Never before have I reviewed an audio interface containing so many features, and with such a good sound, for so little money.

information

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Test Spec

- ESI Pro ESP1010 Windows XP driver version 1.12.
- Hardware: Intel Pentium 4C 2.8GHz processor with Hyperthreading, Asus P4P800

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Let's Switch Again

The rack breakout box is certainly smart in its two-tone grey finish, and was robust enough for me to stand on it without worrying unduly. The line-level

Deluxe motherboard with Intel 865PE chip set running 800MHz front side buss and 1GB DDR400 RAM, running Windows XP with Service Pack 2.

■ Tested with Cakewalk *Sonar* 4.0, Native Instruments *Pro 53*, Rightmark *Audio Analyser* 5.4, Steinberg *Cubase SX* 3.0 and *Wavelab* 5.0, Tascam *Gigastudio 160* version 3.0.

analogue inputs are split between the front and back panels, with four on each, but are otherwise identical, each having a TRS-wired balanced/unbalanced socket with -10dBV nominal sensitivity and 10k(omega) impedance, allowing connection of most consumer-level signals. In addition, inputs one and two also have optional mic preamps with XLR sockets on the front panel, each with its own switch for optional +48 Volt phantom power, an unusual and welcome feature at this price. The bundled software control panel utility is used to switch between mic and line inputs.

The eight analogue outputs are all on the rear panel, again on TRS-wired sockets for balanced or unbalanced use, and again with a nominal -10dBV output level. Outputs seven and eight are also duplicated on two front-panel quarter-inch jack sockets. However, there's a novel twist — these outputs can also act as dual stereo headphone outputs. The supplied software utility can send the stereo signals from output channels 1/2 to both of these two front-panel sockets via a dedicated headphone amplifier designed for phones between 32 and 300 (omega). If you use this option, the rear-panel outputs 7 and 8 both then carry identical balanced versions of the channel 1 output, which isn't particularly useful, but having the choice of two balanced/unbalanced outputs or twin stereo headphone outputs on the front panel adds to the unit's flexibility.

The breakout box is completed by a rear-panel pair of MIDI In/Out sockets, a D-sub connector to attach the associated PCI interface card, and a socket so you can plug in a 9-12 Volt DC 300mA power supply. The latter is not included, and is only needed if you require the full +48 Volt phantom power for your particular microphones — if they can manage with +12 Volts, or you don't need phantom power at all, the entire unit can be parasitically powered from your PC.

The PCI interface card is attached to the breakout box with a heavy-duty two-metre-long umbilical cable, and also houses the remainder of the digital I/O, courtesy of a one-foot-long adaptor cable terminating in two phonos for coaxial digital S/PDIF in/out and two MIDI sockets for the other MIDI In and Out, plus a handy duplicate digital out in Toslink optical format on the card's backplate.

Overall, this is an unusual yet versatile arrangement of inputs and outputs at the price. Splitting the MIDI I/O between PCI card and rack unit will bother few people, and having four audio inputs on the four panel will suit those who are constantly plugging different input signals into their setup. As we'll see shortly, inputs 1 and 2 can provide higher sensitivity that would prove suitable for plugging in electric guitars (prime candidates for front-panel sockets), except that the input impedance is only 10k(omega), which means that doing so will result in high-frequency loss. Nevertheless, if you often find yourself cursing at having to ferret about behind a rack to plug in synths and other line-level signals during a recording session, this might be just the interface for you.

Donning my nit-picking hat, I'm not quite so convinced about the novel output switching arrangements, which effectively make the rear 7/8 outputs redundant in headphone mode. I would also have liked an indicator on the rack's front panel — there's absolutely nothing to show that it's either connected or powered up correctly — while some users may find the umbilical cable rather too short and inflexible (it's half an inch thick).

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Driving Music

Drivers are supplied that support Windows 2000, Windows Server 2003 and XP, but as usual I downloaded the latest 1.12 version direct from the ESI Pro web site. I had no problems during the installation, which detected two devices in turn, and the ESP1010 Panel utility appeared on my taskbar correctly after I'd rebooted. ESI told me that they are currently working on Mac and 64-bit drivers, but no release date has yet been set.

The current drivers use ESI's own EWDM (Enhanced Windows Driver Model) technology, and support MME, WDM, GSIF, Direct Sound and ASIO 2.0. The 'enhanced' element is that these formats are integrated into a single driver that offers full multi-client support, so you can access the drivers simultaneously from any number of applications. This integration also lets you freely patch digital signals between the various driver input and output formats using ESI's Direct Wire technology, so you could, for example, record internally from *Gigastudio* (GSIF) to *Sonar* (WDM) or *Cubase* (ASIO), or from an MME-based application like *Winamp* directly into your sequencer.

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Control Panel

The Panel utility has some similarities to the one supplied for ESI's Julia, but has had a graphic makeover, and this time spreads the controls over two pages. The Mixer Panel page houses a 36-bit digital mixer, where for each of the 10 inputs and 10 playback channels there are digital faders with a 120dB range, peak-reading meters, pan controls, and mute functions (activated by clicking on the dB readout box beneath the faders — even with the faders pulled fully down, you need to use these to completely remove a particular channel from the mix). Overall level is set in the Mix Out section by a further pair of digital faders with a stereo Link button, and monitored on stereo level meters.

Meanwhile, the Control Panel page devotes its left-hand side to inputs and its right to outputs. Inputs 3 to 8 simply have meters to monitor incoming levels, as does the digital S/PDIF input, along with an additional 'Locked' indicator that lights up if a suitable clock signal is detected. With a fixed input sensitivity of -10dBV, I found these inputs a good match for most of my hardware synths.

The ESP1010 Panel utility provides versatile control over the eight analogue inputs, two mic preamps, eight analogue outputs, monitor mixing, and headphone switching options of the ESP1010.

Analogue inputs 1 and 2 are more complex, and initially a little confusing. Each has two gain controls, the first a slider with a 60dB range operating in the analogue domain, and the second a rotary digital gain control offering up to 15dB of additional gain. The text readout at the bottom of each channel displays the combined gain value of these two controls, which can therefore range from -60dB to +15dB. There are also tiny buttons beneath each channel for switching between M(ic) and L(ine) use, which can be changed independently for each input.

In Line mode, and with the gain at 0dB, inputs 1 and 2 are identical in sensitivity to inputs 3 to 8, but when in Mic mode, an additional preamp is switched in. According to the manual you can't use the XLR and quarter-inch inputs simultaneously, but I found both are in fact still connected to the preamp in Mic mode, so you should be very careful not to leave line-level signals plugged in to the quarter-inch jack sockets when plugging a mic into the associated XLR socket.

The phantom power is also confusing. There's a tiny button labelled '12' next to each of the two Mic/Line switches in the Control Panel; these buttons are only operational in Mic mode, and activate +12 Volt phantom power. This will probably be sufficient in many situations, but may limit headroom. To activate +48 Volt phantom power you need to ignore these Control Panel switches, plug a DC power supply into the rear of the rackmount breakout box, and instead use its front-panel +48 Volt switches. As you might expect, it's not recommended to activate +12V and +48V phantom power simultaneously!

The Control Panel Output section contains digital level faders for each analogue output, once again with a 60dB range, and arranged in pairs with a stereo peak-reading meter alongside each one. The S/PDIF output has stereo meters but no faders, but there is a choice of Professional or Consumer formats via a pair of buttons above these meters.

Beneath the output 1/2 and S/PDIF Out meters are two buttons labelled Mix Out. Activating either or both of these removes the normal playback signal from these outputs and replaces it with the combined mix output from the Mixer Panel described earlier, which is handy if you want to monitor your inputs with 'zero' latency during the recording phase and simultaneously hear the outputs from your multitrack sequencer. Above output channels 7/8 is a further button labelled 'HP', which performs the unusual headphone switching described earlier. The Control Panel is completed by a small panel that lets you manually select sample rates if required, choose internal or external clock options, and view a readout of the current clock source and sample rate.

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ESI Pro ESP1010 Brief Specifications

- Sample rates: 22, 24, 32, 44.1, 48, 88.2, 96 kHz from internal clock (22 and 24 kHz not supported by digital I/O).
- Mic preamps 1/2: up to +31dB additional gain, separately switchable +12V phantom power from PCI card, with +48V available from a suitable 9V DC power supply.
- Analogue inputs: eight quarter-inch TRS balanced/unbalanced jack sockets with nominal -10dbV sensitivity (line inputs 1 and 2 have up to +15dB digital gain option).
- Analogue outputs: eight balanced/unbalanced TRS quarter-inch jacks at fixed -10dBV level, plus two stereo headphone outputs.
- Digital I/O: S/PDIF in and out on coaxial phono sockets, S/PDIF Toslink optical out, two MIDI Ins and Outs.
- Dynamic range: input 107dBA at 48kHz, output 112dBA at 44.1kHz.
- Frequency response: not stated.

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In Action

With my now traditional double-blind listening tests, I auditioned the sound quality of the ESP1010 against my benchmark Emu 1820M and Echo Mia soundcards, using a wide of material including solo vocals, guitar, drums and percussion, plus rock band, dance music, jazz trio and classical orchestra.

As you might expect on this occasion, it was easy to pick out the more expensive Emu 1820M every time for its tight, focused sound — the other two cards provided high-quality sound and each instrument appeared in the right place, but with the Emu I felt I could almost reach out and touch them. As always, these were subtle but nevertheless noticeable differences, and while the other two interfaces were slightly different, they were very tricky to tell apart reliably. However, after a prolonged listening session I decided the ESP1010 had a slight edge over the Mia for its more natural top end. When the stereo Mia was first introduced it cost as much as the eight-channel ESP1010, so this is an excellent result for ESI's latest offering.

The ESP1010's analogue inputs are divided between the front and rear panels. The optional DC power supply is not included, and is only required if your microphones need the full +48V phantom power.

Running Rightmark's *Audio Analyser* roughly confirmed the manufacturer's quoted dynamic range figures: I measured 105dBA at 24-bit/44.1kHz, and an even better 107dBA at 24-bit/96kHz, which is unusual considering that at 96kHz the upper -0.1dB fall-off point increases from 20kHz to 43kHz — normally a wider bandwidth results in a higher noise level. The low-frequency -0.3dB points were at a good 9Hz and 13Hz respectively. Total harmonic distortion was a very good 0.0006 percent, while stereo crosstalk was also good at about -106dB. Overall these are a very good set of figures for this price, and taken in conjunction with my subjective auditions, I feel that ESI have done an excellent job.

On the driver side, with my Pentium 4 2.8GHz Northwood PC I managed to drop the ASIO 2.0 latency down to 128 samples at 44.1kHz without any glitching in *Cubase SX3*, resulting in a low latency of just 2.9ms. I had absolutely no problems running the GSIF drivers with *Gigastudio*, while the Direct Sound and MME drivers proved to be fairly similar to most under I've tried under Windows XP, respectively achieving 35ms and 45ms Play Ahead settings with NI's *Pro 53* soft synth.

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Summing Up

Although there are plenty of variations on the 'eight-in/eight-out analogue plus S/PDIF and MIDI' specification already in the market, when I started to look for competitors for the ESP1010 I began to appreciate just what incredible value it actually is. Edirol's UA101 and FA101, Guillemot's Hercules 1612FW and M Audio's Firewire 1814 are all around the £400 mark, and even the budget Terratec Phase 88 Rack FW that I reviewed in *SOS* May 2005 is £360, which I described as an excellent retail price.

ESI Pro's ESP1010 may have a few design quirks, but its audio and driver performance are excellent, and it incorporates several clever twists such as the mono line/stereo headphone outputs on the front panel. It doesn't interface with the more professional +4dBu I/O levels, and nor does it have 192kHz support, but I personally know of no musicians actually using this sample rate in the real world, and even at the suggested retail price of just £249 it's excellent value for money. On the street I've seen it priced as low as £211, and I can think of no other rackmount eight-in/eight-out interface anywhere near that price. If you like a bargain, the ESP1010 will be right up your street. SOS

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SOUND ON SOUND
The World's Best Music Recording Magazine

Gforce *Minimonsta* £140

pros

- It sounds remarkably like a Minimoog.
- It's a huge-sounding, warm, and very appealing 'analogue' polysynth.
- It's capable of an unexpected range of FM timbres.
- It has unique modulation capabilities, and a huge number of them.
- Its 'Melohman' patch morphing can be extremely effective.
- If you over-stretch the CPU, the audio break-up is quite benign.

cons

- There's a bug in the contour generators.
- The ADSD contours do not emulate the Minimoog's response correctly.
- It has no pulse-width modulation or oscillator sync, and very limited effects.

Gforce Minimonsta

Virtual Analogue Synth [PC/Mac]

Published in SOS August 2005

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Reviews : Software

Forget completely flexible software synthesis — what everyone wants, it seems, is emulations of 35-year-old monophonic analogue synths! We check out the latest modelled Minimoog, and see how it compares to the original hardware.

Gordon Reid

It's a decade since the first, tentative steps were taken, but the craze for digital synths that emulate vintage instruments continues unabated. The past few years have seen intense activity in this area, with a virtual ARP Odyssey, an ARP 2600, an Oscar, a Yamaha CS80, an MS20, a Polysix and at least three virtual Minimoogs from Arturia, Creamware and Gforce. What is it about the Minimoog that inspires this degree of reverence? It can't be the facilities, because these are limited, both in terms of sound generation and performance control. No... as we all know, it's the sound. Even today, this remains unsurpassed, and if today's technology can recreate the sound of the Minimoog and couple this to the facilities we've come to expect in the 21st century, we'll have reached some sort of musical Shangri-la.

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What's In A Name?

Before getting stuck in, there's a nomenclature issue to deal with. Although everyone I've spoken to about it always refers to it as *Minimonsta*, properly speaking, Gforce's *Minimonsta* is a joint development between Gforce and French plug-in developers Ohm Force, and its full title is the Gforce *Minimonsta:*

- It has no global tuning control.
- Its patch management is cumbersome.
- It does not support DXi, MAS or HTDM.

summary

Despite a few flaws, *Minimonsta:Melohman* (to give it its full name) is an excellent Minimoog emulator that is also capable of sounds undreamt of when Dr Bob and his team were designing the original synth. If it were presented as a hardware instrument, analogue anoraks would drool so long and hard that they would have to be treated for dehydration.

information

£ £139.99 including VAT.

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W www.maudio.co.uk

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Test Spec

MAC REVIEW SYSTEM

- 1GHz Apple Mac G4 Powerbook with 1GB of RAM running Mac OS v10.2.8.
- Plogue *Bidule* v8.001 and v8.002.

All tests and comparisons were made using the stand-alone and VST versions of *Minimonsta:Melohman*.

Melohman (there's plenty on the Melohman part of the instrument later in this review). However, for reasons of simplicity and length, I'll be referring to it as *Minimonsta* throughout this article.

Even the briefest glance shows that *Minimonsta* is much more than a simple reincarnation of the Minimoog. The central section represents the original, but this is surrounded by numerous new control panels, the functions of which are not altogether obvious. So we'll start by considering those parts that emulate the Minimoog itself...

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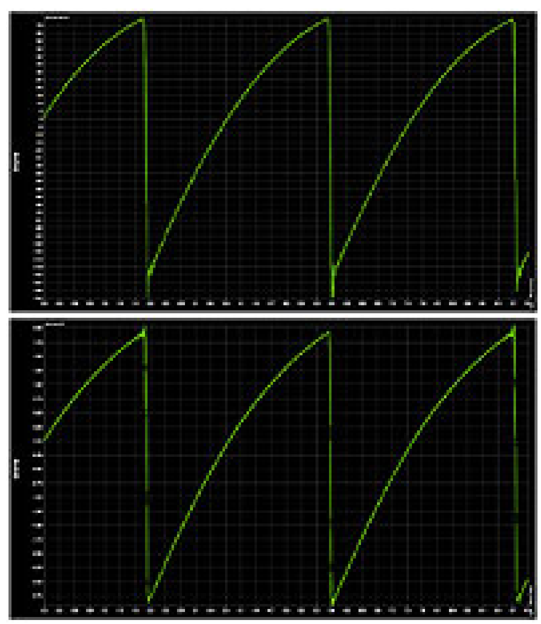
Oscillators & Noise Source

The Oscillator Bank offers three oscillators, all of which range from 32' to 2' (±8 semitones for Osc2 and Osc3) and have a 'Lo' setting for low-frequency duties. And, as on the hardware Minimoog, Osc3 doubles as a modulator, with routings via the Controllers section to the pitches of all the oscillators, as well as the filter cutoff frequency.

As always, the way to test the sound of these is to open the filter as far as possible, ensure that filter emphasis is at a minimum, defeat any modulation, and create 'organ' envelopes for both the filter and audio amplifier. This then allows us — as far as is possible on a fixed architecture — to analyse the sounds of the oscillators themselves.



I set up my original Minimoog, serial number 11285, next to the Mac I was using to host the software synth and, on both instruments, set Osc1 to a ramp wave at 8', with the level in the Mixer (which we'll come to, later) set to '5'. I then matched their pitches and output levels and compared them... Hang on. Which is which? If I didn't know which hand was playing which keyboard, I wouldn't have had a clue! The first two graphs below demonstrate this; the equivalence is clear.

The same level of accuracy is exhibited by the triangle, mixed triangle/sawtooth, square and pulse waves. At all pitches, these proved to be all but identical to my Minimoog. Unlike Arturia's *Minimoog V*, *Minimonsta* does not allow you to alter the width of any of the pulse waves, either manually or by applying modulators, but the only fault I could find (and it's trivial) is that the waveform of the ramp+triangle on *Minimonsta* appears to be inverted, as you can see from the remaining two wave traces below.

I also compared the noise generators of my Minimoog and *Minimonsta*. For both settings — white and pink — *Minimonsta* generates a much 'redder' spectrum than the original synth, with a greater low-frequency content. This is a little disappointing, because without a high-pass filter, it is impossible to match the sounds of the two, but, in the real world, I doubt that it matters too much.

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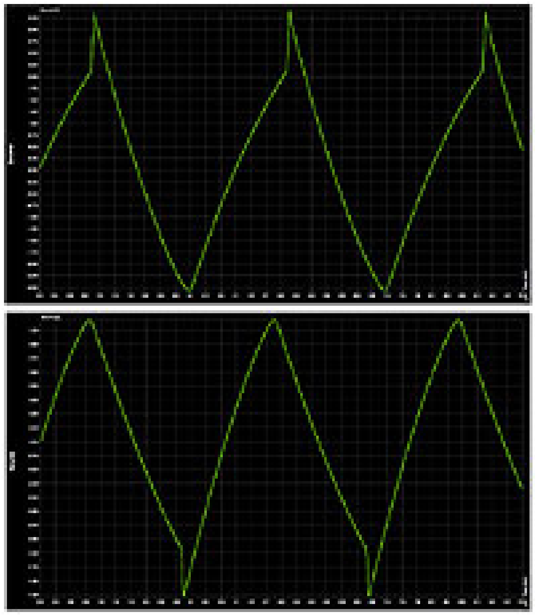
The Mixer



For comparison purposes: the *Minimonsta* ramp wave (top left), the Minimoog ramp wave (top right), the ramp+triangle wave generated by *Minimonsta* (bottom left) and the same wave generated by the real Minimoog (bottom right).

The outputs from the three oscillators, the noise generator, and any signal applied to the external signal input are combined in the Mixer section. On the original, high signal levels allowed you to overdrive the filter input to create a slight, but 'warm' distortion. Observing *Minimonsta*'s output demonstrates that this too generates a subtle distortion at high levels, without the need for any power-hungry compression or overload functions.

Encouragingly, setting two oscillators — say Osc1 and Osc2 — to the same waveform and almost identical pitch and then switching one of them on and off a few times demonstrates that the oscillators are 'free running'; each time that you switch the second oscillator on, the slight beating between the two begins at a different point in its cycle.

As you can see from the screenshot over the page, you can make use of *Minimonsta*'s External Signal Input in a VST environment. In the example shown, *Minimonsta*, Gforce's *Mtron* and Korg's *Legacy Polysix* are all loaded within my chosen VST host, Plogue's *Bidule*. I was able to play everything from my MIDI controller keyboard, and the result (including the summing of the stereo inputs to mono) was just as you might expect. If, for any reason, you wish to disconnect the MIDI signal from *Minimonsta*, you can still pass the external signals through it by pressing the Hold button in the performance panel, and playing a note on *Minimonsta*'s on-screen keyboard to open the Gate.

Minimonsta also recreates the old trick of looping either the Minimoog's High or Low output back into the External Signal Input. Only the High loop is emulated here, so the input is easily overloaded, and feedback dominates the sound unless you set it up carefully. Sonically, the results are dramatic, and you can generate all manner of dark and radical sounds this way, but if I'm honest, I found the sounds not quite as pleasing as the ones I can coax from my hardware Minimoog.

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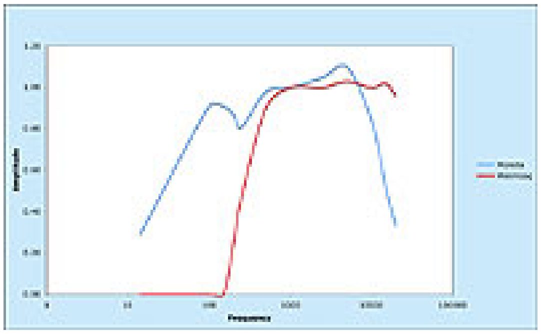
The Filter

Minimonsta's filter copies the Minimoog's, so there are knobs for cutoff frequency, emphasis (filter resonance), amount of contour applied, and the ADS [D] contour generator. To the left are three switches that apply keyboard tracking at rates of 0 percent, 'a bit', 'a bit more', and 100 percent, plus modulation (supplied by the Controllers section) as required.

The range of cutoff frequencies is displayed as 14Hz to 28,672Hz. My measurements showed that the lowest cutoff frequency is around 16Hz (which is more than acceptable). At the top end, it shot off the graph (ie. a frequency higher than 22kHz) when measured at 44.1kHz sample rate. Gforce claim that the top limit is 29kHz; I saw no reason to doubt this, although only bats will be capable of verifying it.



Minimonsta's self-oscillation amplitude compared with that of the Minimoog.

I was pleased to find that, as on a real analogue synth, the filter required no input from elsewhere to begin self-oscillating at high emphasis settings. Setting the filter emphasis to maximum, I was able to observe the behaviour of the self-oscillation thus invoked. I'm afraid that, like all others before them, Gforce have missed a trick here, because self-oscillation continues at a significant amplitude all the way down to sub-sonic frequencies whereas, on vintage Minimoogs, the Emphasis peters out as you approach the bottom end. When the filter is not oscillating, this contributes to the 'punch' of the Minimoog because, as you play at lower and lower pitches, more and more of the bass frequencies pass to the output. So, the *Minimonsta* filter — while very good — does not quite emulate a real Minimoog correctly at the bottom end.

To investigate this further, I set up the self-oscillating filters on *Minimonsta* and my Minimoog to generate an output of 1V at 1kHz, measured the output over a range of frequencies, and then plotted the results. You can see them in the graph above.

It was while making these measurements that I discovered something odd about *Minimonsta*'s 'Main Edit' readout. The frequencies being reported here are as much as 10 percent high or low. To prove this, I made the filter to oscillate at (ostensibly) 440Hz and then switched on the 440Hz reference oscillator on my Minimoog. The *Minimonsta* was sharp, and I had to reduce the figure in the readout to 406Hz to obtain A440. Weird!

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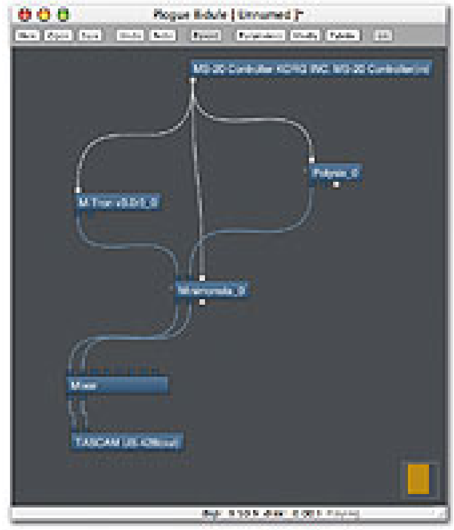
The Contours



Using *Minimonsta* as an effects unit.

As we all know by now, the Minimoog generates four-stage contours (as seen in the left-hand diagram over the page) using three controls: Attack, Decay and Sustain. The fourth stage, Release, is created by flipping a switch to reapply the Decay time as a Release time. However, as you'll know if you read my recent reviews of Arturia's *Minimoog V*, the Minimoog's contours generators exhibit a quirk that contributes greatly to the character of the instrument: the envelopes are reinitialised from the points to which they have dropped during the previous Release. Consequently, successive contours 'climb' (as shown in the right-hand diagram over the page), so you can use keyboard technique to make the sound bright or softer, or louder or quieter, as you wish; a critical element in the character of the instrument. To be honest, I'm not sure whether Gforce have had a stab at emulating this or not. Some patches seem to get a little brighter when played rapidly, and it sounds subjectively as if the self-oscillating filter goes a little sharper if you play very quickly. But even if this is the case, the response is definitely not the same as that of a Minimoog, and objective measurements show that, at best, it's an almost negligible effect. In fairness, though, I've yet to encounter a software synth that *does* emulate this aspect of the Minimoog correctly. Arturia improved their attempt at it no end between versions 1.0 and 1.5 of *Minimoog V*, and it shouldn't be hard for Gforce to get it right. I'm hoping that they will look into doing this for their next revision.

Another important aspect of the Minimoog is the speed and precision of its envelopes. You can observe this by setting A=0, D=0 and S=0, and then displaying the resulting click when you play a note. *Minimonsta* doesn't produce this click; you have to send a signal through the filter and amplifier to obtain one. I created one in this way by leaving the filter wide open, sending white noise down the signal path, and turning all the amplifier contour knobs to zero. The main body of the click lasted just four to five milliseconds, which is in line with the Minimoog itself. However, there is low-level noise after the click, and this continues *ad infinitum* if you keep the key depressed. What's more, this even occurs in real patches with D=0 and S=0, sustaining notes when there should be silence. Normality is restored if you increase the Decay beyond a few tens of milliseconds, but there's definitely a bug here.

Before moving on, you'll want to know the (unmodulated) maximum contour times. Contrary to the figures in the *Minimonsta* manual, these are nine seconds for the Attack, and 16 for the Decay. These are identical to the times generated by my Minimoog.

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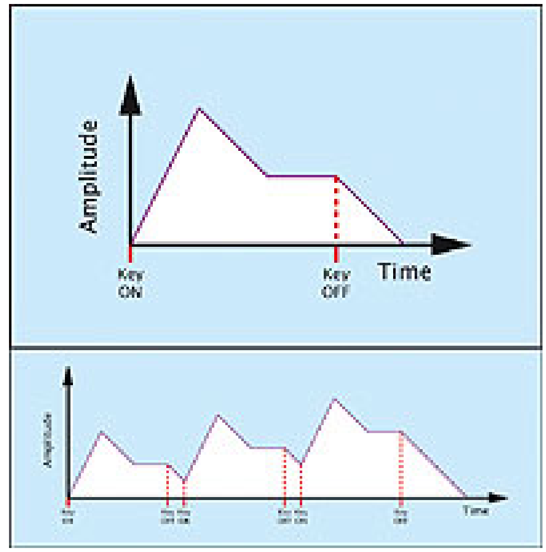
Controllers & Performance Panels



A single ADSD contour (above left) and the contour generated by a real Minimoog played rapidly (above right).

The Controllers panel emulates the Minimoog's precisely, with a tuning knob, a mixer that allows you to select between Osc3 and noise as the modulation source (or any mix of the two), and a portamento (glide) time control. The maximum glide time is rather too short for my liking; less than two seconds from one extreme of the keyboard to the other, whereas my Minimoog takes about five seconds to do the same. Happily, the minimum glide time is right (ie. zero) and there is no zippering or bumpiness as the pitch slews.

Creating FM sounds by shifting Osc3 into the audio range and applying Osc3's triangle or ramp waves to Osc1 and Osc2 produces sounds that are all but indistinguishable from those of my Minimoog. Given how sensitive FM can be to small changes in the modulator's amplitude, this was not the result that I had expected! Select the sawtooth or any of the pulse waves and the result is different, although only in pitch, not timbre (this shifting of pitch when you select different modulator waveforms is one of the Minimoog's quirks, and while *Minimonsta* emulates it, the resulting pitches are different on the two instruments).

Below the Controllers lie the performance controls. The Decay On/Off and Glide On/Off switches act as you would expect, as do the pitch-bend and modulation wheels. There are two differences here. Firstly, you can set the amount of pitch-bend from plus or minus two semitones to ±24 semitones. Secondly, unlike on the original synth, this value applies equally to increases and decreases in pitch. This is one area in which I'm very happy to have less-than-perfect authenticity!

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Minimonsta Specifications

No specification is included in *Minimonsta*'s manual, so I have compiled the following abridged version.

AUDIO-FREQUENCY SOURCES

■ Oscillators: Three per voice.

■ Waveforms per oscillator: Six.

■ Audio ranges: 32' to 2', plus 'Lo' mode.

■ Fine-tuning: Plus or minus eight semitones.

■ Pulse-width modulation: No.

■ Oscillator sync: No.

■ Noise: White and pink.

■ External signal input: Stereo (summed to mono).

■ Polyphony: 1 to 32 voices, depending upon processor power.

■ Unison: 2 to 32 voices, depending upon processor power, with detune.

■ Glide: Maximum two seconds; minimum near-instantaneous.

CONTOUR GENERATORS

■ Standard envelopes: Two Attack, Decay, Sustain (ADS) types with optional Decay to 0V.

■ Minimum Attack: Measured approximately 1ms

■ Minimum Decay/Release: Measured approximately 4ms (but see notes in main text).

■ XADSR envelopes: 36 (I think) Attack, Decay 1, Decay 2, Release (A/D1/D2/R) types.

LOW-FREQUENCY SOURCES

■ Oscillators 1, 2, and 3: 'Lo' mode.

■ Dedicated LFOs: 36 LFOs (I think) offering 10 waveforms with delay, S&H, smoothing, tempo and 'note on' sync.

FILTER

■ Profile: 24dB-per-octave.

■ Resonance: Zero to self-oscillation.

■ Filter tracking: Zero to 100 percent.

KEYBOARD RESPONSE

■ Triggering modes: Single, Multi.

■ Key priority: All notes.

EFFECTS

■ Stereo Delay: With independent left and right times, Regeneration, and Amount parameters, but no MIDI sync.

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Extending Perfection?

Inevitably, *Minimonsta* follows the trend set by other 'virtual analogue' emulations, adding many features not found on the original synth. This is where the extra on-screen panels come to the fore. Click on almost any knob or switch on the main control panel, and its name will appear in the 'Parameter Name' window in the MIDI panel. You can now select a Modulation Source from a drop-down list of 21 options (shown over the page), determine its polarity, select a response curve, and set the maximum amount of deviation using the Sensitivity knob.

Unfortunately, when you control a parameter in this fashion, the knob does not move, nor does the switch flip, so you have no indication of the current value or state of the parameter. This is horrid. But tucked away in the Set Up pages lies an option called Auto Bind. Tick this and click on any *Minimonsta* parameter, and the next physical control that you twiddle on your hardware controller will be allocated to that parameter. Keep doing this until you have set up everything as you wish, and then un-tick Auto Bind. That's better! What's more, the on-screen knobs and switches move when you control them. Oh yes... and you can save and restore configurations for use with different controllers. That's *much* better!

To the left of the MIDI controller section, you'll find the LFO and XADSR section. The LFO offers 10 waveforms, delay, sample & hold of the selected waveform, smoothing, and Note On synchronisation, while the bi-polar, velocity-sensitive contour (which is not a conventional ADSR, but an 'Attack, Decay1, Decay2, Release' contour with a decaying Sustain period, much like a piano) has misnamed Attack, Decay, Sustain, Sustain Time and Release parameters.

Good though this appears to be, a quick glance at the controls fails to tell you the most important thing about this section. *Minimonsta* doesn't have an additional, assignable LFO and an additional, assignable contour generator; it has a dedicated LFO and a dedicated XADSR for *most* of the parameters on the main control panel! What's more, every control in every LFO and XADSR can be assigned to a MIDI controller. This is powerful stuff.

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How Hungry?

If you believed Gforce's published suggested Mac CPU requirements for *Minimonsta*, you'd think it was quite a power-hungry beast. For PC hosts, the minimum suggested requirements are tame indeed (a mere 64MB of RAM, any OS from Windows 98 onwards, and a Pentium 4 processor — although a suggested speed is not given). On the Macintosh, however, they suggest a 1GHz G4 with 256MB of RAM as the bare minimum, and recommend use on a dual-processor 1.8GHz G5. This seems a bit excessive in my experience. All the tests I made, and all the sounds I created, were on a single-processor 1GHz G4. With a typical three-oscillator patch, and numerous other applications running concurrently, I could obtain up to eight-voice polyphony on the stand-alone version before relatively inoffensive digital break-up occurred. That I was doing so with just 128-sample latency at 44.1kHz (about 3ms) was even more encouraging. Mind you, using my Powerbook's internal D-A converter resulted in poorer performance, although this could be improved by switching the audio input source to 'None'.

While we're on the subject of specifications, it seems curious that *Minimonsta* is available in stand-alone, VST, and RTAS formats on both platforms (and also AU on Macs), but not DXi, MAS, or HTDM.

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Delay & Output

At the end of the signal path lies the single effect — a stereo delay with independent delay times for the left and right channels, but common feedback (regeneration) and amount controls. All four controls can be modulated using *Minimonsta*'s LFOs and XADSRs as well as a range of MIDI controllers, so some simple chorus effects can be obtained. You can even create a 'pseudo-reverb' using two short echo lengths and lots of regeneration.

Minimonsta's MIDI modulation options.

Finally, there are knobs for output level and pan, which are also modulatable. Note that the Output Level control comes before the Delay unit in the signal path. This can give unexpected results if you modulate the output level, so beware.

Depending upon how and into which software host you launch *Minimonsta*, you may also see two extra control panels; a narrow strip to the right of the instrument (shown below left), and another above, which you can see at the top of the main screenshot at the head of this review. There's nothing to explain these in the manual, so I contacted Gforce, and experimented a bit to discover their uses. Working out the one shown on the left isn't too difficult; it has an input level control, an external/internal signal mixer, and an output level control that acts upon the whole signal. But surely these are irrelevant... aren't they?

The answer lies in the upper of these two extra menu bars. This is a shell developed by Ohm Force for their effects, and it allows you to process audio files through *Minimonsta* if you wish to use it as a stand-alone filter/effects unit. Press the Menu button, and you have the option to Open or Close a sample file. Once you have selected a suitable file (I found that WAVs and AIFFs worked well) you can play and stop, loop, adjust the tempo to fit your needs, and adjust the playback speed to synchronise with (for example) *Minimonsta*'s LFOs and delay times. This is an interesting set of facilities that allows you to use *Minimonsta* as a filter bank and a beat slicer, and has all manner of other uses.

Unison and polyphony are controlled by a panel in the upper right of the instrument. Most of these controls are self-explanatory, but it's worth noting that Mono mode offers simultaneous high- and low-note (ie. most recent) key priority, thus retriggering the sound whenever and wherever a new note is played, while Legato is a monophonic mode that only retriggers after all previous notes are released. Also, it's not stated anywhere that the monophonic Unison mode evenly distributes the number of notes selected in the Polyphony setting across the total range selected, up to a maximum of +2 semitones to -2 semitones.

The Delay and Output stages.

Now, what about that MS/Beats button? This allows you to switch between beats/bar and millisecond displays for the LFOs, XADSRs and the delay unit. However, you may have to set the tempo manually in the upper menu bar even when set to beats/bar, because not all versions of *Minimonsta* respond to MIDI Clock. This means that sync'ed LFOs and delays will not keep time with your music if there's a tempo change, unless you load a new sound with appropriate changes to the appropriate parameters. Gforce claim that the LFOs and other appropriate parameters will — when set up correctly — follow tempo changes in *Logic*, *Sonar* or *Cubase*, but I wasn't able to test this before this review went to press, as I'm a *Digital Performer* user.

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Memory Structure & *Melohman*

Once you've got your head around everything that *Minimonsta* has to offer as a sound source, there are two more elements you're going to have to conquer before you can use it to its full. These are the weird patch storage system and the *Melohman* morphing system. Let's start with the storage.

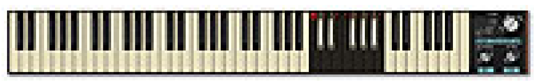
Having created a sound you like, you might expect to store it as an individual patch, but that's not what happens. First you name it in the Patch 1-12 window, and then press the Mem button on the right above the keyboard, followed by one of the buttons '1' to '12' to assign it to a key in what looks like a representation of an octave C-to-B (alternatively, you can save it to all 12 locations simultaneously). Next, you save the whole group of 12 patches (called a 'Meta-patch') by pressing Save, whereupon you are asked to provide a path (folder) and a unique name. It is the latter that appears in the Meta-patch Select window later on. If, as I did on many occasions, you only commit the sound in the Mem stage and do not proceed to the Save stage, then as soon as you click on another patch or load another Meta-patch, you'll lose all the work that you've put into your sounds.

Having saved your 12 patches (now named Presets) in the Meta-patch, and having saved the Meta-patch to disk, you can now recall both from their respective windows, as shown in the screenshot above. You can also recall Meta-patches alphabetically with MIDI program changes if you have previously assigned the list using the curiously titled RTFM button — I can only assume that the designers didn't know what that acronym means to most of us! However, you can't move Presets from one Meta-patch to another. Nor does *Minimonsta* offer you the facilities to move or delete Meta-patches; you can only do so within the computer's operating system. This is a pretty big oversight.

Meta-patches and Presets.

The reason for this very cumbersome system (although not its deficiencies) is explained when you look at the last element in *Minimonsta*, the Melohman section which gives the instrument half of its full name. The keyboard screenshot above shows the Melohman controls and *Minimonsta*'s virtual keyboard, which, as I have configured it (using the Octave knob to the right of the keyboard), has a reversed octave from C6 to B7. The keys in the reversed octave relate to the 12 Preset buttons (which explains why they are arranged in that fashion) and, in Morph mode, pressing any one of these keys selects the Preset corresponding to that location.

Still with me? Well, there's more... If the Morphing Time in the Melohman panel is set to anything other than zero, *Minimonsta* will morph — rather than switch — between patches, stopping (if it has not reached its destination) at whatever parameter values it has reached when you release the key in the Melohman octave. The speed of morph can be made velocity-sensitive, and this can be extremely effective, especially if you're using a five- or six-octave controller that allows you to control Melohman from the keyboard and still have a sufficient playing range. But be aware that you have to programme your patches with identical switch positions to make the morphing effect useable. If you don't, you'll get nasty glitches when, for example, oscillators jump from one octave or waveform to another.



The Melohman section.

There are four more morph modes: Morph Back & Forth, Morph Sequence, Mutate and Mutate Partial. 'Morph Back & Forth' returns the Preset to its initial state (ie. the first Preset selected) when you release the destination key. 'Morph Sequence' steps through one of nine predetermined Morph sequences that you select using the Melohman C to G# keys. 'Mutate' applies quasi-random changes to a Preset's parameters, the amount of which depends upon which key you press in the Melohman octave and, finally, 'Mutate Partial' provides randomisation within strict boundaries, again determined by which key you press.

If you're scratching your head at this point, I don't blame you. Melohman is initially confusing even if you have *Minimonsta* on screen in front of you. But, once mastered, you can use it to create timbral animation that would be almost impossible to obtain by other means, and the randomising system occasionally throws up interesting new sounds.

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Zipping It

All digital synths — whether hardware or software — suffer from the problem of limited parameter resolution and 'zipper noise' to some extent. For example, turning *Minimonsta*'s resonance to maximum and sweeping the cutoff frequency knob on an external hardware controller reveals the stepping introduced by the limited resolution of MIDI (approximately one semitone per step). However, turning the knob on the on-screen representation results in smaller changes, and typing values into the Main Edit screen allows finer movement still.

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Sounds & Problems

First things first: *Minimonsta* is a remarkable emulation of the Minimoog. Of course, it doesn't sound identical to hardware Minimoog number 11285, but, as I've stated before, neither have any of the other real Minimoogs I've owned! If there's a single notable difference between *Minimonsta* and the hardware it seeks to emulate, it lies in the nature of the filter emphasis, which imparts a different character to resonant sounds, particularly in the bass. A second difference, almost as significant, lies in the response of the envelopes when playing rapidly. Nonetheless, it would be silly to claim that *Minimonsta* sounds unlike the original. With a bit of fine-tuning, I think that it could be almost indistinguishable.

As for the sounds themselves... Having the original Minimoog patch book to hand, I checked Gforce's recreation of the 'factory' sounds against #11285. These were far from identical. In addition to the differences in filter resonance and contours, this is largely a consequence of knob calibration: the same settings on the two instruments do not give the same results. Qualitatively, though, *Minimonsta* was not better or worse... just different.

I then programmed a wide selection of new sounds — string ensembles, numerous pads, poly-brass and solo brass patches, weird effects and 1950s' sci-fi soundtrack-type sounds — and none disappointed. Then there was the unexpected bonus of complex FM timbres... When I realised that I could control Osc3's pitches and levels with independent XADSRs, treating it as a modulator, and similarly control the pitches and levels of Osc1 and Osc2, using them as carriers, I found myself straying well into '*DXmonsta'* territory! As for lead synths and bass synths... imagine what you can do with, say, half a dozen Minimoog voices in unison, but detuned for added fatness.

On the other hand, *Minimonsta*'s sound engine is not all-encompassing. The omission of user-definable pulse-widths and pulse-width modulation limits things in the string/pad area, while the lack of oscillator sync imposes limitations on pianos as well as lead and bass sounds. Then there's the lack of a dedicated chorus, and of an arpeggiator, both of which are found on other software synths. Another minor disappointment is *Minimonsta*'s inability to combine polyphony and unison. This is a common limitation on software synths, but it would be nice to wave it goodbye.

Now, let's flag four operational annoyances. Firstly, there's no global tune control, so retuning has to be on a patch-by-patch basis. I tried to use *Minimonsta* to add strings and lead sounds to an existing recording that was just a tad sharp of A440, and editing and saving numerous duplicate patches was a real pain. Secondly, *Minimonsta* can lose track of its input and output devices, so you sometimes have to invoke the setup page to reconnect them. Gforce are aware of this, so I expect there will be an upgrade at some point in the future. Thirdly, the switch graphics are unclear and ambiguous, so I often found myself double-checking their state. Fourthly, the patch memory system provides no ways to perform simple housekeeping duties such as moving and deleting, or even copying Presets from one Meta-patch to another.

Nevertheless, in the weeks that I used *Minimonsta*, I encountered just one significant problem. One evening, I loaded the software only to find afterwards that I had disconnected my USB controller and audio I/O. I plugged these in and, as expected, *Minimonsta* failed to recognise their existence, so I quit the program and launched it again. But *Minimonsta* refused to load. Eventually, with the assistance of Gforce and Ohm Force, I tracked the problem down to corrupted *Minimonsta* Preferences in my Mac's System Library — deleting them solved the problem. This problem did not recur, and Gforce's technical assistance was first class.

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Red, White & Blue

To make things clearer, some of the white 'silk screening' of a *Minimonsta* knob lights up in blue (not unlike an Anniversary Edition Bob Moog Voyager) when controlled by an LFO or XADSR. Similarly, parts light up in red when controlled by MIDI, as do the LEDs associated with various switches. If things are getting *really* complicated, the two coloured sections are complementary, so the surround of a knob can be both red and blue. This is a boon if you are trying to track down why something is happening to a sound when you think that it shouldn't.

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To Buy Or Not To Buy?

Notwithstanding the noted inaccuracies in its filter resonance and contour response, the sound of Minimonsta is very close to that of the original synth. But more than this, it is indeed a monster, capable of warm and engaging polyphonic sounds, and monstrous multi-oscillator, detuned monophonic sounds. With the exception of the bug in the contour generators and its occasional refusal to load, it seems problem-free, and despite its hunger for CPU power, it was well behaved on what is, by 2005 standards, a lesser-powered Mac.

Minimonsta has staunch competition in the form of Arturia's revised *Minimoog V*, which offers a different approach, but, in its latest revision, really captures the essence of the Minimoog sound. Whether Melohman patch morphing, almost perfect waveforms, a huge number of LFOs and a huge number of contour generators appeals more than a more accurate contour response, a mod matrix, extended waveform facilities, a dedicated chorus, and an arpeggiator, is, of course, down to you. But if you require DXi, MAS, or HTDM versions, the choice is taken away from you; only *Minimoog V* supports these, so maybe you don't have the luxury of making that choice. Whichever you choose, the classic Minimoog sound is now within reach for a fraction of the price of the original, as is that persistent dream of the past 35 years, the Polyphonic Minimoog.

In conclusion, while I've flagged a number of deficiencies and annoyances in *Minimonsta*, I feel that — as with the Minimoog itself — its value is determined not by the number of facilities it offers, nor by its problems; like its inspiration, *Minimonsta* is defined by its sound. Forget the fact that it's software and that it needs another revision or two to reach its full potential... these are good times for synth *aficionados*. SOS

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pros

- Compact but well-built.
- Plenty of physical controllers — and a three-zone master keyboard!
- Flexible audio I/O and zero-latency monitoring option.
- Large, bright display.

cons

- Only a single Firewire port.
- Excellent *Enigma* editor not included on driver CD (although it's only a download away...).

summary

A joint Firewire-based controller and interface could have been a muddled, hard-to-use flop, but the Ozonic is a top effort — no mean feat for the first in its class. It's a great buy and a flexible tool for potential users, as well as a hard target for the competition to hit.

information

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Test Spec

M Audio Ozonic

MIDI Controller/Firewire Interface

Published in SOS August 2005

Print article : Close window

Reviews : MIDI Controller

M Audio have been producing impressive, inexpensive controller keyboards for some time, and have recently moved seriously into Firewire interfacing. The new Ozonic seeks to combine both capabilities in one product — and all for less than £400...

Derek Johnson

Photos: Mark Ewing

M Audio have offered combined MIDI controller keyboards and USB-based audio interfaces for a couple of years now, but their recent efforts have been directed at producing Firewire audio interfaces rather than USB ones, so perhaps it's not surprising that their latest combined MIDI controller/audio interface runs on Firewire.

The Ozonic offers a 37-note velocity-sensitive, aftertouch-equipped full-sized controller keyboard capable of addressing an 11-octave note range with the help of a Transpose button. Onto this is grafted a flexible hardware control surface — assignable wheels, buttons, knobs, sliders, and a joystick — which can be configured to tweak and edit any MIDI instrument, mixer or effects device, whether in hardware or software.

The final piece of the Ozonic puzzle is a four-in, four-out audio interface capable of handling audio at up to 24-bit, 96kHz resolution. The whole lot can be buss-powered via the Firewire connection, if your computer can handle it, but a PSU is provided for those machines that can't (any with four-pin Firewire connectors, that is). And if that sounds like a lot to be editing and interfacing with via a two-line LCD, you'll be pleased to know that you don't have to: Ozonic has the same large, bright LCD as M Audio's Keystation Pro 88, and also offers access to computer-based editing via the company's free *Enigma* editor (see the box

MAC REVIEW SYSTEM

■ 450MHz Apple Mac G4 with 896MB of RAM running Mac OS v10.3.9.

PC REVIEW SYSTEM

■ 3.06GHz Pentium 4 PC with 1.25GB of RAM running Windows XP.

■ Propellerhead *Reason*, Steinberg *Cubase SX*, Ableton *Live*, Cakewalk *Sonar.*

■ Ozonic version reviewed: v5.10.0.5034 (PC drivers), v1.5.1 (Mac drivers).

opposite). There's even a cut-down version of Propellerhead's *Reason* as part of the package (more on this in the box at the end of this article), so you'll be able to start getting some sounds out of your computer, even if you don't have any other software! M Audio finish off the bundle with a quality six-pin to six-pin Firewire cable, a printed *Quick Start* guide, and CD containing the full manual as a PDF and drivers for both Windows XP and Mac OS X. Other flavours of Windows are not supported, and your Mac will need to be running version 10.2.8 or 10.3.4 (or higher).

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Digging Deeper

M Audio's recent USB-equipped controllers, such as the O2, have offered an edit mode that hijacks the controller's main music keyboard, turning black and white keys into parameter edit and value switches, and so it is here; faint screening above Ozonic's keyboard indicates which keys have which function. This is a great way of doing things and enables the controller's operating system to be streamlined and accessible. Below the display are six clearly labelled function buttons, with the Edit button enabling the keyboard-based options I've just explained. The Bank Select button chooses one of two banks of 10 preset configuration memories, which are selected by one of the 10 buttons flanking the joystick. The supplied presets include useful profiles for several *Reason* devices, General MIDI, Native Instruments *B4* and *Pro 53*, and Steinberg's *Halion*, but you can edit and save configurations at will.

The useful 'Control Mute' button, accessed by holding down the third and fourth button under the display, disables the transmission of controller data. This is a great option for use on stage, since it means that no data is transmitted if the controls are accidentally nudged. It also means you can set the knobs and sliders to a desired neutral position before starting to edit a particular device. And of course, you can transmit a snapshot of the current controller positions by pressing down the second and third buttons under the display simultaneously.

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Enigma Variations

As relatively easy as Ozonic might be to edit from its front panel and large display (it's certainly easier than the two-line displays many of us have put up with for years), if you're after something even more flexible, your wish is granted by M Audio's free *Enigma* application, available to registered users as a download from the company's web site (see www.maudio.co.uk/index.php?do=enigma.register).

This is a tidy little application, which works on several M Audio and Evolution products. It provides an on-screen graphic of the keyboard, and you simply click on any assignable physical controller (knob, slider, joystick and so on) to bring up a controller menu, inside which you can make your controller number assignment.

The job is made even easier by a massive list of software instruments, each of which has all of its parameters ready defined for you to simply drag and drop onto the Ozonic display.

Enigma also allows you to create banks of 20 configurations which you can download to the keyboard at any time, thus overcoming its 20-configuration limit. All configurations can be named in the software, although the Ozonic display, large as it is, has no facility for showing these names, and this is a shame given the LCD real estate available! Still, this is a minor quibble on a great piece of free software.

M Audio's free *Enigma* makes editing Ozonic controller assignments easy — you can just drag and drop controller assignments from the surrounding menus on to the Ozonic graphic.

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In Control

The Ozonic has a busy front panel, a fact emphasised by M Audio's decision to number the controls and buttons which transmit MIDI data, revealing that there are no less than 40 individual controllers on board. Between them, virtually any MIDI data can be transmitted, independently on any MIDI channel if desired. The knobs and sliders, for example, can handle controllers, RPNs, NRPNs, System Exclusive data, program changes and bank-select messages. Buttons can transmit note ons, program changes, MIDI Machine Control functions (including standard transport controls) and SysEx messages, and toggle MIDI notes and continuous controllers on and off. If the numbering of the controls on the front panel seems to skip a few in the mid-20s, you're seeing right. However, the back-panel Sustain and Expression pedal sockets, which can also be configured to transmit custom MIDI data (the former as a switch, the latter as a controller), are counted as numbers 26 and 25 respectively, and the aftertouch generated by the keyboard is counted as number 24, since this can also be customised. It doesn't have to transmit only aftertouch data, either.

There are nine sliders, rather than the more usual eight, but if you think of them as a possible eight channel faders plus a master fader in a mixer configuration, everything starts to make sense. An assignable button can be found under each slider, and these can function as mute controls on your imaginary mixer if you wish. Likewise, the collection of eight knobs next to the sliders could be assigned to panning duties if you wish, and the five buttons below these are clearly designed to be configured as sequencer transport controls (they're labelled as such), although of course they, like all the other controls I've just mentioned, can be assigned to any parameter you like on a target hardware or software MIDI device, depending on your particular requirements.

The pitch-bend and modulation wheels are counted as controllers 22 and 23 respectively, although as you might be expecting by now, they're not confined to transmitting pitch-bend and modulation messages. And no less than four controllers can be assigned to the joystick — one at the end of each axis (designated 18 to 21; this is very faintly screened in grey below the joystick). This arrangement affords you some pretty serious real-time sound-design potential. For example, I assigned joystick control elements to parameters such as FM amount, Oscillator one and two's phase-offset amount and filter frequency and/or resonance in *Reason*'s *Subtractor* synth, with particularly worthwhile results. This sometimes led to distortion (often no bad thing in this context), but it also opened up subtle, rich variations in my sounds, with a wavetable-like edge.

Ozonic's rear panel offers all the connectors you'd expect of a Firewire interface combined with those of a controller keyboard: a standard Firewire connector, five-pin MIDI I/O, sustain and expression pedal jacks, and the balanced and unbalanced analogue I/O, complete with high-impedance guitar input, phantom-powered XLR, and headphone jack.

Beyond its control surface, the Ozonic has some surprisingly sophisticated master keyboard options. It's equipped with three zones, each of which can have its own note range, MIDI channel, octave setting and transpose value. In addition, a number of general Ozonic controllers can be set independently for each zone; these include program change/bank select messages, pitch-bend and modulation wheels, aftertouch and the pedal assignments. Not bad for such a compact device.

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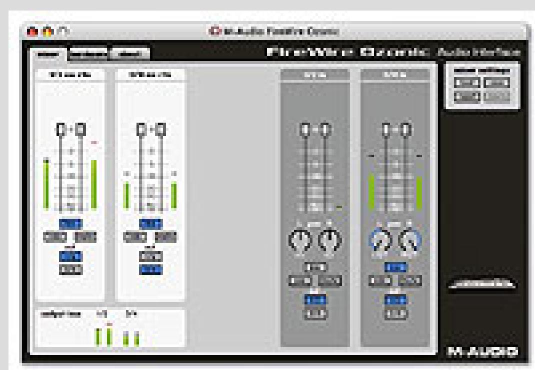
M Audio Ozonic *Control Panel*

Where would the modern digital audio interface be without its mixer/ control panel application? The one supplied with Ozonic is straightforward enough, offering control over the incoming audio, audio returning from the host software, and finally, over the two output pairs. There is full metering, plus Mute/Solo options. Stereo pairs can also be linked or disabled for mono operation, although linking inputs one and two (the XLR and instrument-level inputs) might not make much sense. The inputs also have panning controls.



The basic mixer page in Ozonic's software *Control Panel*: see what's going where, complete with metering.

Further windows keep you advised of the current sample rate (and buffer size on the PC), along with the version numbers for software, the dates for the firmware, and so on.

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Audio I/O

There is just enough audio connectivity to cope with Ozonic's Firewire-based audio-interfacing capabilities, all of it ranged across the back panel (see above). As you may recall, there are four 24-bit/96kHz-capable inputs. The first offers an XLR connector, complete with mic-preamp circuitry and a phantom-power switch (an LED on the top panel lights up to indicate when phantom power is active), whilst the second is an high-impedance instrument jack, suitable for passive guitars and basses. Both have their own input level controls, arranged as a dual-concentric stack on the front panel; signal and clipping LEDs finish off their facilities. The remaining two connectors are unbalanced line-level jacks, and you're free to use these as a single stereo or dual mono input.

Four outputs complete the audio I/O facilities, providing two complete stereo output streams from your computer. And the outputs are configured as two stereo pairs, A and B; one (A) is on balanced jacks, and the other is unbalanced. The main level of the output audio is governed by a pair of sliders just above the mod and pitch-bend wheels; a separate knob controls the balance of outputs A and B in the headphone jack on the Ozonic's rear panel. This useful function could work well in live and DJ situations, where one audio stream needs to be monitored while another is being played out to the PA.

A general headphone level knob and a monitor level control complete the Ozonic audio hardware. Some of the labelling of these controls may seem a little confusing at first, but their functions become clear within a short time of powering the interface up, or by having a look at the manual. Most importantly, the 'monitor' knob should more correctly be called a direct monitor level control. With this, you'll be able to monitor any incoming audio from the four inputs *directly* through the main audio output, before it's passed to and from your computer. Of course, this means you have the option of instant zero-latency monitoring whilst recording and overdubbing audio in your audio software of choice; you just need to remember to disable track monitoring in your software. It's a great option, simply implemented.

All of the audio connections are at the rear, where there's also a five-pin MIDI In/Out pair and the all-important Firewire socket for connection to your computer. The implications of the five-pin pair are that the Ozonic can be used as a general MIDI controller even if you don't have a Firewire connection handy, and also that the MIDI connections are available on top of all the other facilities when the interface is hooked up. In other words, MIDI can be routed into your software and sent back out to MIDI-equipped hardware at the same time as the Ozonic is functioning as a MIDI controller and audio interface.

I was a little taken aback by the single Firewire connector. In many circumstances, this needn't be a problem, especially if the host computer has two or more Firewire connectors of its own. But those users with a laptop that might only be equipped with a single Firewire connector may find themselves compromised: if you want to record your audio files to an *external* Firewire drive, there can be some circumstances where it's better to daisy-chain the drive from the audio interface, so that the interface is first in line. Placing the hard drive *before* the audio interface has the potential to interfere with the audio and MIDI data streaming in and out of the interface.

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Reason Adapted Express

Given that M Audio are the distributor for Propellerhead products in several territories (including the UK), it's no real surprise that a version of *Reason*, the virtual electronic studio, is included with Ozonic. The two make for a perfect pairing, though, and anyone just starting out will get a rousing first impression of making music with their computer, even from this cut-down version.

Reason Adapted Express comes with just the basics: its virtual rack offers one *Remix* mixer, one *NN19* sampler, one *Subtractor* analogue synth, one *Redrum* drum machine, one *RV7* reverb and one *DDL1* digital delay. You can add no more devices.

A patch library is supplied, though, for *NN19*, *Subtractor* and *Redrum*, and you can create your own from scratch; there's also full automation via *Reason*'s linear sequencer. You'd be surprised at the mayhem that can be achieved with just this setup! Don't forget that you can load your own samples into *NN19* and *Redrum* — and in the latter case, the samples don't have to be drum hits, despite the instrument's name. You can use loops if you wish (as long as they're at the right tempo for the overall song) or sound effects. You can even export *Reason Adapted* audio and re-import it for triggering in *Redrum* or *NNXT*.

Such flexible thinking will really get you in the mood for the full version of *Reason*. And of course that's the main idea of *Reason Adapted* — to tempt you with what's on offer, and provide an upgrade on favourable terms.

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What Do *You* Think?

I had a great time with the Ozonic: the controls were a doddle to customise, and it seemed to fit in with every bit of software I tried, handling both audio and MIDI with ease. The bundled *Reason Adapted* package obviously behaved itself, but so did the full version 3 of *Reason*, Steinberg *Cubase SX* (running a host of virtual instrument plug-ins) and Ableton *Live* all on both Mac and PC, and Cakewalk *Sonar* (on PC only, naturally).

Audio performance was great — mic, instrument and line inputs all performed well, sounding suitably hi-fi and noise-free, especially considering the price of the unit. If four ins and four outs are all you need, they're flexibly implemented here. The surprise for me was the latency: the in/out delay was pretty good, even on my old Mac — it typically measured under 10 milliseconds — and of course I was able to monitor with zero latency while recording anyway. M Audio's drivers seemed quite efficient, too, and pops and crackles were not a regular occurrence during the review period.

On the subject of processor speeds, M Audio recommend a minimum of an

800MHz G3 or 733MHz G4 for Apple machines (and Pentium 3 at 800MHz or better for PCs). It was interesting, therefore, to install the drivers on my 450MHz G4; doing so produced a warning that my machine was running at less than 500MHz, and that the interface might not work properly. But it let me carry on, and in the end, it worked fine for modest sessions in *Cubase SX* and *Reason* 3, though for the latter I needed to tweak various system settings. But then *Reason*'s not always happy running on an older Mac anyway...

MIDI performance was also exemplary: that's a lot of controllers for one compact, multi-purpose device, and editing was easy, thanks to the Ozonic's nicely sized display — although who wouldn't use the *Enigma* editor? Aside from its slightly terse documentation, this is a great utility: I just wonder why it's not included on the bundled Ozonic CD-ROM? Finally, the keyboard itself is worth a mention — it's a typical plastic synth job, but offers a smooth sense of 'resistance' rather than feeling clattery and cheap.

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Conclusions

During this review, I became quite enamoured of integrated MIDI control and audio interfacing. The Ozonic offers enough physical controllers to make the combined controller/interface approach worthwhile, and audio is handled elegantly (especially the monitoring options). The build quality is quite sturdy, too, which is always a plus. I see the Ozonic as a definite step forward, and look forward both to what M Audio might do with Firewire in future, and to what their competitors might muster in retaliation! SOS

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SOUND ON SOUND
The World's Best Music Recording Magazine

In this article:

Mackie Tracktion 2

MIDI + Audio Sequencer [Windows/Mac OSX]

Published in SOS August 2005

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Reviews : Software

Mackie *Tracktion 2* £129

pros

- Streamlined user interface is comfortable to work with.
- MIDI editor is very well implemented.
- Looped audio recording works nicely.
- Good collection of plug-ins included.

cons

- One or two minor glitches to be ironed out.

summary

Tracktion 2 is a powerful and user-friendly application, boasting some original design ideas. Already impressive, it may yet develop into something quite unique.

information

£ £129 including VAT.

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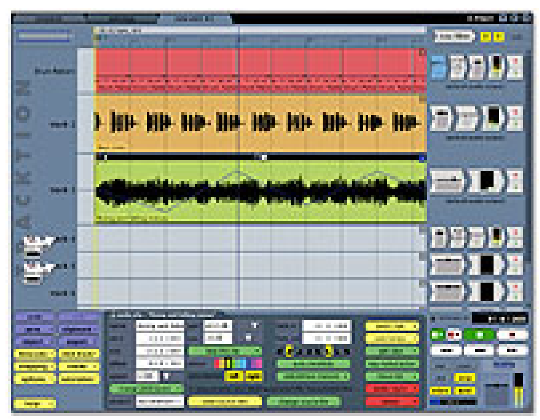
Test Spec

- 1.8GHz AMD Athlon PC with 512MB RAM and two 7200rpm Maxtor hard drives and Emagic Audiowerk2 soundcard, running Windows XP with Service Pack 2.

***Tracktion*'s designers have given the program a thorough overhaul and added some significant new features. But have they succeeded in retaining its freshness and ease of use?**

Paul Sellars



A year or two back I found myself exchanging tetchy emails with a software developer who'd been working for some time on a MIDI and audio sequencing application. He'd been annoyed by a previous reviewer's comment that his software had 'too many dialogue boxes', and was determined that the same thing shouldn't happen in any review I might write. I could understand his concern. He'd invested countless man-hours developing the software, adding features, flexibility and options, and all the reviewer can find to say was 'too many dialogue boxes'. It must have been more than a little frustrating.

However, I could also sympathise with the reviewer. Complex applications do sometimes seem to require the user to search for parameters among a wearying profusion of boxes and windows. It's all too easy to end up with a screen littered with the things, often partially obscuring one another, and generally placing unwelcome layers of abstraction and obstruction between you and what you're trying to do.

Mackie's *Tracktion* aims to overcome — or at least reduce — these problems with a user interface designed to be cleaner, more intuitive and freer of clutter. "It won't," Mackie tell us, "pretend to be a mixing desk or show you panels with pictures of screws that are accurately copied from a real piece of hardware." And why should it? *Tracktion* has been around for a couple of years now (see John Walden's original review in *SOS* April 2003: www.soundonsound.com/sos/apr03/articles/rawtracktion.asp), and was first released as a download-only package by independent developers Raw Material Software. Since then it has been adopted by Mackie, and version 2.0 sees it transformed into a fully fledged boxed product.

To recap, *Tracktion* is an audio and MIDI sequencer for Windows and Mac OS X, offering as many tracks as your CPU and hard disk will allow, supporting sample rates up to 192kHz, and boasting a variety of other features, including Rewire and VST plug-in support, a useful track 'freeze' function, a built-in virtual sampler, and more. A range of new features has been added for version 2.0, including an improved MIDI editor, new sync capabilities, Quicktime video playback, enhanced MIDI controller mapping, and support for the Mackie Control and C4 hardware control surfaces, among other things. All this, Mackie hope, has been achieved without compromising that intuitive user interface.

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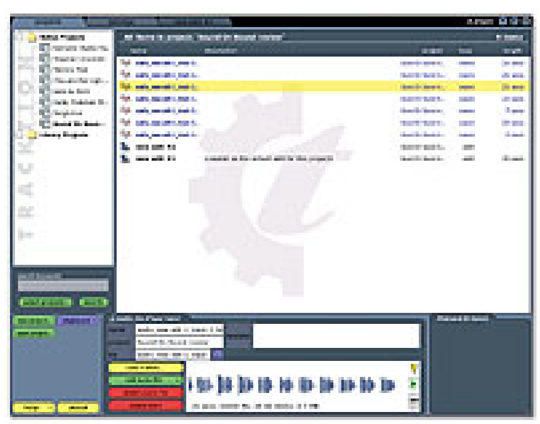
Getting Started

Installing *Tracktion* is reasonably straightforward. The installer program (under Windows) auto-runs when the installation CD is inserted, and the familiar installation wizard begins. A couple of mouse clicks later, and all the relevant files and documents have been copied to their proper places. Running the application for the first time, you're presented with a handsome blue splash screen, followed shortly by a dialogue box explaining that the software is running in 'demo' mode, and inviting you to 'unlock' it. Unlocking requires a serial number, which is supplied in the box, and ideally an Internet connection (although there's a workaround for machines without one).

You're asked to supply an email address and choose a password for your Mackie.com account, and to enter your serial number. Having done so, one more mouse click completes the process and you're ready to go. It would be nice not to have to go through this little procedure, but by the standards of software copy-protection systems, it's quite straightforward and trouble-free.



Tracktion's Projects page provides an overview of all the currently active projects on your machine.

Tracktion is quite well documented. A brief but helpful printed user's guide is included, which tells you everything you need to know to get started. This is duplicated as a PDF file for on-screen reading, while a more detailed reference manual is included in PDF format only. One ironic drawback of *Tracktion*'s somewhat unconventional user interface is that, if you've spent time becoming familiar with more the conventionally laid-out MIDI and audio sequencing packages on the market, you may initially have no idea what you're looking at!

What you're looking at, it transpires, is the Projects page, a kind of 'file manager' showing your currently 'open' projects and all their associated files. It's worth noting that when *Tracktion* talks about an 'open' project, it doesn't mean in the sense of a file that's open for editing. Rather it means a project that's been marked as a work-in-progress, and not filed away as finished. How 'intuitive' this is is arguable, but once you've got used to it, it can be quite useful.

Tracktion's different 'pages' are accessible from tabs arranged along the top of the main window. The Projects page has its own tab, while a Settings tab takes you to a page where you can make all the necessary settings to get things working, select your audio and MIDI devices, nominate a plug-in directory, edit shortcut key mappings, set up external hardware controllers and so on. Another tab allows you to access the Edit page. An Edit, in *Tracktion* terminology, is essentially an arrangement. You can have more than one Edit within a project, but you can only have one at a time open for editing. Again, we could argue about how appropriate a name Edit is in this context, but once you get used to the convention, it's no problem.

The Edit page is where you'll spend most of your time, and *Tracktion* is designed with the intention that you'll rarely need to look elsewhere in order to accomplish ordinary music-making tasks.

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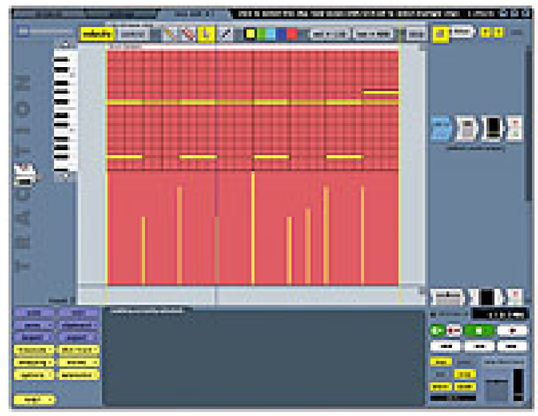
The Basics

While the promotional literature makes claims about a 'revolutionary' user interface, it would probably be fairer to describe *Tracktion*'s Edit page as 'evolutionary'. It doesn't, in my opinion, represent an enormous departure from conventional sequencer design, although it arguably does a better job than most of presenting the standard features we've come to expect from an application of this type.

Most of the Edit page is occupied by horizontal tracks running left to right across the screen, and calibrated with vertical bar lines. Pan, volume, mute and solo controls appear at the right-hand end of each track rather than in a separate mixer window. This arrangement will be more or less familiar to anybody who's worked with Sony's *Acid* loop sequencer, although in that case the controls reside at the left-hand end.

New 'filters' (*Tracktion*'s term for VST or other plug-ins) can be assigned to each track by simply dragging and dropping the 'new filter' icon to the end section of the appropriate track. When the icon is dropped, a menu appears from which you can choose from all the available filters. When you've chosen a filter, an icon will appear, and the icons of any other filters will shrink to make room for it. Quite complex chains of filters can be set up quickly, and rearranged simply by dragging and dropping as required. A number of useful filters are provided, including a four-band EQ, a reverb, delay, chorus, a compressor/limiter, a low/high-pass filter, and (new in *Tracktion* 2.0) aux send and receive busses. Rewire slaves are also accessed in this way, by dragging the 'Rewire Device' filter to a track. See the 'Bundled Plug-ins' box for more about some of the other filters included with *Tracktion*.



The new, improved MIDI editor still opens within the main Edit window.

It's worth pointing out that the volume and pan controls and level meter assigned by default to each track are themselves implemented as filters. They can be deleted if you don't want to use them, and you can assign more than one set of volume and pan controls to a track, if you feel the need to.

Select a filter by clicking its icon, and all its associated controls and parameters are displayed in the Properties panel beneath the main arrangement area so that adjustments can be made without having to open a new window. If the filter is a third-party plug-in with its own graphical user interface, this will 'float' in a new window above the Properties panel. This somewhat defeats the all-in-one-window design paradigm, but there's an ingenious workaround. You can optionally tell *Tracktion* to automatically hide VST plug-in interfaces from view whenever the mouse pointer isn't over them, having them reappear only when you move the mouse pointer back over the Properties panel. A nice touch.

The Properties panel doesn't only apply to filters: input devices (audio and MIDI interfaces) also have their own icons, and selecting these causes various associated parameters to be be displayed there. Input device icons live at the left-hand end of the tracks, and assigning an input to a track is as simple as dragging and dropping to make a connection. A connected input shows an arrow pointing to the relevant track.

Similarly, when an audio or MIDI 'clip' within the arrangement is selected, the Properties panel shows a number of controls and parameters, allowing you to adjust offset, start and end times, quantise and groove (for MIDI clips), pitch-shift and time-stretch (for audio clips), and more.

The remaining controls on the Edit page consist of a set of fairly conventional transport controls in the bottom right-hand corner, accompanied by a CPU load meter, a master output section with volume and pan controls, and a level meter. Filters can be added across the master buss in much the same way as with ordinary tracks. In the bottom left-hand corner, meanwhile, are a dozen or so buttons covering straightforward things like importing, exporting and saving files, clipboard functions (cut, copy, paste and so on), and various other functions.

Tracktion's Edit page certainly provides a comfortable environment in which to work. Audio and MIDI recordings can be made quickly and easily, mixing with filters is straightforward and intuitive, and regular glances at the Properties panel allow you to keep track of numerous different parameters without getting bogged down in menus and dialogues. Revolutionary or not, it's clean, logical, easy to learn, and a lot freer of clutter than certain other sequencers I could mention.

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Rendering & Dither

Just as I was beginning work on this review, I stumbled across a couple of threads on a couple of different web forums, describing what sounded like quite a serious bug in *Tracktion*'s audio file rendering code. Several *Tracktion* users had been reporting what seemed like essentially the same problem. Where files were exported containing sustained periods of low-level sound (long decaying reverb tails, for example) a rising hiss or similar noise could apparently be heard, followed by an abrupt truncation before the sound had fully decayed.

Initially I was sceptical, assuming that if such a bug did exist, the chances of it going unrecognised and unreported for so long were minuscule. Nevertheless, a couple of people seemed adamant that something was wrong, and had even posted sound files which they claimed demonstrated the problem. I decided I had better look into it. I tried creating my own test files, using a sampled snare drum and *Tracktion*'s default reverb filter. I exported a number of different files, using various different combinations of options in the 'Rendering' dialogue box, and prepared myself for the worst.

After 15 or 20 minutes playing the files back at increasingly high volumes, on speakers and headphones, I was none the wiser. Everything sounded OK to me. Certainly nothing leapt out at me as 'wrong'. Puzzled, I contacted Mackie, and asked if they could enlighten me. A couple of days later I heard back from *Tracktion*'s product manager who confirmed that, yes, a problem had already been found, and fixed, and a patch was undergoing testing as we spoke.

The problem, it turned out, was a fairly minor one. Essentially, *Tracktion* was applying its dithering algorithm whenever it rendered an audio file, regardless of whether or not the user had enabled the dithering option. Dither works by introducing a small amount of noise into a digital recording, in an attempt to improve subjective sound quality when reducing bit depth, such as when going from 32 to 16 bits. A few sensitive-eared users had picked up on the fact that dithering noise was present even when dithering wasn't supposed to be enabled.

Mackie have devised a fix for this, and in future *Tracktion* will only automatically employ dither when exporting 32-bit tracks. A patch should be available around the same time this review appears in print.

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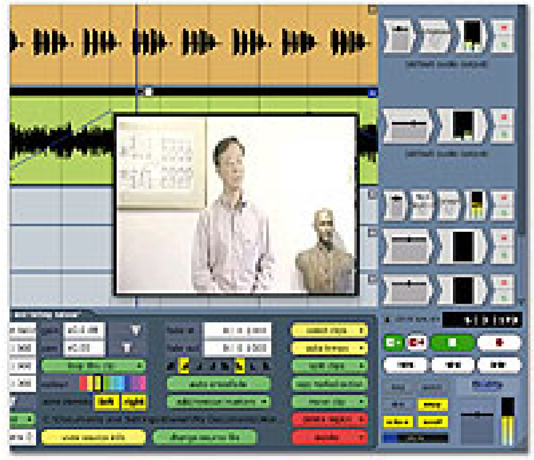
New In 2

One of the improvements Mackie are keen to advertise is an enhanced MIDI editor, and it does deserve a mention. In keeping with the all-in-one-window idea, *Tracktion* has no special, dedicated MIDI editor window to open or close. Rather, any MIDI clip that is displayed at a sufficiently large size will automatically appear with a 'piano roll' style editor superimposed on it, much as MIDI regions do in *Pro Tools*. Tracks can be resized by dragging vertically. Double-clicking a MIDI clip automatically increases its height by the amount required to view the piano roll.

When the mouse pointer is positioned over any MIDI clip large enough to display the piano roll, a set of tools and controls instantly appears, floating above and to the left of the clip. These remain visible until the mouse pointer is moved away. The controls include a miniature piano keyboard up the left-hand edge, which can be used to audition notes, and pencil, eraser, selection and line-drawing tools for all the usual editing tasks. There are also two buttons labelled 'velocity' and 'control' which allow you to choose whether to have MIDI velocity values or MIDI controller data displayed (and edited) beneath the notes on the piano roll.



Tracktion 2 can now import and play back Quicktime movies.

This system of floating and disappearing controls may sound peculiar, but it's actually quite intuitive and very easy to use. Although very detailed editing may require you to zoom quite a long way in on a clip, it's still nice not to have to obscure your arrangement with a whole new window (which would then need to be closed or hidden). It may be only a subtle refinement, one less window and a few less mouse clicks, but it makes a big difference to the feel of things.

Also new in *Tracktion* 2 is a step entry mode for MIDI data, which allows notes to be entered one at a time from a MIDI controller connected to the active track. A new note appears at the current play marker position, after which the insert point jumps ahead to the next snap position. Chords can be entered as well as single notes, and are recognised appropriately. Straightforward and, yes, intuitive.

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Films For Music

Another new feature is the ability to load and play Quicktime movies in synchronisation with your track. The implementation is basic, but nonetheless useful for anybody wanting to score to picture. However, while movie playback works, is straightforward and certainly usable, I found that there were one or two quirks associated with it.

First of all, movies are displayed in an entirely separate window. This is understandable, since there's no way anything but the smallest of video players could have been incorporated directly into the Edit window. Even so, having a separate movie window — which disappears from view the moment you click anywhere in the Edit window — is a bit of a pain, and perhaps not the kind of thing you'd expect from an application with a revolutionary user interface. The movie window does at least have a 'Keep window on top' option, which solves the problem of it disappearing, and ought to be enabled by default, but isn't. (This may seem like excessive nitpicking, and it perhaps says something about how smooth and well designed the rest of *Tracktion*'s user interface is that this one little quirk seemed quite so noticeable. In a different application, I might not even have mentioned something so small.)

Another, stranger quirk concerns showing and hiding the movie window. The 'proper' method for doing this is to click the 'options' button, then either check or uncheck the 'show Quicktime movie window' option in the pop-up menu that appears. This works perfectly well. However, if, like me, you make the mistake of closing the movie window either by right-clicking its title in the Windows taskbar and choosing Close, or with the standard Alt + F4 keyboard shortcut, you may find it then becomes impossible to get the movie window back. Even reloading the file afresh didn't provide a fail-safe cure for this. Mackie have confirmed that they are aware of this problem, and expect to have a patch available soon.

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Bundled Plug-ins

Tracktion's box proudly announces that 'more than $500 worth' of plug-ins are bundled with the software. These include IK Multimedia's *Sampletank LE* sample-based instrument and *Amplitube LE* amp simulator effect, and Linplug's *RMIV* drum machine, along with a good selection of freeware plug-ins.

The *RMIV* drum machine is an impressive VSTi, which marries flexible drum synthesis with sample playback, and makes a valuable addition to the bundle. Similarly, *Sampletank LE* and *Amplitube LE* are both well worth having, the former being a very usable sample-based virtual sound module, the latter a useful source of good guitar (and other) sounds.

Mackie also bundle their own 'Mixing Suite' of plug-ins, including a de-esser, mono and stereo side-chain compressors, and *Final Mix*, a powerful mastering plug-in combining six-band parametric EQ and three-band multi-band dynamics. Various other effects and instruments are included, many of which will be familiar to anyone who's ever gone browsing for freebies on the web. These may or may not be useful, but are at any rate a thoughtful addition, and will save new users the bother of having to go looking for extra gizmos to play with.

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In The Loop

Less problematic is the new loop record function for audio clips, which works faultlessly and is very nicely implemented. To begin looped recording, simply place *Tracktion*'s punch-in and punch-out markers around the section of the arrangement you want to loop, activate loop mode in the same way as for looped playback, and click Record. *Tracktion* records a new file for each looped pass, and when you've finished recording these are all 'layered' into one clip. The layered clip appears with a '+' symbol in its bottom right-hand corner, which can be clicked to reveal a drop-down menu where you can choose which of the recorded files you want the clip to play.

If you copy the layered clip to another track, the copy will contain all the same layers. So, for example, you might record four or five passes of a vocal, copy the resulting layered clip onto an adjacent track, then select your favourite take in the first clip and your second favourite take in the second clip. Assembling double-tracked choruses couldn't be easier!

There are a number of other new features in *Tracktion* 2, including the option to import files in both the 'Broadcast' WAV format and the proprietary format used by Mackie's HDR, MDR and SDR hard disk recorders. Integrated support for Mackie's C4 and Mackie Control hardware control surfaces is also included. Sadly I didn't have access to either a Mackie hard disk recorder or a control surface, and so I wasn't able to test these particular developments.

External sync is another new addition. *Tracktion* can now send MIDI clock, MIDI timecode and MIDI machine control (MMC) data, and can be slaved to MIDI timecode and MMC. My brief experiments with slaving a hardware drum machine via MIDI clock looked promising, before I got distracted by the *RMIV* virtual drum machine (see the 'Bundled Plug-ins' box).

Also new is an option to 'use 64 bit math' to optimise sound quality when mixing arrangements containing lots of tracks, although this comes with a health warning attached, as it makes fairly substantial demands of your computer's processor. A nice bonus for the really obsessive audiophiles though.

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Features Or Ergonomics?

If the primary goal of *Tracktion*'s developers is, as the promotional literature suggests, to develop an intuitive user interface with everything visible in a single screen, then they may be fast approaching a problem. *Tracktion* is already complex. Presently, it succeeds in allowing easy and comfortable access to a large number of parameters, mostly within one window. However it's hard to see how very many more features could be added without entailing some deviation from the everything-at-a-glance philosophy embodied by the current design.

The trend among most software developers seems to be towards adding ever more gadgetry to their applications; more options, more choices, more technical possibilities for the user to consider. Since many software companies depend upon paid upgrades for a proportion of their revenue, this is hardly surprising. However, if Mackie want to preserve the cleaner user interface that they've identified (correctly, I think) as *Tracktion*'s unique selling point, they'll need to proceed with caution from here. To add very many more options would be to run the risk of diminishing the very advantage that *Tracktion* currently has over many of its competitors. Extra parameters would have to go somewhere, almost inevitably spilling over into new windows and dialogue boxes, or cluttering the already rather full-looking edit page.

I personally hope that Mackie will resist the temptation to allow *Tracktion*'s future development to degenerate into a race to match their competitors gimmick for gimmick and gadget for gadget. *Tracktion* already boasts a feature set that could reasonably be called 'comprehensive', and isn't obviously lacking anything an 'ordinary user' (whatever one of those is) would often require. How much more interesting it would be to see what further simplifications and ergonomic refinements could be dreamt up for future revisions, instead of simply struggling to squeeze more options into the existing layout. Rather than adding new features, could still better ways of organising and presenting the existing features be devised? It would be no small challenge, but if successful, it could really put *Tracktion* in a class of its own!

Oh, and *Tracktion* doesn't have too many dialogue boxes. Yet. SOS

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SOUND ON SOUND
The World's Best Music Recording Magazine

In this article:

Mackie X200

pros

- Very flexible.
- Great user interface.
- Supports VST plug-ins.
- Emulates 24 channels of Mackie Control.

cons

- Noisy fans.
- No plug-in delay compensation as yet.

summary

Once the noise and delay-compensation problems are fixed, the X200 will be a thoroughly professional desk that is a joy to use.

information

£ See 'Options & Pricing' box for details.

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Test Spec

- Mackie X200 OS v1.030.
- Apple Mac G5 dual 2.5GHz with 4GB RAM, running Mac

Mackie X200

Digital Production Console

Published in SOS August 2005

Print article : Close window

Reviews : Mixer

Not content with acting as a high-spec control surface and Firewire audio interface for your studio computer, this new console seems determined to outshine it with its slick touchscreen graphical interface and onboard VST plug-in hosting.

Paul White

We've been keen to get our hands on Mackie's new Digital X Bus console ever since it was first previewed as the Mackie dxb. Apparently the name was changed after complaints from Dbx that dyslexic engineers might think they were buying a rack compressor! There are two versions of the console, the X200 and the X400, the main difference in the larger X400 being greater I/O capacity, better surround mixing support, and an included Universal Audio UAD1 card. The UAD1 card may be fitted as an option to the X200 and was included with the X200 I had for review.

Photos: Mark Ewing

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Digital X Bus

Measuring 43.6 inches wide, the console is quite deep at 31.8 inches, but you don't have to stretch far, because all of the controls are within easy reach, with the monitors and card slots taking up the space towards the rear of the enclosure. The console is pretty weighty at around 75lbs, but, amazingly, the more powerful X400 is exactly the same physical size. In terms of design philosophy and general layout, these X Bus mixers sit somewhere closer to Sony's recently discontinued DMX R100 console than to the earlier Mackie d8b, specifically because of the use of dual touchscreens as key parts of the user

OS 10.3.9.

- Apple *Logic Pro* v7.1.

interface.

Although physically compact (relatively speaking), the X200 is actually a pretty big console on the inside, with a maximum capacity of 68 inputs, 76 outputs, eight busses, and the ability to utilise spare I/O as channel inserts, aux sends/returns, and direct channel outputs. The routing system is very flexible, so you can use the available I/O in just about any way you need to. There's full mix automation with graphical automation editing, support for third-party plug-in effects, and the ability to emulate a 24-fader Mackie Control with all its optional DAW 'personalities'. Full MIDI Machine Control is implemented, so MMC-compatible recording systems can be armed from the X200 as well as having full remote transport control with jog wheel where supported. Unlike the d8b, the X200 has large, illuminated buttons and clear screen graphics, giving it a much more up-market feel, and because the screens take over the work of a large number of physical controls, the user interface doesn't feel at all cluttered.

One thing the X200 has in common with the d8b is that a PC motherboard is at its heart, in this instance utilising a Pentium 4 processor running at 3GHz and powering an embedded version of Microsoft's Windows XP. This lean, mean operating system accesses a 60GB hard drive to store the OS and project data, as well as the installed plug-ins, and there's 1GB of RAM as standard with a maximum capacity of 2GB. There is also a purpose-built DSP which runs the console's dynamics and EQ, but mixing is done on the host processor in 32-bit floating-point format.

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UAD1 Card & VST Plug-ins

Because of the PC architecture, Mackie have been able to give the Digital X Bus the ability to run third-party VST plug-ins (but not Direct X ones as yet). As mentioned earlier, the UAD1 card can also be fitted into a PCI slot to add more powerful plug-ins, but currently the TC Powercore PCI card can't be used, as it is physically too large. Having said that, a new smaller Powercore card has been announced, so this situation may change in the near future — apparently Mackie are still awaiting one of these cards for tests. Authorisation of third-party plug-ins is exactly the same as for any other computer-based system, and those plug-ins requiring an iLok are no problem, as you can simply leave the iLok plugged into one of the rear-panel USB ports or into a USB hub.

VST plug-ins also run on the host processor up to the maximum CPU capacity available, but at this stage there's no automatic plug-in delay compensation, so you have to handle that manually. Apparently Mackie are already working on implementing it, although it may take some time yet. As an intermediate step, they are hoping to introduce automatic delay reporting, so that you can more quickly dial in the relevant compensation for yourself.

Although there is no plug-in delay compensation, the onboard EQ and dynamics should cause no problems, because they generate less than one clock cycle of delay — none, in practice — and it also isn't a problem for reverbs fed from sends (as reverbs tend to be used with a little pre-delay anyway), but those plug-ins that do take more processing time could present a problem unless compensated for. Mackie have tried to make this as painless as possible by providing a compensation delay of up to 500ms that can be introduced into multiple channels simultaneously, though as plug-in delays are usually measured in samples, it might have been more practical to express the compensation delay in samples too.

All the option cards currently available for the X200 are shown in the top picture (left to right): Sync Card, Mix Out Card, Line Card, Digital Card, Mic/Line 8 Card, AES Card, Mic/Line 4 Card, and Firewire Card. Of these, the first two are included with the X200 as standard, and the eight remaining slots can be filled with any combination of the other six option cards. A few miscellaneous data connections, including USB and MIDI sockets, can be found at the other end of the rear panel (bottom picture).

Where a plug-in is being used in an effects loop, the appropriate send is routed to the plug-in and then the plug-in's output is routed back into a spare input channel. There are no dedicated input, monitor, or aux-return channels on this desk — just channels. This actually simplifies the operation considerably. Channels are routed in a very conventional way, where the user can choose from a list of all possible input sources, while effects routing is handled very simply in a dedicated Effects Rack screen.

In the current software revision, there is one pre-fade and one post-fade insert slot per channel for plug-ins (plus an assignable hardware insert point), though Mackie are looking into ways of creating 'macro' combination effects within the Effects Rack that can be dropped into a single slot. Analogue inserts are also available using the installed I/O to get signals into or out of the computer, and because the internal routing can access any I/O port, hardware effects and processors can be left permanently connected to save on patchbays and wiring. Up to 12 VCA-style fader groups can be set up and faders may be included in more than one group. Onboard tools include a spectrum analyser that can look at the channels, busses, or just about any other audio source you can think of, and there are three test oscillators available.

Although there's a 60GB internal hard drive, there's currently no means to connect the system directly to the Internet and no built-in CD-ROM drive. Software updates can be loaded via USB memory stick, USB drive, or external USB CD-ROM drive. Because the two touch-sensitive 15-inch colour screens take care of so much of the mixer's routine operation, the physical control surface is actually very streamlined and is centred around 25 motorised Penny & Giles faders (with a resolution of 1024 steps) plus 24 assignable rotary controls (called V-Pots) arranged beneath the screens.

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Interfacing Slots

As supplied, the X200 has just about enough I/O to make it useful as a headphone amplifier, and it includes no mic preamps as standard! Because there are so many different digital and analogue I/O configurations possible these days, Mackie have wisely stuck with a slot system where eight slots can handle up to a maximum of 64 channels of I/O — when you add these to the Mix Out card that comes with the mixer (dual stereo control-room outputs, digital I/O on both AES and S/PDIF, main mix output, and phones socket), the absolute maximum capacity of the mixer is 68 inputs and 76 outputs. This capacity is maintained at all sample rates from 44.1kHz to 96kHz with DSP EQ and dynamics available on 64 simultaneous channels, though the count is halved if you opt to work at 176.4kHz or 192kHz. [*Mackie informed us just before we went to press that a new software update has been released to remove this limitation. The higher samples rates apparently now only disable three of the card slots — Ed.*] The ADAT/TDIF cards include two ports each to facilitate working at higher sample rates (using the SMux protocol) without having to halve the channel count, but if you want to taste the heady air of 192kHz, something has to give — the ADAT I/O count drops to four channels per card.

At the normal 44.1kHz and 48kHz sample rates, only one of the two ADAT ports is used. A Firewire card is currently available that provides a total of 24 streams in and out at the lower sample rates. At 96kHz, this is reduced to 24 audio streams in total, for example 16 inputs and eight outputs. The Firewire card currently supports ASIO and Core Audio on the PC and Mac platforms so represents a practical way to get audio data moving between the console and a suitable computer DAW. However, in its current state, it doesn't have the capacity to enable the console to be used anywhere close to its limits. Note that the console functions as a Firewire device, not as a computer with a Firewire port, so you can't connect Firewire drives or TC Powercore boxes to this port. Maybe this is something Mackie will be able to remedy in future versions by adding a conventional Firewire port. There is currently no MADI card, but this option is under discussion.

Included with every basic system is a Sync card to handle word-clock I/O plus SMPTE in and out; next to it, there is a blanked out slot which is currently only relevant to X400 owners, as it enables high-density I/O to be added. The I/O card slots are divided into three main areas known as cages, where each cage can handle a maximum of 24 I/O channels.

With the console switched off, cards can quickly and easily be installed on site (above left), and you can also access extra motherboard connections by unscrewing a rear-panel plate (above centre) — the four extra USB sockets in particular could prove useful given that mouse and keyboard, MIDI interface, and plug-in dongles such as iLok (above right) may all require USB connectivity.

Interestingly, the mic preamps fitted to the Mic/Line option cards aren't based on any existing Mackie designs, but on the rather sophisticated TI/Burr-Brown PGA2500 chip, which has the advantage of being compact enough to build into multi-channel mic cards and also has the ability to have its gain digitally controlled. Many otherwise fine digital consoles fall down in not having the mic gain trim under digital control, making it impossible to recall these settings. Those who have fallen in love with the sound of the new Mackie Onyx preamps can plug in one or more Onyx 800Rs via ADAT/TDIF I/O cards.

Internally, the console has eight mix busses and direct channel outputs in addition to the main stereo buss, and supports up to eight aux sends (eight mono or four stereo), each configurable as pre- or post-fader. Four further sends provide a pair of stereo (or four mono) cue mixes in much the same way as they did for the d8b. On top of its functions as a fully automatable digital mixer, the X200 also has a MIDI control layer that emulates a Mackie Control plus two expanders, and which works with any of the DAWs supported by Mackie Control. I tested this with Apple *Logic Pro* — any buttons not available as hardware counterparts on the X200 come up on the touchscreen along with the text data that normally comes up in the displays of the Mackie Control. In this example, the X200 was recognised by *Logic* as a Logic Control and it behaved exactly like my dedicated Logic Control hardware.

All the connections to the console — other than the mains supply, the MIDI In and Out, the nine-pin RS422 connector, a pair of footswitch jacks (talkback can be controlled via one of these), and the usual computer ports (two USB ports and Ethernet RJ45) — are via the expander cards. Note that the nine-pin socket is as yet only functional on the X400 model. Four further USB ports are hidden behind an access panel along with connections for a standard keyboard and mouse. However, you don't need a keyboard, even for naming things, as a full-size QWERTY keyboard is available on the touch-sensitive display whenever it is needed.

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Booting Up

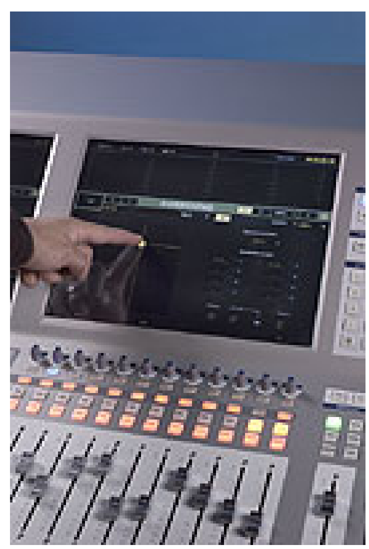
Once powered up by means of the rear-panel mains switch, the console is up and running in around 30 seconds, but because it has a computer at its heart it must be shut down computer-style at the end of each session. On this current version, it's also wise to pull down all the faders prior to powering the unit up, otherwise there is the possibility of calibration errors occurring.

Plug-in parameters are available directly from the Effects Rack screen, but many plug-ins also have a dedicated graphical user interface which can be accessed via the GUI button.

Although no noisier than a moderate PC, I found the fan noise of the X200 to be louder than expected — it was a little quieter than my Mac G5, but there you have the option of siting the computer under the table at the opposite end of the room. If Mackie want the X200 to be taken seriously as a professional console, this has to be improved. There's also a very noticeable thump through the audio outputs when the console is being booted up or shut down, which could have been handled more elegantly, possibly through the use of timed relays.

As regards synchronisation, the X200's sync card can read or generate word clock, or the console can run to its own internal sync. The S/PDIF and AES-EBU inputs can also be used with an integral sample rate converter but oddly there's no option to sync to external AES-EBU, S/PDIF or ADAT sources — something I expected as standard. In most cases it makes sense to run your console as sync master, as that's where the converters are, but situations can arise that require the console to slave to another source. There's also no way to connect the X200 directly to the Internet, which can make life more awkward than necessary for software updates or plug-in authorisation, though it's obvious the designers have made this choice due to worries about viral infections. Should the computer pick up a virus from another source, the whole OS can be trashed and replaced from the OS CD-ROM, but as there's no internal CD-ROM drive, you'll need a USB CD-ROM drive to be able to do this.

All 25 motorised faders have a beautifully smooth action and are electronically damped to reduce chattering during busy mixes. This means that, although automation changes always happen as quickly as they should, the faders may lag behind very slightly on playback, but this is still better than audible chattering. The V-Pot controls sit beneath the displays, but because their values are shown on the display, there's no ring of LEDs around each one as on Mackie Control.


Although you can control

The main section of the console comprises 24 identical channel strips with the V-Pot at the top and the 100mm motorised fader at the bottom. Select, Assign, Solo, and Mute buttons sit between these, and between Channels 12 and 13 is a row of buttons which determines the role of the Assign button (Record, Left-Right, Read, and Write). The latter two functions work in conjunction with the automation system for which there are further buttons beneath the centre section to select Trim, Fader, Mute, Pan, All, and Bypass modes.

surround panning from the assignable rotary controls, it's easier to just drag the surround panner around on the touchscreen.

The EQ and dynamics are accessed from the touchscreens and have very nice graphical user interfaces. Metering is also shown on screen, so there's no need for hardware metering at all. Three dedicated bank-select buttons bring up channels 1-24, 25-48, or 49-72, with further buttons accessing the masters, busses, and MIDI-control layer. In addition to the Mackie Control emulation facility, 12 MIDI channels can be set up, with the X200's physical controls sending any type of standard MIDI information.

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Master Section

To the right of the main channels section is a refreshingly uncluttered master section with control-room monitoring modes for stereo or surround up to 7.1 (assignable to any physical outputs), as well as switches for two sets of monitors. Dim and Mono buttons are available, as is the ability to specify two user monitor sources that can be selected using the '1' and '2' keypad buttons. There are two sets of headphones that can be sourced from just about any audio stream (two dedicated buttons are available for setting these up) or from the main left-right mix. Talkback is equally configurable, and the amount of dimming that happens when the talkback is activated (or when the Dim button is pressed) is user adjustable. A small talkback mic is built into the console, and when this is activated the adjacent monitor Dim light comes on as well as the Talk button's lamp. As usual, talkback uses a momentary-action button so you can't accidentally leave the talkback on.

The dedicated monitoring section includes control-room monitoring, two independent headphone feeds, and talkback facilities with a built-in mic.

The Solo buttons can be switched to operate in either AFL, PFL, or Mixdown mode, and there is a physical level knob as well as a Solo Clear button. All the switches on this console have integral lighting, and it's all very bright and clear with suitably large legending. Cue 1 (Aux 9+10) and Cue 2 (Aux 11 +12) have their own little section, while for the remaining aux sends there are eight numbered buttons as well as L-R Pan and F-B Pan buttons — the latter enables the V-Pots to be used for front-back panning in surround mode, though using a finger to guide the 'blob' on the touchscreen is just as easy. A Centre Percentage parameter allows the user to decide whether front-centre material in a surround mix should come from the centre or left-right speakers, and in what proportion. In surround mode, the main monitor's level control affects all the outputs being used to output the surround monitor mix.

Moving down, there's a row of Macro buttons, which are really programmable function keys. There are eight of these, plus a Shift key to double the number of options, and these are useful for directly opening different windows or pages — or just about anything else you do often. Standard Alt and Ctrl keys are also provided to add functionality and shortcuts to some of the parameter adjustments. I felt there should have been a dedicated Undo key though, even though you can assign one of the Macro keys to do this for you.

As only 24 channels can be seen on the displays and controlled by the faders at any one time, there are dedicated buttons for the three numbered banks of channels, as well as for the masters, groups, and MIDI control layer. However, one very neat function is that any channels you want to keep on screen can be locked so that they won't vanish when you change the page view. This could be the input channel you're currently using to do an overdub, or the master outs if you need to keep an eye on them, for example.

Other than the master fader, the lowermost part of the master section is really dedicated to hardware control, with full-sized transport buttons, Edit Start/End buttons, a Snapshot button, ten numeric keys, Loop buttons, and Locate, Store, Set, and Enter keys. There's also a large data wheel with a switchable Scrub function. The console is fully MMC compatible, and to test this I hooked it up to my Alesis ADAT HD24 where it operated the transport, selected tracks for recording, and so on with no problems.

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Mackie Control Emulation

The MIDI layer of the X200 can be switched to control the following software:

- Digidesign *Pro Tools* v4.1 or higher, although users of v6 and higher require a Legacy MIDI Controller Patch from the Digidesign web site.
- Apple *Logic Express*, *Logic Pro* and *Garage Band*.
- Steinberg *Nuendo* v1.52 or higher.
- Steinberg *Cubase SX*/*XL*.
- Cakewalk *Sonar* v2 or higher.
- MOTU *Digital Performer* v4.1 or higher.
- Sony *Vegas* v5 or higher.
- Adobe *Audition* v2 or higher.
- Magix *Samplitude*/*Sequoia*
- RML Labs *SAW Studio*

■ Mackie *Traktion*

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Getting Around The Screen

Most areas of the multi-channel mixer's condensed channel-strip display (left) can be touched to access the more detailed settings displays shown to the right. Touching the metering display switches between per-channel and global metering displays.

One of the first things you see when you boot up the console is an upside-down logo on each of the screens. The reason for this is that Mackie have had to invert the screens to give the best viewing angle given that the screens are placed on a slant, and they've inverted their video drive signal to get the display the right way up again. What you see during bootup are the screen manufacturer's icons, which are then replaced by the Mackie screen.

The default screen for the console comprises 24 channel strips with metering (user selectable pre- or post-fader) at the top. Touching the meter strip toggles between showing the relevant 24 meters on the screen and all the meters for the console at once. You'll notice that the channel strip is divided into familiar sections, and touching each section brings up a window allowing you to access that section in more detail. For example, below the meters are the familiar routing buttons, and touching these brings up a larger, more detailed channel-routing page. This also also handles the channel insert points and channel direct outputs, as well as providing mic preamp gain control and phase and phantom-power switching.

As is now fairly standard, channels can be linked by holding down both Select keys, whereupon a screen lets you tick off which parameters to link and which to leave separate. A separate routing page is available from the Windows menu at the top of the screen (or from a Macro key) to allow you to set up sources for all the channels. This is, in effect, a digital patchbay between the physical I/O and the channels. Directly below is a panel for selecting all the aux sends, including the Cue sends, where each send level is shown via a horizontal bar in the channel display.

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Channel Dynamics

Then comes the channel dynamics section, denoted by a representation of a VU meter, which is normally grey and doesn't move unless the compressor is on. Touching this opens up the dynamics window, which comprises a straightforward variable-ratio compressor and a gate with a side-chain filter. The switchable hard/soft-knee compressor has Threshold, Attack, Release, Ratio, and Output controls as well as a graphical display of the compression curve, the threshold, and the amount of gain reduction being applied. Additional meters show the input and output level to the dynamics section.

Although the gate is at the right-hand side of the page, it comes before the compressor in the signal path, which is as it should be. Side-chain filtering is provided as a single filter with Q and Frequency controls, rather than as separate high-cut and low-cut filters, and there's also a choice of Key Trigger Input via a drop-down menu that allows access to all the expected auxes and busses, plus the reverb, oscillators, and talkback mic.

When the compressor is active, the VU meter icon turns white and the meter moves to show signal level (selectable from input, output, or gain reduction). Just a thought, but some indication of gate activity would be useful when the dynamics window is closed — a virtual LED in the VU meter for example. Also, the VU meters could usefully reflect the channel level when the compressor is bypassed — when the main metering is shown in global (all channels) mode, it's not always easy to see which meter relates to which channel.

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Four-band Equaliser

Currently the four-band equaliser has a choice of curves for each band (parametric peak, high/low shelf, or high/low-pass), but a number of users have asked Mackie to add additional high-cut and low-cut filters to save having to use their main EQ bands — apparently this is currently under consideration by Mackie for a future software revision.

When the EQ page is open, the EQ points can be dragged on a graphical representation of the EQ curve, or adjusted directly using the V-Pots beneath the screen. When switching from one EQ setting to another (as opposed to using dynamic automation, which is also possible), a user-defined morph time may be set up.

The EQ is post-dynamics, which is normally where I would choose to use it, but as the effect of using EQ pre- and post-compression can be very different, I feel that the order of the EQ and compressor should have been switchable, though still leaving the gate as the first processor in the chain. Again, as this is a software-based mixer and there's no real limit to what can be added on screen, this type of improvement should be possible if sufficient people ask for it. Both the EQ and dynamics sections have automation Read and Write buttons, the status of which is shown below the meter in the channel strip.

The X200 includes a real-time spectrum analyser and three test oscillators.

At the bottom of the strip, just above the positional display for the physical V-Pot below, is what looks like a cross-hairs gun sight. This is in fact the pan control display and it's designed this way so that when the console is being used in surround mode, it shows the left-right and front-back pan position. Touching it brings up a large version that can be controlled directly from the touchscreen or the V-Pots, and it also allows the surround busses to be assigned and the surround mode to be selected. This page includes LFE level and filter frequency as well as a Centre Percentage control to determine how much of a front-centre-panned sound comes from the centre speaker and how much from the front-left and front-right speakers. Again there's a morph function so that one pan snapshot can merge smoothly into the next rather than the positions jumping abruptly.

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Automation

The automation can either be done as a series of snapshots or it can be fully dynamic. Dynamic automation starts off from an initial snapshot or template that defines the state of the console at the beginning of the mix, after which automation moves can be written. The automation works fairly conventionally, with switchable Read and Write modes for selected channels and the ability to select Latching or Write Flyback mode. In Latching mode, the automation value stays where you last left it when you release the fader, whereas in Write Flyback mode it reverts to any previously written values. There's also a Trim mode where existing automation data can be nudged up or down using the fader. A control in the automation window allows the fader return time in Write Flyback mode to be set by the user, and it's here that you select what's to be automated — Fader, Pans, Mute, or Other. The Other setting comprises Aux, EQ, Dynamics, Busses, Phase, and Misc, each of which can be activated separately. Misc needs to be active to allow effects to be automated.

A familiar 'breakpoint' graphic display is generated as automation data is written, and this may be copied, pasted, deleted, or edited, all on screen. This includes automatic data thinning so you don't end up with more automation data than you really need when editing or drawing in new moves. When cursor scrolling is active, the display always aligns itself around the current time location so that you can see the automation relevant to the audio you are hearing.

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User Impressions

It would be wrong to say that you don't need to open the manual at all, but I'd say that you could figure out 95 percent of this console just by doing what appears to be obvious and seeing what happens! There are some hidden features such as double-clicks and the use of modifier buttons, but these are often used to speed up something that can be done another way. For example, holding down Alt allows you to adjust in smaller increments, while holding down Control and moving a knob resets the parameter. In fact it's only the automation that needs much in the way of guidance, and even then one quick skim through the manual will probably suffice. That's not to say Mackie haven't missed a trick, though, as having 'mouse over' information available on tools and windows could help out the newcomer, as could context-sensitive help pages — just as long as you can switch them off when a client comes in so you don't look as though you don't know what you're doing! And while I think of it, I often found it difficult to see the decimal point in the parameter value windows, such as when setting compressor release times or EQ frequencies.

As pointed out in the main body of the review, there are little ideas and improvements that could be incorporated in future software upgrades, and even as I write this review, I know there's a version 2 of the software only two or three months away. Hopefully that will address issues such as inserting multiple effects and swapping the compressor and EQ as a user option. That's the advantage of a computer-based console — it can improve with age. In fact the only flaws with this console that I consider to be 'serious' are the lack of automated plug-in delay compensation and the excessive fan noise. Even sub-£200 sequencers now seem to have delay compensation as standard, and in a professional environment you can't go messing around with manual delay compensation every time you add a plug-in.

One other point, which is a suggestion rather than a criticism, is that I feel there should be some way to feed the video signal from your sequencer or hard disk recording hardware into the back of the console, then switch between it and the mixer's own displays using physical buttons, such as the assignable function buttons. This would free up a lot of studio real estate, and allow sequencer users to edit their audio and MIDI tracks from behind the console, giving a better feeling of working in an integrated environment.

These observations aside, though, the X200 is friendly and very flexible — in fact it's so easy to use that I kept thinking to myself 'There must be more to it than this!' The way the touchscreen has been organised makes it very easy to navigate around the console and in some cases it's faster than getting round an analogue console with the same number of channels. There's a simple but practical file manager so that you can keep track of your sessions, save your own EQ/effect presets and channel settings, access the undo history list, and import session data from the Mackie d8b. The EQs and dynamics are different, so some sonic changes are to be expected, but these should be for the better!

A set of six buttons between the two main banks of faders (above) select areas of the console's operation for automation purposes, making mixing a very hands-on process. However, if you wish to edit mix automation in detail, the on-screen Mix Editor window offers much the same functionality as you'd expect from a software sequencer.

I was most impressed by the Mackie Control emulation, but then, as the same company wrote the Mackie Control software in the first place, that shouldn't be in any way surprising! Having to buy an external USB, multi-port MIDI interface to use all 24 channels of control is less friendly, though it does seem that direct USB communication is planned for a future software release. By way of facilities, the console comes with a simple but decent-sounding reverb designed by Sanewave, a bunch of very clever ex-Mackie engineers headed up by Bob Tudor, who now do a lot of Mackie's digital design work on a contract basis. The same company also provide a simple stereo delay effect, but other than that you only get *Final Mix* as standard — a mastering tool comprising three-band dynamics, six-band EQ, soft clipping, and gating, but oddly no limiter. Of course all the channel dynamics and EQ are standard, as they're not plug-ins, and you can access these on the busses and master outs (up to a maximum of 64 at up to 96kHz sample rates) as well as on the channels.

Now we come to the sound of the console, which is one of the most difficult things to evaluate. The converters, though not top-of-the-tree esoteric models, sound pretty sweet to me and the new EQ has a nicely analogue feel to it, even though you still need to pile it on rather more thickly than an analogue EQ to get the same subjective result. While the dynamics section doesn't offer anything particularly fancy, it sounds firm and positive, but I'd strongly recommend adding a UAD1 card, as their vintage EQ and compressor emulations are excellent, as is their optional plate reverb. I've no complaints about the mixing engine itself, although as it currently uses 32-point float maths it is probably little different to mixing within a serious computer-based sequencer fitted with good-quality converters. What does differ from a software sequencer is that all the channel EQ and dynamics are supported by separate DSP, leaving you more power available for running plug-ins.

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Options & Pricing

- X200 console, £10999.
- Mic/Line 4 Card, £745.

Incorporates four XLR mic inputs with phantom power and four balanced line-level inputs.

- Mic/Line 8 Card, £995.

Incorporates eight balanced mic/line inputs, with mic phantom power, on two 25-pin D-Sub connectors.

- Line Card, £595.

Incorporates eight balanced line-level inputs and outputs on two 25-pin D-Sub connectors. Nominal operating levels can be switched between +4dbu or -10dBV via jumpers on the circuit board.

- Digital Card, £395.

Incorporates eight channels of ADAT and TDIF I/O with dual ADAT ports for SMux operation at higher sample rates.

- AES-EBU Card, £595.

Incorporates eight channels of AES-EBU digital I/O on one 25-pin D-Sub connector. Supports single-wire operation for eight channels at 96kHz or four channels at 192kHz.

- Firewire Card, £645.

Incorporates Firewire input and output ports for direct audio streaming to and from PCs or Macs. Compatible with ASIO 2 for Windows XP, or Core Audio for Mac OS X.

- Universal Audio UAD1 Project Pak, £329.

Increases the plug-in effects horsepower of the Digital X Bus consoles. Bundled with the following plug-ins: *CS1 Channel Strip*, *DM1 Delay Modulator*, *RS1 Reflection Engine*, *EX1 EQ & Compressor*, *Nigel*, *Realverb Pro*, *Pultec EQP1A*, *1176SE Limiting Amplifier*, *Preflex Amp/Cab Simulator*, *Gate/Comp*, *Phasor*, *Mod Filter*, *Trem/Fade*, *Mod Delay*, and *Echo*. For more information on the UAD1 card, see the review in *SOS* October 2001.

- Universal Audio UAD1 Studio Pak, £749.

Matches the Project Pak bundle, and adds *LA2A*, *1176LN*, *Cambridge EQ*, *Dreamverb*, and *Fairchild 670* plug-ins.

- Universal Audio UAD1 Ultra Pak, £949.

Matches the Studio Pak bundle, and adds *Plate 140*, *Precision EQ*, *Precision Limiter*, and *Pultec Pro* plug-ins.

Prices include VAT.

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Conclusions

For those people who still need a mixing console as opposed to a control surface, the X200 has much to recommend it, both in terms of its flexibility and its operational simplicity. However, there are a couple of issues that need to be sorted out before I'd be prepared to stamp to word 'professional' on the box. Specifically, the computer fan noise needs to be addressed as a matter of urgency, because having something almost as noisy as an average PC sitting under you nose when you're trying to mix just isn't on. There are apparently ducted silencing kits available as retrofit options, but I can't comment on these until I get to hear a console with one fitted. The other big issue in my view is that of plug-in delay compensation, both in the channels and busses. You simply can't afford to switch off your creative brain during a session to start doing manual delay-compensation calculations every time you insert a plug-in, so this needs fixing as a priority. Other than that, the only shortcoming that rates higher than 'minor' in my book is the paucity of external sync options.

In most other areas, the console is a joy to use — it is quick to navigate and it has plenty of I/O for all but the most ambitious projects, and if you really need more, there's always the 24-bus X400. Having the ability to run third-party plug-ins removes one of the main objections to a hardware desk, though if you're using the mixer in conjunction with a software DAW, you also have the option of doing some or all of the automation and plug-in processing within the DAW itself. In this respect, the Mackie Control layer gives you the best of both worlds, as you can still do just about everything you want to from the mixer's control surface.

A lot of thought has gone into the design of this console to make it friendly without sacrificing features, and on the whole the designers have done a great job. It brings the world of automation and plug-ins to those working with hardware recorders, and also provides DAW users with a mixer that integrates smoothly into their natural way of working. SOS

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SOUND ON SOUND
The World's Best Music Recording Magazine

Miglia Harmony Audio

Firewire Audio Interface [Mac OS X/Windows]

Published in SOS August 2005

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Reviews : Computer Recording System

A two-in, eight-out interface is the first audio product from British Firewire specialists Miglia Technology.

Paul Wiffen

Just at a time when British brands like Evolution and Novation are being absorbed into other companies, it was a pleasure to come across a new and British name in Firewire audio interfacing at this year's Mac Expo in San Fransisco. Tring-based Miglia Technology have been making Firewire-based products for some time now, but the Harmony Audio is their first foray into the world of audio. The Harmony Audio's key component is also British: it is the first product to feature Oxford Semiconductors' new 970 Firewire controller chip. The unit provides two analogue inputs on quarter-inch jacks and eight analogue outputs, all on mini-jacks. There is no digital I/O.

Miglia Harmony Audio £210

pros
- Small, lightweight design.
- No Mac OS X drivers required.
- No power supply required with six-pin Firewire.
- Soft Clip feature for old-style tape compression effects.

cons
- No phantom power.
- Mini-jack connections won't suit everyone.
- No digital I/O.

summary

A neat British-designed device for stereo recording and multi-channel playback at the right price, with a unique Soft Clip feature providing the icing on the cake.

information

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W www.miglia.com

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Miglia Technology: The Story So Far

Miglia Technology already have several successful products in areas that some *SOS* readers may not have come across. Their Director's Cut bridged the gap between the simplicity of digital video editing via Firewire connections and the significantly more fiddly business of getting analogue video in and out of your computer. By connecting this Firewire box to your Mac or PC, you could fool video-editing applications — from *iMovie* through Adobe *Premiere* all the way up to *Final Cut Pro* — into thinking that their source material was DV when in fact it was dear old VHS (or anything else that has standard analogue video in and out). As far the computer was concerned, it was talking to a DV camcorder (except of course that the transport couldn't be controlled), so you didn't need any additional Codecs in software.

With more flexible setups in mind, Miglia's second take on the concept, Director's

Cut: Take Two offered individual video and audio connectors on phono and for S-Video, the higher-quality four-pin connectors which keep the luminants and chrominants separate. There are actually two sets of video outputs so that you can connect to a preview monitor to see in advance what your picture is going to look like, or even write to two VCRs at once if you need to reduce the time to make multiple copies.

A third product, the Dual Disk, is a clever and affordable way of achieving hard disk data transfer rates that can exploit the full bandwidth of the Firewire 800 standard, using two hard drives striped together. Transfer rates are fast enough to support the DVC Pro HD Codec in *Final Cut Pro*, as well as the most bandwidth-hungry multitrack audio setup.

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Plug And Play

I should no longer be surprised when Firewire devices come up first time as an option in the Mac's Sound Control Panel as soon as they are connected, but there is still that moment of wonder when the Device name appears as if by magic in the Input and Output lists. Opening up the Audio MIDI Setup application showed all two ins and eight outs operating at 24-bit resolution, with four sample rates available: 32, 44.1, 48 and 96 kHz.

At first, the sample rate seems like the only user-adjustable option for the Harmony Audio, but you can set the master output level (which is very useful in multi-channel operation if you're using powered speakers and don't have a single hardware monitor level control), and there is also an innocent-looking button marked Configure Speakers. Apart form the basic Stereo Configuration, where you only get to choose which of the eight outputs your left and right signals are connected to, all the interesting stuff is under Multichannel. When you click on this, the window grows in size to accommodate a diagrammatic layout of your speakers. Again, a pull-down menu beneath each one allows you to decide which of the Miglia's eight outputs is being fed to that speaker.

Six configurations are available. Three are listed under the Surround heading — 5.1, 6.1 (which adds a centre rear speaker) and 7.1 (which offers an additional pair of rear speakers placed even further back) — and three under Geometric. The Geometric options should appeal particular to those doing sound installations for artistic exhibitions and other experimental work, with four, six and eight speakers placed at equal distances around the listener.

The Harmony Audio's analogue outputs are all on mini-jacks, which saves space but might make interfacing awkward in some circumstances.

When you're sat at the computer keyboard, this is the total extent of the user's control over the Harmony Audio unit. However, the front panel of the device also features some hardware controls. Each of the two input jacks (a stereo mini-jack input is also available) has its own rotary knob for Gain, with a Hi/Lo switch, corresponding to Mic and Line ranges, and a second switch mysteriously labelled Soft Clip. Whilst there is little to say about the Gain controls other than they work and the ranges are fine for a whole variety of input devices (and that I always prefer to set gain with a rotary knob as it makes fine adjustments easier), the Soft Clip bears closer examination.

An LED shows when the Soft Clip function is being activated, allowing you to vary the amount of saturation you want to introduce into your signal as you decide whether to briefly light the LED on the loudest transients or push the entire signal into the red. It is not clear from the documentation whether Miglia are introducing this soft clipping via analogue circuitry or numerical manipulation of the digitised signal (I suspect the former) but it doesn't really matter as however they are achieving it, it sounds great. There are very few sounds which do not seem sweeter to the human ear with a hint of Soft Clip applied, and some which really come alive when you whack loads of it on — cheap and nasty synths, for example. The only problem you might have is that is a bit like a drug: once you start using it, you may have trouble listening to anything which doesn't have it. Obviously, with individual signal sources this is fine, but if you are digitising complete mixed tracks, you may want to exercise the self-discipline of leaving the Soft Clip switched out.

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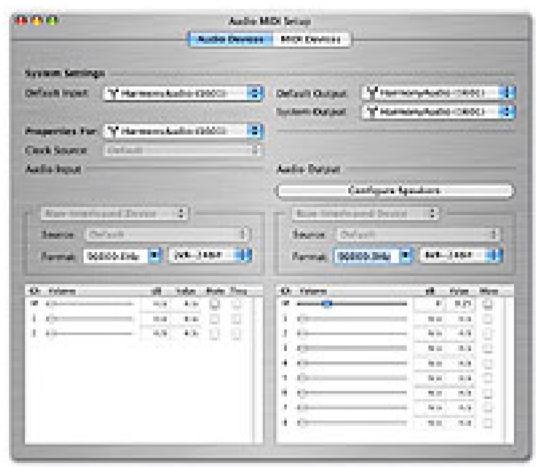
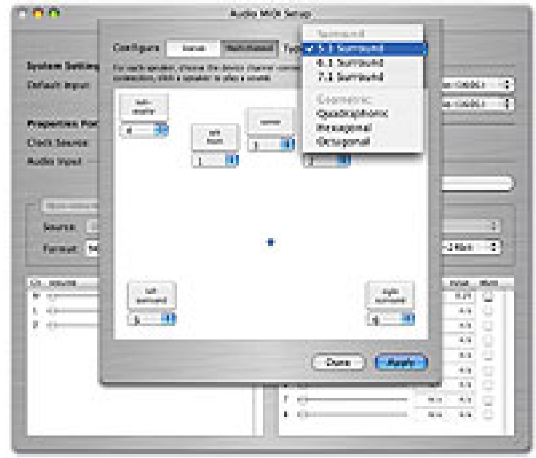
The Mighty Mini



No additional drivers are required to use the Harmony Audio within Mac OS X.

The only other control on the front panel of the Miglia is a headphone level control next to the mini-jack connector for plugging phones in. This theme is continured on the back panel with five more mini-jacks, one for an alternative Line In (which you can leave plugged up, as the front-panel connectors take precedence) and four to provide the eight line outs in pairs. To some, this might seem excessive use of the humble mini-jack, and no doubt the professional engineers out there will be whingeing about the lack of battle-proof XLR connectors complete with locking quick-release clips. However, at this sort of price, the mini-jack is king, and I for one find them more convenient and reliable than phono connectors or full-size jacks. Obviously, I would be foolish to claim that they are better than XLR connectors, but these are bulky and



If you're working in surround, you can set the Harmony Audio's output channels up to your requirements.

weighty and need much heavier cable to provide their full benefit. The Harmony Audio unit would not be the trim size and shape it is (easily fitting into the side pocket of a laptop carry case) if it had to carry 10 female XLR connectors on the back panel.

However, the mini-jacks could make things a little tricky when using outputs 3/4, 5/6 and 7/8 to drive speakers in a multi-channel surround setup. As your subwoofer may not necessarily be near your centre speaker, and your rear stereo pair in a 6.1 setup may be on 6 and 7, you might need to resort to making custom cables to get these outputs to your speaker configuration. However, you can easily change speaker assignments in the Audio MIDI Setup page to make life easier, and any self-respecting *SOS* reader should be able to solder his own speaker connections to a mini-jack.

There are two trim pots on the back panel for the line input so that you can make fine adjustments to the levels coming in from your various sources. The four stereo outs have no such control, but this is much better achieved in software anyway. Of course the lack of XLRs means that there is no +48V phantom power supply, which may rule the Harmony Audio out for some potential buyers.

Next to the outputs are the two six-pin Firewire connectors (having two means you can use it anywhere in a chain of Firewire devices) and a socket for a power supply, although you'll only need this if your computer is not capable of supplying power over the Firewire cable. All Macs do so, but many PCs, especially laptops, do not.

Miglia gave me early copies of the PC drivers required to make Harmony Audio compatible with the Windows platform, and these worked first time in making the unit work with a Sony Viao (once we had found a suitable power supply). The first beta they gave me did not address the delay caused by the latency buffers in Windows but later versions did allow applications like *Cubase* to set up their smaller buffers in ASIO drivers to compensate for this (Mac OS X users need not worry about this, as Core Audio provides any audio application with a way to reduce buffers to a negligible latency of under 5ms).

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Simply Does It

The Harmony Audio is one of the simplest Firewire devices I have ever used in terms of operational setup. It always operates at 24-bit resolution, leaving it to the software to discard the last eight bits if you're recording at 16-bit. The only required user setups are to set the sample rate from a choice of four, adjust the incoming levels in hardware with or without Soft Clip and set the outgoing overall level in software. The rest of the time, it is a plug-and-play dream, totally transparent in operation and extremely light in transportation (especially if you don't have to carry a weighty wall-wart around to make it work). It makes an ideal addition to any Mac portable and PC owners with Firewire can take advantage of its capabilities for the price of an external power supply.

The addition of Soft Clip to its capabilities makes it stand out from run-of-the-mill add-on audio devices, while the ability to set up the unit for various different multi-channel configurations makes it the perfect device for interfacing a computer with a surround sound setup. The lack of XLRs and phantom power may alienate high-end users but the average home musician will find this box has the right combination of connectivity and control. SOS

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SOUND ON SOUND
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In this article:

MXL V6 • 990 • 992

Condenser Microphones

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Reviews : Microphone

Three new large-diaphragm mics from MXL include a model which attempts to recreate the sound of valves using solid-state technology.

Paul White

MXL mics are built in China, but their body designs and finishes are quite distinctive and their circuitry is optimised especially for each model. Under review this month are three of their new large-diaphragm mics: the new V6, with its intriguing Silicon Valve technology, the 990, and the 992.

MXL V6, 990, & 992

pros

- Nicely styled.
- Solid, musical sound.
- Sensibly priced.

cons

- The V6 may be slightly sibilant with some vocalists.
- With the exception of the MXL 992, the noise figures are only average.

summary

The MXL V6 gets very close to the sound of the tube V69 at a lower cost, and offers the same classic good looks. Best suited to singers who need more 'air' and high-end projection. The MXL 990 and 992 are more conventional, but still offer a good combination of sound quality and value.

information

£ V6, £223.83; 990, £79; 992, £149. Prices include VAT.

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Silicone Valve Technology

Photos: Mike Cameron

Their V6 is slightly unusual insomuch as the California-based design team have deliberately sought to create a classic valve-mic sound using solid-state circuitry — it's no surprise that a FET is at the heart of their circuit, as FETs have similar characteristics to tubes. The FET circuit (the 'Silicone Valve') differs slightly from the FET preamps used in countless other capacitor mics, because it is biased to produce a similar level of second-harmonic distortion as is characteristic of a tube circuit. The output stage is transformerless, which surprised me, as the characteristics of transformers are often an integral part of what users think of as the tube sound. Full 48V phantom powering is required to run the mic, and the conventionally wired XLR output has gold-plated pins to ensure a good connection.

The capsule itself is fairly conventional. It's a one-inch cardioid element with a six-micron gold-coated mylar diaphragm and, like many classic mics, it is centre terminated. The mic's frequency range is quoted as being 30Hz-20kHz, but this is a pretty meaningless figure given that there are no limits specified and no frequency plot to refer to. The sensitivity is rather better specified at 22mV/Pa and the maximum SPL before distortion exceeds one percent is 130dB. This is high enough for all normal studio applications, including vocals, and is again typical for the type of microphone. There are no pads or roll-off switches on the mic, so in high-SPL situations, it may be necessary to use the pad on the mic preamp or mixer to which the mic is connected. At 1kHz, the dynamic range is a healthy 116dB. Noise-wise, the quoted figure is 16dBA, which is a respectable figure, though not as quiet as that of some of today's mics. A standard phantom-power supply (48V ±4V) is required for operation as it is for all three mics in this review.

Cosmetically, the mic follows the same classic cylindrical lines as the MXL V69, weighs in at 0.52kg, and measures 55 x 55 x 215mm. It is heavy enough to feel substantial, but not so heavy that it gives a mic stand a case of the droops. The mesh basket is finer than usual, and support metalwork is plated with 24-caret gold, which is certainly distinctive and also practical, as gold is very resistant to tarnishing. A green/grey metallic paint is used on the body, and the live side of the mic is denoted by the embossed MXL logo and the cardioid icon at the bottom of the basket.

Internally, the microphone follows the type of construction favoured by most Chinese manufacturers, where two metal rails run the length of the microphone, supporting both the circuit boards and the connector. The tubular body sleeve is held in place by a machined locking ring around the XLR outlet, and peeking inside reveals two glass-fibre circuit boards, one using surface-mount components and the other containing the discrete components that comprise the FET 'Silicon Valve' part of the circuitry. The standard of construction raises no concerns as regards reliability, but perhaps a little more could have been done to damp ringing in the metalwork, specifically the grille and the body sleeve.

The microphone is supplied in a neat, velvet-lined cherry-wood box with a separately packed shockmount. For recording studio vocals, a pop shield is also mandatory, but, as almost nobody seems to include these, one needs to be purchased separately.

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Auditioning The V6

MXL kindly supplied one of their V69 tube microphones for comparison, and as this is certainly the tube mic they tried to get close to with the V6, it provides an ideal reference. After working with the two mics for a while, I came to the conclusion that neither model sounded obviously tube-like, which is generally a good thing, as a tube mic with too much character has probably been engineered to introduce excessive levels of distortion. The V69 has a fairly neutral and open sound unless used close up, in which case the proximity effect can be used to introduce an enhanced impression of weight and depth. With some voices the mic can sound a little sibilant, but rotating the mic so you're working at around 45 degrees off axis fixes that nicely without otherwise dulling the tone unacceptably.

The V6 has perhaps slightly less lower mid-range density than its true tube counterpart, but it actually comes very close the sound of the V69. Because of the observations regarding sibilance, and by observing the way the mic behaved with typical vocalists, it's clear that there is a deliberate presence peak and, in my view, the mic is best suited to those seeking a more airy or intimate sound. It is perhaps less well suited to those with thinner-sounding voices unless it is used slightly off axis and close up to capitalise on the proximity bass boost. Because of its open, articulate sound, the V6 also works well on acoustic guitar and percussion instruments, making it a good all-rounder for the small studio.

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MXL 990

By contrast with the V6, the MXL 990 is a more conventional mic, albeit with a short body, giving it a somewhat stubby profile. Most of the mic comprises the basket, with the electronics and connector occupying around one third of the length of the mic. Overall the standard of construction is what you'd expect from a competent mic of Chinese provenance, and the metalwork is nicely finished and satin plated in what the manufacturers describe as a Champagne finish. A shockmount is included, as well as a regular stand adaptor, and the whole kit comes in a practical foam-lined plastic case. The MXL 990 is designed primarily as a cardioid-pattern vocal mic, based around a gold-sputtered six-micron diaphragm (20mm) in a capsule that has a nominally flat response from 30Hz to 20kHz. There's a gentle presence peak centred around 8kHz, but the effect of this is fairly subtle and it helps balance the proximity bass boost when the mic is used close up.

The circuitry uses all discrete components (including a FET preamp) mounted on a single glass-fibre circuit board, and the output stage is transformerless. There are no external pad or roll-off switches (the MXL 992 model has both a low-cut switch and a 10dB pad). This is a strictly no-frills cardioid mic, but the technical spec is generally respectable with a 20dBA equivalent noise level. This isn't exactly low by today's standards, but is more than adequate for close-miking vocals. While the maximum 130dBSPL isn't as high as some, it's more than enough for routine studio work, though I probably wouldn't stick it in a kick drum. The sensitivity is a fairly typical 15mV/Pa. The MXL 990 weighs 1.2Ibs and measures 60 x 130mm.

As a vocal mic, the MXL 990 produces a subjectively neutral and nicely balanced sound with just a hint of high-end sizzle that can help out vocalists who have difficulty getting the definition they need. I used it on a serious session with a respected New Zealand female vocalist, and it suited her voice very well. It is subjectively quiet in this application and delivers a reasonable amount of weight without sounding muddy or out of focus. Being ruthlessly honest, I'd say that a lot of similarly priced Chinese mics offer similar characteristics, but that in no way detracts from the fact that this mic is capable of fine results when used carefully. It also works well as a voice-over mic and handles acoustic instruments with no apparent problems, though, as usual, a pop shield is essential when recording vocals.

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MXL 992

Interestingly, the cardioid MXL 992 has a similarly sized basket to the 990, but utilises a larger 25mm capsule, and the body is physically around twice the length of that of its sibling. Again both a shockmount and a stand adaptor are included, but this time the case is aluminium rather than plastic. Once again the construction is what we've come to expect from most Chinese-built microphones insomuch as the main glass-fibre circuit board is bolted to two metal struts that also support the XLR output connector. A second circular board just below the basket holds the 10dB pad and high-pass filter switches — the filter rolls off at 6dB/octave below 150Hz. The circuitry is again all discrete and the output stage is transformerless. Both the switches are miniature toggles that protrude from the body just below the basket.

The capsule's six-micron diaphragm provides a nominally flat response from a little over 20Hz to 20kHz, with a subtle presence peak centred around 8kHz. In general this is similar to that of the MXL 990, but the graph shows the response curve to be noticeably smoother and flatter. Equivalent noise level is lower than average at 10dBA, and while the maximum SPL isn't as high as some (130dB, the same as for the MXL 990), it's more than enough for typical vocal and instrument applications. The sensitivity is again 15mV/Pa and the overall weight is 2lbs.

By way of performance, the MXL 992 is quieter and smoother sounding than the MXL 990, with a less obvious high-end sizzle, but otherwise it feels quite similar in character. Where the extra high-end bite isn't needed, I feel it produces a more natural, homogenous sound than the MXL 990, but that shouldn't detract from the fact that the MXL 990 is still a very competent mic.

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Summing Up

While the V6 costs a little more than typical 'me too' studio mics, it is in no way expensive and its distinctive styling is refreshing in a world where everyone seems to borrow from the original layout of Mr Neumann's classic models. If you're expecting a thick, treacly tube sound, then you may feel disappointed, but in my experience it's very hard to tell a good tube mic from a good solid-state mic anyway — the most obvious-sounding tube mics are usually deliberately hyped and may end up being one-trick ponies that suit only one particular style of vocal. Certainly the V6 has a lot in common with the V69, though you can just about tell them apart in a direct comparison.

If you're after an affordable all-rounder that will sound good with a range of vocal styles and instruments, then the MXL 990 is a strong contender, though as pointed out earlier there are lots of similar-sounding mics at a similar price. The more expensive MXL 992 has a smoother, classier sound (as well as pad and roll-off switches) and still offers good value. My only issue with many of the current Chinese-built microphones is that their circuitry is often no quieter than the classic mics designed fifty years ago, and while the MXL 992 has a good noise spec, I can't see why the other two shouldn't be just as quiet given the improvements in components, circuit design, and assembly methods over the past few years. In terms of UK pricing, the MXL 990 and 992 aren't any cheaper than most of their competition, although the street price appears to be quite heavily discounted by many dealers. SOS

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pros

- Fair selection of presets.
- These versions sound as good as the full ones.
- Affordable.

cons

- Lack of control can be frustrating, especially with *Pro 53 Xpress*.

summary

When you need a classic keyboard sound fast, *Xpress Keyboards* might just be the answer.

information

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T Arbiter +44 (0)208 207 7880.

W www.native-instruments.com

NI Xpress Keyboards

Software Synths

Published in SOS August 2005

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Reviews : Keyboard

Sam Inglis

Native Instruments' *Xpress Keyboards* features cut-down versions of their *B4* tonewheel organ emulation, *Pro 53* Prophet 5 copy and *FM7* DX-alike. What's cut down is not the sound, but the amount of control available to the user. You can load presets and make tonal changes to them, but can't program your own sounds from scratch.

About 50 presets are available for each of the three instruments. Most of them are very usable, and deliver a good selection of classic Hammond, Prophet and DX7 sounds. In each case, six rotary controls provide reasonable room for manoeuvre when it comes to sonic tweaks.

B4 Xpress allows you to adjust the amount of overdrive, percussion and vibrato, and to switch the virtual Leslie on or off and between fast and slow settings. The obvious difference compared to the full-blown *B4* is that there are no drawbars; tone controls are limited to Bass, Treble and Brilliance. This isn't too much of a problem, because the presets cover a decent range of drawbar settings, but you can't do those classic swells and fades, where you move the drawbars whilst holding notes.

The lack of control is more frustrating with *Pro 53 Xpress*. Anyone who's familiar with subtractive synthesis will find themselves thinking 'I'll just change that LFO speed and release time... oh.' The six parameters affect overall brightness, filter resonance, envelope attack and decay, 'Shape' and effects level, and don't offer enough control to be useful. A subtractive design like the Prophet 5 is pretty simple anyway, and there's not much to be gained by limiting the options in this way.

By contrast, the DX7 is notoriously difficult to program, and the way NI have boiled the controls down for *FM7 Xpress* works well. As well as envelope attack, decay and release times, you can adjust the overall brightess and the amount of harmonic content in the sound. There's an effects wet/dry mix knob (the effect in most presets is delay). Provided there's a preset that's in the right ballpark, these controls will probably be enough to get you where you want to go, and they'll certainly let you get there quicker than you would have done with a DX7! SOS

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The World's Best Music Recording Magazine

Plug-in Folder

Mini-reviews of hottest new Plug-ins

Published in SOS August 2005

Anarchy Sound Software *Anarchy Rhythms* Formats: Windows VST & stand-alone

Anarchy Rhythms is a novel VST plug-in, which can also be run as a stand-alone application, and which its author accurately describes as "a hybrid between an effect and a drum machine". It takes an incoming audio signal and applies a mixture of effects including amplitude modulation, band-pass filtering and compression, while a built-in step sequencer provides rhythmic modulation of several key parameters. Internally synthesized tones can also be mixed in with the sound.

Anarchy Rhythms' user interface is quite clear and well laid-out, but nevertheless has plenty of controls crammed in. Fortunately, well written and nicely laid-out help files are included, which provide clear and straightforward explanations of what everything does.

At the top of the interface is the main pattern sequencer grid. By default this shows a single four-beat bar divided into 16th notes, although other variations are possible. The grid can be set to anything between one and 16 beats, with odd numbers allowing for some interesting polyrhythmic effects. The number of 'sub-steps' per beat can also be varied from one to eight, allowing for still more complexity. A 'shuffle' slider allows the phrasing to be loosened up considerably; the grid lines redraw themselves as shuffle is increased, giving a clear visual indication of how the phrasing is changed.

The sequencer grid features four 'instrument' channels, each of which corresponds with a kind of virtual mixer channel carrying a duplicate of the original input signal, optionally combined with that channel's own synthesizer. Each active step in the sequence opens a gate, triggers the synthesized sound, and allows the incoming signal through. When a step is empty the gate remains closed and no sound is heard.

The four channels are independent, colour-coded (with editable names), and all feature the same set of parameters. Each one has a 'channel strip', with a level fader and Mute and Solo buttons. There are also Len and Feed knobs, which respectively control the duration of each step and a 'feedback' parameter allowing sound to be 'carried forward' from one step into the next, to create repetition or echo effects. Comb and Comp control the amounts of comb filtering and compression applied; Filter and Osc control the width of a band-pass filter and the level of an in-built oscillator, while Pan and Pitch control stereo panning and oscillator frequency. The latter is calibrated in MIDI notes rather than Hertz.

To the right of the mixer channels are more parameters, including five graphical envelope displays. These are also colour-coded, changing to indicate which of the four channels the current settings apply to. There are envelopes to control volume, pan, filter width and oscillator pitch, as well as an envelope to allow dynamic interpolation between two different oscillator types.

Even when fed with the most uninteresting signals, *Anarchy Rhythms* is capable of producing some surprising results. I began testing with a simple sine wave, and spent a happy 20 minutes before remembering I could use other sound sources as well. Drum tracks can be utterly demolished, while vocals, strings and pads can likewise be chopped up and generally mangled to death. In fact almost any sound could probably be turned into something interesting with *Anarchy Rhythms*. It's a fairly 'blunt' instrument — don't expect much in the way of ethereal subtlety — but it can certainly be effective. The 'anarchy button' patch randomiser makes searching for new effects all but effortless, generating an entire bank of new presets with a single click.

Anarchy Rhythms is an unusual plug-in, with a range of possible applications. It's also easy to use, enjoyable, and might be just the thing to spark your imagination on those difficult days when inspiration is in short supply. The stand-alone version is a nice addition, and could be useful for real-time sound processing in a live situation. In an overcrowded plug-in market, *Anarchy Rhythms* stands out as something pleasingly different, and at just £29.99 represents good value for money. It's available for Windows (versions 95 to XP) and a free demo version can be downloaded from Anarchy Sound Software's web site. *Paul Sellars*

£ £29.99.
W www.anarchysoundsoftware.co.uk

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In this article:

Roland FANXUP1 & Fantom Xr

OS Upgrade + v2 Editor For Fantom X

Published in SOS August 2005

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Reviews : Sound Module

The Fantom X is Roland's best-ever workstation, but it has suffered from one or two annoying omissions, such as the ability to import Roland's own sample format. Can the v2 OS and editing software provide the solutions?

Nick Magnus

The FANXUP1 Sample Tools Expansion Kit is an operating system upgrade for the Fantom X range of workstations, namely the X6, X7, and X8 keyboards (reviewed in *SOS* September 2004, or at www.soundonsound.com/sos/sep04/articles/rolandfantomx.htm), but not the cut-down Fantom Xa. The upgrade is also available for the rackmount Fantom XR, in which case it is more properly known as the FANXRUP1 upgrade, but both versions offer very similar new features for both keyboard and rack Fantoms, including improved Fantom X Patch/Performance editing software, a Fantom Librarian, proper large-scale, on-screen sample editing, a multisample editor and — at long last — an S700-series sample-library converter. The major difference between the rack and keyboard upgrades is that the keyboards' sequencers receive the benefit of eight fully sync'ed audio tracks. These are not available in the XR Rack upgrade, which is understandable, as the XR has no onboard sequencer, but rather less fairly, the XR upgrade is the same price than the keyboard one.

Photo: Mark Ewing

The upgraded Fantom XR used for this review, with the additional *Sample Tools* printed manual and the bundled disc containing the v2 *Editor* software.

Roland supplied *SOS* with an XR Rack pre-installed with the FANXRUP1 kit, so this review pertains to the XR version, and there will be no discussion of the eight-

Roland FANXUP1 £49

pros

- Patch and Performance editing is very much easier on the Fantom XR than it was.
- The new software offers clear, on-screen sample and multisample editing.
- Roland S700-series libraries can be imported.

cons

- Sample import involves frequent, inconvenient USB replugging, and rebooting of the Fantom.
- Converted S700 sample library TVF and TVA settings are incorrect and have to be edited, and imported WAV-file loop points are still not recognised.
- Alternate looping is not supported.
- No difference in price for the XR version (which does not include the eight audio tracks).

summary

This upgrade brings significant new functionality to the Fantom X. However, certain aspects of the Fantom design have led to a clunky and inelegant implementation of the Roland sample-import facility, giving the impression that it has been put together as an afterthought. Nevertheless, this is an essential purchase to bring your Fantom up to spec, especially at the relatively low cost.

information

£49 including VAT.

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www.roland.co.uk

track audio features. Aside from this, though, comments pertaining to the XR also apply to the X6, X7 and X8.

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Fantom X Editor

The v2 *Fantom X Editor* now has three principle parts: a Patch/Performance editor, a sample editor and a Multisample editor. The Patch/Performance editors (shown on the right) are just as easy to use and immediate as those in the previous version of the Fantom X. Once you have specified the appropriate communication port, you go to the Edit menu on the toolbar and select 'Synchronise'. All the XR's data is transferred to the editor, and you can then select a Patch or Performance to edit. The *Fantom X Editor* makes getting around the XR a breeze — and infinitely preferable to the button-intensive menus and small display found on the XR itself. If there is one complaint to level here, though, it is that control over many of the parameters is too coarse. All too often, the exact values you wish to set are 'skipped' by the knob or fader, and no means is available to make 'fine' settings. So you have to delve into the XR's own edit menus to set them precisely — which somewhat defeats the object of having this Editor!

The Patch Editor screen from the *Fantom X Editor* showing the LFO and Patch Common details. The Performance editing screen can be seen below, with its optional keyboard.

The Fantom X's Version 2 OS now allows fully detailed individual Patch editing from within Performance Parts, which is good news. However, edited Parts must be saved 'by hand', one by one — there's no option to write these using Performance Write (unlike on the earlier JV, XP and XV synths) which is not quite so convenient.

The upgraded Editor adds two new screens: the Sample Editor and the Multisample Editor. Both of these require samples to have been loaded into the XR. These can exist either in the XR's User RAM or be resident on a inserted Compact Flash card. Note that samples in the Preset bank must be saved to the User or Card memory area first before they can be edited.

The Sample button at the top-right corner of the editing software invokes the Sample Editor in a separate window. Whenever this is running, the LCD display on the XR shows the words 'PC Mode' and the XR's

hardware controls are locked out. The
Sample Editor window (shown overleaf) is in two halves — on the left-hand side is the sample list, with buttons above to show either the User bank or the Card bank sample list. On the right-hand side are two waveform displays — the upper display is an overview of the entire selected wave, whilst the lower display is the 'detailed' view, and can be zoomed either vertically or horizontally using the '+' and '-' scaling buttons. The waveform can also be zoomed horizontally by dragging the left or right extremes of the yellow boundary box in the waveform overview.

Below the waveform displays are numerical fields for entering sample start, loop start and end points. These points can also be dragged into position in the waveform window, or moved very precisely using back/forward nudge buttons. Each point has its own speaker icon, and clicking on one of these causes the wave to play back in a continuous loop from that point for auditioning purposes. Below these are further settings for loop tuning, wave gain, sample fine-tuning, sample level, original key and loop mode. Four loop modes are available: One-Shot, Forward, Reverse Loop and Reverse One-Shot. Curiously, the Fantoms do not support Roland's Alternate (bi-directional) looping option — more on this later.

At the bottom of the screen, a Preview button allows you to audition what you have been doing to the sample, whilst the remaining six buttons open separate windows to perform 'hardcore' editing functions. The time-stretching, truncating, emphasis, normalising and sample amplitude processes can either overwrite the existing wave data or make new versions of the processed waves. The 'chopping' process in particular benefits from this visual form of editing — chop points can be added, deleted and dragged with the mouse, and the resulting 'chunks' can be auditioned via 16 virtual pads, similar in appearance to those found on the keyboard Fantoms (see below). It is important to remember that none of these sample edits become permanent until they are written to User or Card memory. This must be done via the Fantom XR's panel controls — the Editor provides no Write functions. Similarly, other sample-management tasks such as deleting and loading samples have to be performed on the XR itself, so be prepared to do a certain amount of 'menu surfing' on the main hardware unit.

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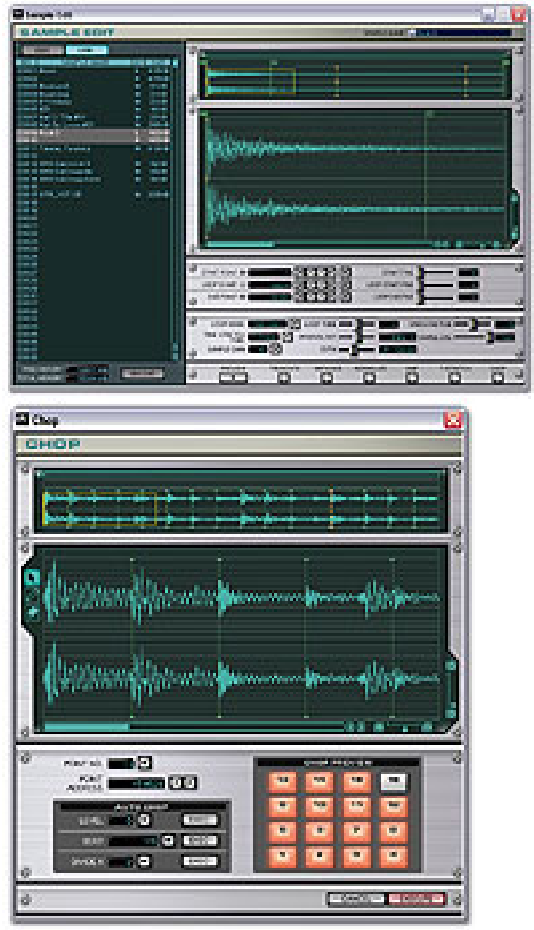
The Multisample Editor

Editing multisamples on the XR was never an entirely intuitive process. The update adds some enhancements to make the task a little more bearable on the XR's small LCD display, but nevertheless it's still very fiddly compared to the clear, on-screen method enjoyed by Roland's original S-series samplers. The new Multisample editing screen harks back to those halcyon days, providing an efficient, large-screen means of assigning samples to key ranges in order to create a playable multisampled 'instrument'.

The Multisample Editor button is located in the Patch WG section of the *Fantom X Editor*. If a Patch contains no multisamples, the button is inactive (greyed out).

To activate it, a multisample must first be created using the XR's 'Create Multisample' function, and of course this requires that some samples have been loaded into the XR. It doesn't matter what samples the multisample you create contains, just as long as you create one. Alternatively, you could select an initialised multisample as the waveform source in the WG section. Once a multisample is in place, you can access the Multisample Editor and adjust this multisample to taste, or even go on to create as many additional new ones as you need.



The Sample Editing screen in the v2 Editor, and (below) the Chop screen with its slice-auditioning pad.

As you can see in the screenshot of the Multisample Editor over the page, the editor screen consists of three sections — the sample list on the lower left, the Zone list on the lower right, and the keyboard-mapping tool at the top. To create a new multisample, you select any initialised (empty) multisample from the drop-down menu to the right of the multisample's name — these are all named 'INIT MSMPL'. The Zone list and keyboard map will then show as empty. If you then drag the mouse across the keyboard map for the desired range of your first sample, a red boundary box appears above to reflect this range. To assign a sample to this range, you just select a sample and click the Assign button. The red box turns orange to show an assignment exists for that Zone, and a small orange arrow between the red box and the keyboard shows the original or root key of the sample. You can drag this arrow over the desired key if you wish to change it. The Zone box shows the same info in a text list, and here you can also change the original key, level and tuning of this Zone. Nudge buttons are also provided as an alternative means of setting the upper and lower range of the Zone. Zones can be erased individually or *en masse*, and swapping one Zones' sample for a different one is as easy as selecting a Zone in the list, selecting a new sample from the sample list, and clicking Sample Select. Finally, the Oct buttons to the left of the keyboard shift the keyboard's viewing range by up to ±2 octaves, whilst the Shift button nudges the selected Zone's range up or down in semitone steps. As with edited or newly imported samples, edited multisamples are only temporary, and must be written to Card or User memory to be stored — there is room for 128 multisamples in each of the User and Card banks.

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Installing The Update

All recent Fantom Xs ship with the upgrades pre-installed, but if you're performing the upgrade on an older X, the FANXUP1 kit will supply you with a Compact Flash card containing the upgrade software, and a PC card adaptor. To install the upgrade, you plug the Flash card into the PC card adaptor, insert this card assembly into the XR, power up and follow the on-screen instructions. Afterwards, the Compact Flash part should never be needed again, but don't lose it just in case — and the adaptor will be required if you wish to import compatible wave data from any other Compact Flash card. The XR unit provided by Roland had already been pre-installed with the upgrade, so I can't comment on the ease or otherwise of this upgrade process, but it sounds pretty straightforward. The next step is to install the XR's USB driver on your computer. This enables the XR to communicate with all the included software utilities, as well as providing an additional USB MIDI I/O port in your sequencing software. Before installing the driver, the XR must first be set to USB MIDI mode in the System settings, and then powered off. You then connect the USB cable to the computer, power up the XR and install the driver. See the box on USB modes over the page for more on this.

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S770 File Converter

This separate application (shown on the opposite page) enables Roland S700-series sample libraries to be imported into the Fantom XR. Converting and loading these Patches is quite an involved process, and it can be rather off-putting if you simply wish to audition material. Firstly, the XR must be in USB Storage mode in order to convert the Patches, with all the attendant USB cable unplugging and powering on and off (see the 'USB Modes' box below). Next, USB Storage is selected from the XR's main menu to establish a USB link to the computer. If all is well, a drive icon appears on your desktop named 'FANX_USER'. This contains a folder structure, within which is a sub-folder named 'PATCH_IMPORT', and it is in here that the converted Patches must be saved. Now you can insert a native Roland format sample CD-ROM and fire up the *S700 File Converter* utility. As you can see in the screenshot on the opposite page, the user interface for this is very simple — all Patches on the disk appear in the lower list (and can be sorted by type or alphabetically). For the most part, you'll want to leave the sample-conversion method on the Auto setting. You select the desired Patch from the list, or hold down the Shift key and click to select multiple patches. When you've selected everything you want, you then click 'Convert' and choose the 'PATCH_IMPORT' folder as the saving destination. However, you mustn't change the file name from 'IMPORT.SVD'! The manual is not clear about this process; only one .SVD file is written, and it *must* be named 'IMPORT.SVD', even when you are converting several Patches in one pass. My first impulse was to convert each Patch one at a time and save each file with a name of my choice, but this does not work. When conversion is complete, and before you can proceed with the actual import, the XR's desktop drive icon must be 'ejected' to terminate USB communication with the PC. Then, and only then, you enter 'Import Patch/Rhythm' from the XR's Utility menu to import the converted Patches. However, the task is not yet over — in order to play or edit your imported samples, you must return the XR to USB MIDI mode, with all the usual unplugging and powering on and off... For this reason, it's a good idea to always maintain a five-pin MIDI connection with the XR (in addition to the USB one) so you can at least audition what you've just loaded without having to change USB mode.

Incidentally, individual Roland partials and samples can also be converted using this utility, the only differences being that the converted files must be saved into the folder named 'AUDIO_IMPORT', and imported to the XR using its Import Audio function. The imported files are added to the sample list, and as always, they must then be saved before powering off the XR, or they'll be lost.

Editing Multisamples can now be taken care of in this new screen — certainly an improvement over using the diminutive display on the Fantom XR!

So how successful are the Patch conversions? Well the good news is that keymapping, pitch settings, velocity crossfades and switches make the journey intact. For Patches using switched or layered samples, each layer appears as its own multisample, which can be further tweaked if necessary using the Multisample Editor. The bad news is that TVA and TVF (ie. amp and filter envelope) settings appear to be ignored completely — both the filter and amplifier parameters default to a 'vanilla' setting, and need to be set up from scratch. Since the Fantom synth architecture is not significantly different from Roland's own XV5080 synth/sampler, which manages to load Roland sample data with complete accuracy, it's a mystery why the Fantom is unable to do the same. Another problem is that the Fantoms, as mentioned earlier, do not support Alternate sample looping, so any imported samples that use this are interpreted as having a Forward loop. Four out of 10 — see me afterwards!

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Sample Auto Patch

In addition to the User, Card, Preset and GM Patch groups, two new Patch groups have been added, called USAM and CSAM. These groups allow a single sample to be played like a normal Patch, so samples need to be stored in either the User or Card memory, or both, for these Patch groups to work. Each USAM or CSAM Patch's sample is automatically picked from the sample memory in numerical order — so USAM 001 will play sample 001 from the User memory, USAM 002 will play sample 002, and so forth. If no sample exists in a particular location (if there is no sample in slot number seven, say), its 'host' Patch will display 'USAM 007 - -' and no sound will be produced. By the way, these two Patch groups set their samples' Tempo Sync parameter to 'on' by default, so unless you're playing drum loops this way, you'll want to turn Tempo Sync off, otherwise it becomes impossible to play the sample far beyond its root key without it sounding like a broken record!

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USB Modes

The XR has two USB Modes in its System settings, MIDI mode and Storage mode. MIDI mode is used for normal playing or sequencing (when using the USB I/O port, instead of the traditional five-pin MIDI I/O ports) and also for using the editing and Librarian packages. Storage mode is for transferring files, importing samples directly from a computer, and for importing Roland's S700-series sample libraries (although the XR can be in either Storage or MIDI mode when importing ordinary wave data from Compact Flash cards).

It should be mentioned that if you are frequently swapping between Patch editing and PC file-importing tasks, the XR's USB mode has to be changed on a frequent basis. This involves the tedious process of powering off, unplugging the USB cable, powering on, changing the USB mode, saving the System settings, powering off, plugging the USB cable back in, and powering up once again — all of which the manual insists must be adhered to rigidly. After a while it all becomes a bit much, especially considering that many users will mount their XR in such a way that the rear USB connection will be inaccessible, and some will also have their computer in a soundproofed cabinet, or even located in a separate room. All of this simply causes inconvenient and undesirable breaks in the musical workflow.

Additionally, the manual insists that the XR must always be powered up last for the USB connection to function properly, and that the USB cable should never be connected or disconnected while the XR is powered up. The manual even suggests powering off the computer when disconnecting the USB cable — and, frankly, life's just too short. So I tried leaving the XR powered up when replugging the USB cable, and sure enough, I always ended up having to reboot the XR in order to re-establish USB communications with the computer — as well as having to restart my sequencer to get the USB MIDI ports back on line again!

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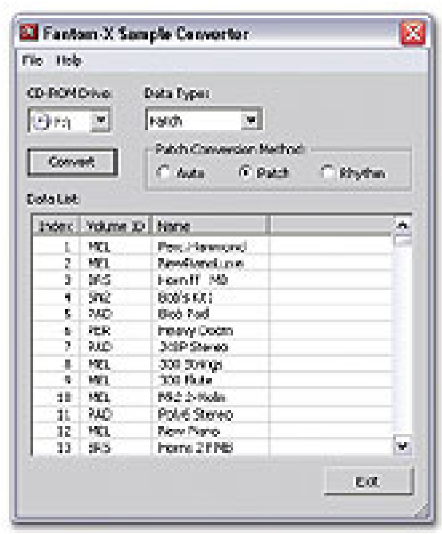
Conclusions



Firstly, let me say that everyone who owns a Fantom X6, 7 or 8, or XR should definitely buy this upgrade. For a modest outlay you'll receive the benefits of graphical sample editing and multisample editing, an improved Fantom X editor and the ability to load Roland format sample libraries — albeit imperfectly. Why this last feature was ever absent from the original Fantom specifications is a mystery — and this is probably at the root of the frustrations encountered whilst using this particular feature of the upgrade. If Roland sample-library imports had been central to the original Fantom design, and the import process had been streamlined so that file-conversion and importing was a single, seamless operation, then perhaps the whole Fantom-to-computer file-transfer processes would have been a lot less frustrating. As it is, far too much replugging of USB cables and rebooting of the Fantom is necessary to access these facilities, so the criticisms must ultimately be directed towards the Fantom design itself, rather than this upgrade package which has of necessity been built around the Fantom's inherent idiosyncrasies. That said, it must be reiterated that this is an essential upgrade for existing Fantom X owners. SOS

Converting S700-series samples to run on the Fantom is finally possible — albeit in a rather complicated way...

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In this article:

Roland VS8F3 £299

pros

- Runs a selection of respected third-party plug-ins within the VS environment.
- Five useful Roland plug-ins bundled with the card, including some nice analogue-modelling algorithms.
- Increased 56-bit processing resolution compared to the VS8F2.
- Effects can track tempo of host multitracker.

cons

- Plug-ins and their patches cannot be controlled by the onboard VS automation.
- Tempo detection will not work if you're using MIDI Time Code for synchronisation.
- Lack of stereo linking/ unlinking facilities in some of the plug-ins limits their usability.
- Roland's choice and ordering of processing blocks in the bundled plug-ins don't always make a great deal of sense.

summary

Even without third-party plug-ins, the VS8F3's extra processing fidelity and bundled plug-ins are easily worth the outlay, notwithstanding the odd operational niggle. However,

Roland VS8F3

Plug-in DSP Card For VS-series Multitrackers

Published in SOS August 2005

Print article : Close window

Reviews : Effects

This new card narrows the gap between Roland's VS-series machines and computer recording systems by allowing the use of third-party plug-ins within the multitracker environment. We take a look at the card, its bundled plug-ins, and the first of the brand-name offerings from Universal Audio.

Mike Senior

For a while hardware multitracker manufacturers have been trying to defend their corner against the lure of computer recording systems. One of Roland's latest strategies, announced back at the NAMM show 18 months ago, is the VS8F3 card, a DSP processing card designed to run third-party software plug-ins. Although this card has now been available for some months, complete with a bundle of Roland plug-ins, the third-party support has taken a little longer to materialise. However, the first third-party plug-ins are now beginning to arrive in the UK, so I decided to get hold of the VS8F3 and test it out with Universal Audio's new VS-compatible compressor plug-ins, *VS1176LN* and *VSLA2A*.

The VS8F3 card can be used with the VS2480, VS2400, VS2000, VS1880, VS1824, and VS1680, although you'll want the latest version of the

the facility to run third-party plug-ins from some of the leading manufacturers should make this product hard to resist for almost any VS-series multitracker owner.

information

£ Roland VS8F3, £299; Universal Audio *VS1176LN* and *VSLA2A*, £134 each. Prices including VAT.

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Test Spec

- Roland VS2480 multitracker running OS v2.5.
- Roland VS8F2 and VS8F3 cards.
- Universal Audio *VS1176LN* v1.0 and *VSLA2A* v1.0.
- Urei 1176 compressor (black-face model).
- Universal Audio 1176LN compressor.

multitracker's operating system in each case to ensure compatibility. Once you've clipped the VS8F3 card into the recorder, it's straightforward to install the five bundled Roland plug-ins from the included CD: *Tempo Mapping Effect*, *Vocal Channel Strip*, *Preamp Modelling*, *Stereo Reverb*, and *Mastering Tool Kit*. The plug-in authorisation process links plug-ins permanently with the VS8F3 card located in the first of the internal card slots — called the Key Card. The installation CD actually seems to be an unfinalised CD-R, and the identity of the Key Card is burned to this during the authorisation process. After installation, the resulting CD-ROM will only install plug-ins which can be used with that specific VS8F3 Key Card.

What this means in practice is that you can use your plug-ins only on the machine where your Key Card resides. To use the plug-ins in two different VS machines simultaneously you need two installation CD-Rs. You can have as many VS8F3 and VS8F2 cards as you have free slots in your particular machine, and if you have more than one VS8F3 in your recorder, authorising the plug-ins for the Key Card lets you use them on all the other cards as well.

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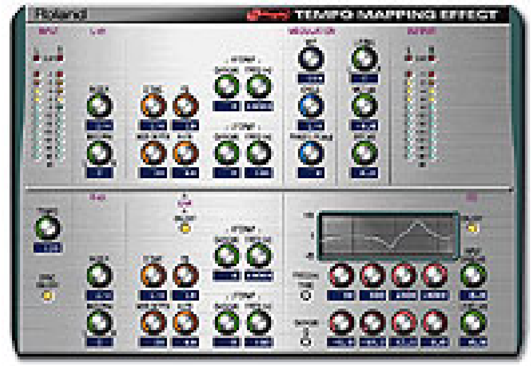
Using The VS8F3

At best, the VS8F3 card will run a different two-channel plug-in within each of its two effects slots, allowing you to divide the four processing channels between a selection of mono and stereo signals as you see fit. However, some plug-ins require so much processing power that they hog the whole card — of the plug-ins under review here, *Tempo Mapping Effect*, *Stereo Reverb*, and *VS1176LN* fall into this category. Furthermore, some of the plug-ins only operate in stereo-linked mode, and this is really annoying in the case of the Universal Audio plug-ins, because one of the channels is wasted when compressing mono signals.

Notwithstanding the odd limitation like this, the new card is certainly more powerful than the VS8F2, not least because it can operate at the same 56-bit internal resolution as the VS mixer, which will perhaps convince more people to use the VS8F3's *Mastering Tool Kit* than that of the 24-bit VS8F2. It will also function at up to 96kHz, but at higher sample rates each VS8F3 card only offers one effect slot. Furthermore, some plug-ins are too processor intensive to be run at all. For example, the *VS1176LN* will only run at up to 88.2kHz.

One side-effect of the new DSP hardware is that the real-time spectrum analyser and RSS panning functions available using the VS8F2 on some of the VS-series machines cannot be run on the VS8F3. It's also currently not possible to change plug-ins or patches under the control of the VS2480's Automix dynamic automation.


Roland's bundled *Tempo Mapping Effect* can detect the tempo within your multitracker project, and then use this to set delay and modulation times.

The graphical interfaces for the various plug-ins as shown on the optional VGA monitor can be seen from the screenshots. On the LCD, given the more limited display space, the effects parameters for the Roland plug-ins are split up into pages in a similar way as on the VS8F2, but without the useful blocks overview which allows you to easily bypass the different effects in a chain. However, where the VS8F2 editing pages have very little in the way of graphical niceties, the VS8F3 interface is heavily inspired by the 'virtual front panel' style of computer plug-ins. The advantages of the visual overhaul are that you get much more metering in the plug-in windows, the lack of which on the VS8F2 was a long-time complaint of mine. However, the disadvantage of the snazzier look is that opening up and switching between the plug-in pages is a bit sluggish.

Something to be aware of with the new non-standard graphics is that although the bundled Roland plug-ins show the current parameter highlighted, the Universal Audio ones don't. You can just about navigate between *VS1176LN* or *VSLA2A* parameters 'blind' using the cursor keys, but in practice the mouse ceases to be an optional extra when using these plug-ins. You can still do without a VGA monitor, however, and all the plug-ins I've seen so far seem to operate fine on the LCD display. Another minor niggle with the Universal Audio plug-ins is that you can't open up the effect parameters page while the song is playing, even though you can edit the effect during playback if that page is already on screen.

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Tempo Mapping Effect

Now let's have a closer look at the bundled plug-ins. A less well publicised improvement provided by the VS8F3 hardware is that it can detect the host multitracker's tempo setting, and *Tempo Mapping Effect* has been created to take advantage of this. Each channel has two delay lines, each having a delay time of up to a second, expressed either in milliseconds or as a note duration ranging from 1/1 (a whole note or semibreve) to 1/64t (a triplet 64th-note or triplet hemidemisemiquaver). The channels can be linked for stereo work.

The second delay line in each channel is the interesting one, having its own feedback and cross-feedback paths with high- and low-frequency damping. On top of this, its delay time can be modulated with a choice of four waveforms (sine, square and both directions of sawtooth), at a rate which can be set in Hertz or in terms of note duration — again, any setting between 1/1 and 1/64t is allowed. You can even set the relative modulation phase of the two channels, which allows for some nice stereo treatments. Following the delay and modulation processing, a four-band equaliser can be applied to the signal before it is returned to the VS mixer — this EQ has a choice of nine different filter responses for each band, but pretty much mirrors the channel EQ in terms of sound.

Preamp Modelling is probably the highlight of the Roland plug-ins, combining dynamics and EQ with a processor which models analogue preamp circuitry.

The delays are clean and the modulation is very smooth and predictable. Because the delay times can be adjusted all the way down to zero, you can create phase, flange, and chorus effects, in addition to all kinds of complicated delays. Tweaking the delay time while the track is playing causes the effect to slew to the new tempo, rather than creating any nasty glitching sounds, and this offers some great creative possibilities. I also like the fact that the tempo-related modulation speeds aren't restricted to the maximum that you can set in Hertz (10Hz) — you can create something akin to frequency modulation by dialling the tempo setting really high manually, and then selecting 1/64t as the modulation rate.

The only operational quirk I encountered was that the plug-in initially refused to recognise the tempo of my project. It turns out that you need to set the VS2480 to output MIDI Clock messages to get it to work. If you need it to send out MIDI Time Code instead, as I do to synchronise with my sequencer, the automatic delay-time detection doesn't work.

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Vocal Channel Strip & Preamp Modelling

The *Vocal Channel Strip* takes its inspiration from the VS8F2's *Vocal Multi*, but where the VS8F2 offered a single-channel effect chain with a stereo chorus at the end, the VS8F3 opts for a more sensible dual-channel mono setup. However, the plug-in cannot be linked for stereo operation, despite the presets apparently designed for processing stereo sources! Even more illogically, the Bypass button at the bottom of the plug-in parameter screen bypasses both channels together. It seems that Mr Common-Sense was on holiday at that stage in Roland's software development process...

The effects chain features compressor, expander, enhancer/de-esser, equaliser, pitch-shifter, chorus, and delay, in that order. I would question the positioning of the expander after the compressor in this chain, because compression modulates the noise floor, presenting the expander with a moving target. Another problem is that you can't use the enhancer and de-esser simultaneously, which is a shame given that psychoacoustic enhancement often necessitates the use of a de-esser.

The compressor here is a new design for the VS8F3, modelled on analogue electronics, and you get a choice of five different compression characteristics: one solid-state design; two valve designs adding even-order harmonics; and two valve designs adding odd-order harmonics. You can also decide between soft-knee and hard-knee compression curves. Otherwise, the controls are as you'd expect of the compressor in the VS mixer, and input, output, and gain-reduction metering are all present and correct. Comparing the sound of the compressor block with the channel compressor, the solid-state option sounded pretty similar, while the four tube options all subtly enhanced the signal, even when the effect was driven quite hard. I liked what this compressor could do for vocals, making them both more solid and crisper, and I can imagine using it a lot. The expander block is identical to the mixer-channel version.

Although the new *Stereo Reverb* algorithm is clearly a step up in quality compared to the VS8F2's *Reverb*, the pre-reverb dynamics blocks may be rarely used in practice.

The enhancement and de-essing processes share detection-frequency and sensitivity settings, but have independent level controls. The enhancer was a very pleasant surprise, and I suspect that it works very differently to the one already available on the VS8F2, because it's much better at brightening up a signal without sandpapering your eardrums into the bargain. The new de-esser is also fairly good, operating only on a specified upper region of the frequency spectrum and displaying less of a tendency towards lisping than the ones on the VS8F2. My only wish was for a virtual indicator LED to show when processing was active, as this would have made setting things up much easier.

Following the four-band EQ (the same as in *Tempo Mapping Effect*), the remaining blocks offer exactly those facilities found in *Vocal Multi*: pitch-shifting of up to an octave either way, with cent resolution; chorusing with rate, depth, and pre-delay settings; and delay with time and feedback parameters. All three of these blocks also have level controls for the wet and dry signals, so that you can choose how much of the effect you want to hear.

The pitch-shifters of *Vocal Multi* and *Vocal Channel Strip* are pretty similar — I found that I actually preferred the older one for polyphonic material, but there wasn't much in it. On the other hand, the chorus effects are like chalk and cheese! Where the wet sound of the older block was always a fake 'chorus-flavour drink' effect, sounding heavily blurred and out of tune with modulation, the VS8F3 chorus gives you the proper freshly squeezed organic version — a single clean double-track which modulates smoothly. Although the VS8F2's chorus effect was passable on occasion, this one deserves a lot more use. There's not much to say about the delay line, which does what it says on the tin, but there is one more thing to mention before moving on — where the VS8F2 offers negative values for feedback, effect, and dry levels in the pitch-shifter, chorus, and delay effects, the VS8F3 doesn't.

The *Preamp Modelling* plug-in ditches the last three blocks from *Vocal Channel Strip* and replaces them with a processor which attempts to emulate the sounds of classic analogue preamps, including (if the less-than-cryptic parameter names are to be believed) units from Avalon, Focusrite, Manley, Millennia, and Neve. The first way it attempts to do this is via a kind of two-band EQ, comprising a Warm band and a Bright band adjustable over 20Hz-2kHz and 200Hz-20kHz respectively, both bands having ±15dB gain range. The other means for changing the sound are three Harmonics controls, which can be used to add various degrees and colours of harmonic distortion.

Reducing the level of harmonics to zero, I first had a play with the EQ controls, and found them to offer slightly more of a tonal change for a given setting than the high and low bands of the channel equaliser, although the difference was more subtle than I was expecting. At this stage there was little difference to be discerned when switching between the different preamp models, but as soon as the harmonics were added back in they all took on distinctly different characteristics. We're not talking massive changes here, but it makes this a more musically interesting alternative to EQ. Until now, the only real option for this kind of tone tinkering was a low-gain *Guitar Amp Simulator* patch on the VS8F2, but now you've got a much greater range of usable flavours to choose from.

The combination of compressor, enhancer/de-esser, and preamp modelling is a powerful one, and beats the socks off the VS2480's channel facilities for giving you that 'more of everything' sound that digital recordings can often lack. I can see *Preamp Modelling* becoming a firm favourite of mine, especially as you can link it for stereo operation, unlike *Vocal Channel Strip*.

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Universal Audio *VS1176LN* & *VSLA2A*

The two Universal Audio plug-ins under review here are the *VS1176LN* and *VSLA2A*, which are modelled on the company's own premium hardware recreations of the classic Urei 1176 and Teletronix LA2A compressors. These shouldn't really need much in the way of introduction for regular readers — barely an issue seems to go by without one being mentioned in an *SOS* interview. However, for those new to these two units, they make a good pairing: on the one hand, the 1176 is well known for its in-your-face sound, a result of its FET gain-reduction element and savagely fast time constants; on the other, the LA2A is perhaps the archetypal 'transparent' compressor, using a unique electro-optical gain-reduction stage and programme-dependent release times to achieve even heavy gain reduction with negligible side-effects.

Both compressors use a fixed-threshold system, so you control the amount of gain reduction by adjusting the input level. The *VS1176LN*'s Input control and the *VSLA2A*'s Peak Reduction control effectively add gain to the input signal, pushing it up against the fixed threshold and increasing the amount of compression. The Output and Gain controls respectively then adjust the output level. The elegance of this two-knob control system has given both units a reputation for being very quick to set up.

In addition to the main Input and Output controls, the *VS1176LN* has rotary controls for Attack and Release, calibrated simply from one to seven. The attack time can be adjusted over a 20-800µs range and the release time over 50-1100ms — as on the original unit, turning either of these controls clockwise shortens the respective time constant. The compression ratio is set using the four buttons on the left-hand side of the virtual VDU display, and you can also engage an 'all buttons' mode, which emulates the weird compression effect created when all four ratio buttons are jammed in at the same time — a common studio trick. Switches for three metering modes complete the facilities of the *VS1176LN*, and these let you meter gain reduction or output level.

Mike compares the sound of the *VS1176LN* to that of an original black-face Urei 1176 provided by FX Rentals.

The only processing option on the *VSLA2A* beyond its Gain and Peak Reduction controls is the Limit/Compress switch, which selects between two different ratio curves. The remaining switches at the right-hand side of the virtual front panel are for bypassing the processing and selecting the metering mode.

The first thing most people want to know about recreated compressors like these is how well they model the units they are based on. So I contacted FX Rentals who kindly sent over both an original black-face Urei 1176 and one of Universal Audio's hardware 1176LN recreations for comparison purposes. Lining these up against the *VS1176LN* demonstrated that the emulation is very faithful indeed, even when mimicking the distortion characteristics imposed by the faster limiting settings on bass and drums. I found the attack response between the units varied a little between the processors with matched settings in 'all buttons' mode, but at such extreme settings it's pretty tough to get two hardware units tracking closely, so it's hardly much of a criticism.

The bottom line for me is that the emulation has no right to be as good as it is when the VS8F3 and *VS1176LN* together cost less than a tenth of a pair of Universal Audio's hardware 1176LNs in the UK. But don't just take my word for it. Make your own mind up by listening to the comparison sound files I made during the review process — they can be found on the DVD with this month's magazine. I've also included some examples of the *VSLA2A*'s processing using the same source tracks.

Irrespective of questions of realism, the second thing VS users are likely to want to know is whether these two plug-ins are worth having over and above the dynamics processing already on hand in the multitracker. Having compared both processors to the VS2480's channel dynamics, there is no doubt that *VS1176LN* has more warmth and attitude, and that the *VSLA2A* is smoother and more transparent. Furthermore, when I set up the compressors by ear to be as close as possible, both plug-ins seem to provide greater subjective volume for a given peak level. In this comparison, I'd plump for the plug-ins every time. Even the new soft-knee compression algorithm on the VS8F3 only closes the gap slightly, and only really with the *VSLA2A* in my opinion. Overall, I think that it would be a rare VS-based studio indeed that would find the comparatively small investment in these plug-ins wasted.

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Stereo Reverb & Mastering Tool Kit

When coding the new *Stereo Reverb* algorithm, I take it that the Roland software developers had some processing bandwidth to spare after sorting out the main reverb block, so they added in a few extras. What I don't quite understand is why a pre-reverb compressor and expander topped their list of potential bonus features. Given the lack of EQ in the VS effect returns, I'd have thought a pre- or post-reverb equaliser would have been a much more useful choice, as in the original *Reverb* effect. And if you were going to put any dynamics process before a reverb, wouldn't something like a de-esser be a more sensible option? It's not that there are no uses for such a configuration: the compressor could be used to duck the reverb in the presence of the direct sound, or the expander could give the reverb an extra kick on the loudest notes. What's silly is that you'll have to sacrifice another of your effects slots if you want to tweak the tonality of the *Stereo Reverb* or de-ess its input — and chaining send effects is not exactly straightforward on some VS machines as it is. I'd rather that more useful processes were built in so that I only needed to sacrifice another effect slot to achieve the more unusual effects.

In terms of available parameters, the reverb block is almost identical to that in the VS8F2's *Reverb*, but with the same choice of reverb types provided in *Reverb 2*: two rooms, two halls, and a plate. A difference with the VS8F3 reverb, though, is that it has a stereo input, where the VS8F2 reverb sums its input to mono before processing. This means that the reverb will subtly reflect the stereo image of the input signal — an input panned to one side will produce a reverb return which favours that side. Both pre-reverb dynamics processes are the same as their counterparts in *Vocal Channel Strip*.

I checked out the new algorithm against the VS8F2's *Reverb* algorithm, and there was certainly a noticeable difference, even with the dynamics and equalisation of each of the algorithms switched out and the parameters matched as closely as possible. Completely removing the early reflections from both patches revealed the tail of *Stereo Reverb* to be thicker and less splashy, while isolating the early reflections of both algorithms demonstrated the smoother sound of the more recent coding in this department. Drums and vocals in particular seemed to work better with *Stereo Reverb*, but I also found that *Reverb* remained useful, despite losing out to its successor in terms of realism. A dense acoustic-guitar sound, for instance, was rather overwhelmed by the new reverb, whereas the sparser sound of the old one complemented it much more readily within the mix.

In terms of feature set, little has changed in the *Mastering Tool Kit* algorithm during its transition to the VS8F3 from the VS8F2, but the increased 56-bit resolution of the new card may now encourage more people to use its powerful array of processing.

The next test was to line up both VS algorithms against my own Lexicon MPX550. Starting from the Medium Room preset on all three processors, I tweaked the settings to try to reach some kind of sonic consensus. Switching between the three emphasised the thinness of *Reverb*, but also highlighted that the VS8F3's sound had more of the metallic overtones characteristic of budget reverb units than were present in the MPX550's returns. I also found that the Lexicon reverb seemed to sit better with the dry sound than did the Stereo Reverb — perhaps this difference could have been reduced had Roland dumped the dynamics blocks and thrown all the available processing into the reverb instead. That said, *Stereo Reverb* is still much smoother than *Reverb*, especially when processing transient sources or using patches which rely heavily on early reflections.

Finishing up the Roland plug-ins is *Mastering Tool Kit*, basically a souped-up version of the original VS8F2 algorithm of the same name. The enhancer block uses the new nicer design, so there are some sonic improvements, but basically you know what you're getting if you've used the VS8F2. The bottom line is that if you want to compare your demo with mastered tracks on pretty equal terms, then this plug-in will do the job. That said, I'd still be inclined to use *Mastering Tool Kit* only for processing individual tracks (where the matter-of-fact brutality of which its powerful processing is capable is more often an asset), leaving the mastering to someone with the monitoring system, ears, and experience of a mastering engineer.

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VS8F3 Verdict

Compared with a computer software plug-in, you might see a VS plug-in as a bit of a swizz; even with a VS2480 fully loaded with VS8F3 cards, you'll only get four *VS1176LN* plug-ins running. However, I think this argument is not that relevant to people who have chosen to use multitrackers, because they've already made the choice for hardware over software, despite the inevitable limitations in flexibility. For existing VS-series workstation owners, the ability to use top-quality brand-name plug-ins in almost exactly the same way they'd have previously used the built-in effects algorithms can only be seen as a wonderful new opportunity.

Even without the ability to load third-party plug-ins the VS8F3 already makes a solid investment, improving the processing fidelity and adding some nice modelled 'warmth' options. But when you add in the ability to use other manufacturers' processors within the VS environment, the card becomes pretty much a must-have for anyone wanting to upgrade their production sound. If the discussion of the *VS1176LN* and *VSLA2A* hasn't already whetted your appetite enough, then the prospect of forthcoming TC Electronic reverb, Massenburg EQ, T-Racks mastering processing, and Antares pitch-correction plug-ins should provide ample reason to get out your wallet. SOS

Thanks to FX Rentals (+44 (0)20 8746 2121) for supplying the comparison units used in this review. The daily rental for the hardware Universal Audio 1176LN in the UK is £47 including VAT.

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Reviews : Sound/Song Library

DFH Custom Vintage *****

VST+AU INSTRUMENT

Back in *SOS* March 2005 I reviewed *DFH Superior*, which is a 30GB sample-based drum sound module which went to extraordinary lengths to offer greater than expected dynamics, the ability to remix the mics on the original kit, and the option to choose what level of detail you wanted in the crosstalk between the drums. The sounds were superb. *DFH Custom & Vintage* uses the same engine as *Superior*, so if you want to see what the interface looks like and what facilities for adjustment it has to offer, simply check out the original review. If you already have *DFH Superior*, then the new sounds are added to your existing drum library, otherwise the program loads a new VST or AU plug-in. Like the original, *Custom & Vintage* provides around 32GB of drum sounds spread over five DVDs.

The package is based on old and often rare drums recorded using classic recording equipment (including an EMI TG desk) and techniques. Drums sampled include a Ludwig Black Beauty snare drum dating back to 1920 and a number of Noble & Cooley drums. During recording, the drums were screened to give a drier sound, but three ambience mics are employed to enable the user to add in more ambience if required. There's also an extra mono track processed via a Helios F760 compressor to get that classic big rock sound.

The main kits are Noble & Cooley Star, Slingerland Studio King, Camco Oaklawn, and a selection of snare drums including a Slingerland Radio King, a Ludwig Black Beauty, and a Ludwig Supraphonic. There's also a Craviotto, a couple of alternate Noble & Cooley models, including their Star Classic, Alloy Classic, and Zildgian models, and a Canopus Zelkova. These are augmented by cymbals and hi-hats from Zildgian and Paiste, all recorded using classic gear.

Back in the early 60s, some of the 'classic' drum sounds were pretty bad, and there was a tendency to mix drums so far back in the mix that only the snare remained audible half the time. However, by the '70s we had some great records with huge drum sounds from the likes of John Bonham and Pink Floyd's Nick Mason. That seems to be the flavour the producers of *DFH Custom & Vintage* were after, and I have to say that the result is every bit as good as I expected it to be after hearing their original *DFH Superior*. The sound is warm, solid, well defined and, thanks to the user-adjustable miking arrangement, extremely malleable. In a word, excellent. *Paul White*

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ASL ****

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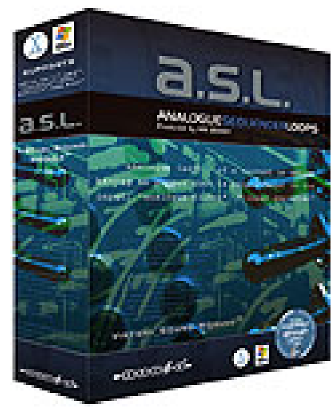

This is another in Zero G's series of libraries based on the *Intakt* software instrument — for details of how this works, check out the *Intakt* instruments review back in *SOS* February 2005. The musical contents of *ASL* (or *Analogue Sequencer Loops* to give it its full name) have been created by Ian Boddy and have been designed to complement Ian's *Kompakt*-based *Morphology* library, reviewed by Paul White back in *SOS* August 2004. The emphasis in *ASL* is on rhythmic, melodic, and bass loops — although the sound sources are, again, all derived from classic analogue synths and sequencers recorded at 24-bit resolution.

In all, around 1000 loops, mostly running to four bars, are provided within the 1.5GB library. The loops themselves are divided into seven groups: Analogue Drums, Bass Sequences, Electronic Percussion, FX Loops, Hi-hats & Noise Loops, Melodic Sequences, and Sine & Pure Sequences. Within each group, a further split into seven tempo subgroups (90-150bpm) is provided, although this is for convenience only, as the player software provides excellent tempo-matching facilities. All the loops use *Intakt*'s Beat Machine mode, which offers *Recycle*-style beat slicing. Usefully, each loop can be loaded in three forms. Firstly a 'sliced' version is provided with the full loop mapped to C1 and individual slices mapped from F1 upwards; secondly, each loop is provided in a 'pitched' format, with the original loop mapped to C3 and pitch-shifted versions provided on keys an octave either side of this; and finally, for the contents of each single-tempo subgroup, all the loops are provided in a single patch with each loop mapped to a different key — this makes it easier to mix and match loops on the fly.

The contents of many of the loop groups are pretty much as expected given their titles. For example, the Analogue Drums and Electronic Percussion groups are based firmly around rhythmic loops, while the Bass Sequences provide — well, some sequences played via bass sounds! Perhaps slightly more surprising is that the Melodic Sequences and Sine & Pure Sequences groups are also fairly rhythmic in nature. In the main, these are not really straight melodic parts in a traditional sense, but are melodic in that many are based on chord arpeggios. There is some excellent stuff amongst this lot and, with a suitable pad or texture sat underneath (for example from *Morphology*), a pretty full mix could be created from just two or three *ASL* loops. What is noticeable about all the loop groups is a consistently high audio quality and the genuine analogue character of the sounds. Many of the loops have been processed, whether through careful use of various filters, panning, or a combination of both, and the end results are full of movement and warmth. Interestingly, the vast majority of this processing has been done at source — *Intakt*'s own Modulation and Effects options are not extensively used.

In terms of musical styles, I could imagine these loops working in a number of contexts ranging from synth-led Depeche Mode or Soft Cell through club styles (ambient trance anyone?) to a Mike Oldfield/*Tubular Bells* sound. Blended with something like *Morphology*'s darker soundscapes, it would also be possible to create some very unsettling moods ideal for scoring under horror scenes. For fans of a genuine analogue sound, *ASL* ought to have considerable appeal and, given the UK price, the library offers respectable value for money, even if it doesn't quite shout 'bargain!' However, as with the other *Intakt*/*Kompakt* libraries that *SOS* has looked at over recent months, you are locked into the Native Instruments front end. Good though *Intakt* is, this may deter some purchasers who would like the content, but perhaps prefer to use it via an alternative looping or beat-slicing tool. *John Walden*

£ *Intakt* Instrument (including VST, DXi, Audio Units, RTAS, and stand-alone versions), £129.99 including VAT.

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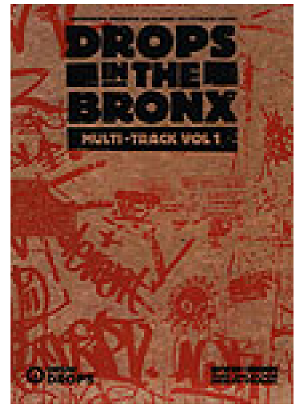
Drops In The Bronx Volume 1 *****

MULTI-FORMAT

This is another library from UK-based drum specialists Drumdrops — their *Fistful Of Drummers* title got the thumbs up in the *SOS* April 2005. As with other titles in their range, this one is available in two formats: either 16-bit stereo loops or, at a slightly higher price, 24-bit multitrack performances. The drumming itself is provided by Jan Kincaid from the Brand New Heavies and, as you might therefore expect, the playing is top notch. The recording approach is an all-



analog one, with vintage mics set up around well-aged Ludwig and Gretsch kits, and recordings carried out using a 1970s Neve console and some classy EQ and compression. The result is a sound that is full of character, and while it might not suit those after a highly produced modern sound, for a retro 'old school' vibe it is spot on.

The standard collection contains about 900MB of loops. This includes some 400 full-kit loops plus a further 400 percussion loops and a good collection of single hits. The loops are organised by original tempo (70-130bpm), although these can, of course, be adjusted given suitable tempo-matching software. The multitrack version is somewhat different in format to that of *Fistful Of Drummers*. While the same multitrack recordings are provided, these are also based around two-bar to four-bar loops, rather than a full track-length performance. I actually preferred this loop-based format, as it is easier to construct your own arrangement from these shorter building blocks. The multitrack loops are well organised and the naming conventions make it very easy to work out which snare loop goes with which kick loop, and so forth. This is important given that there is, of course, natural leakage between the various mics in the multi-mic recording, so although it is possible to mix and match loops, it does leave some 'ghost' hits. The overhead tracks are excellent and allow complete control over how much of the original room ambience is blended into the drum mix. Some of the performances also include an additional track featuring spring reverb applied to the snare, and blending in a little of this can add to the vintage sound.

As might be expected given the title of the library and the person behind the kit, these loops are pitched mainly at hip-hop producers. Nevertheless, there is plenty here that would work in a soul or R&B context and, by giving the tempos a small tweak upwards, many could easily sit behind a funk or rock track — provided that it required a good groove rather than a straight-ahead thrash. There is also a good mix of less busy loops and breakbeats. While the stereo loop set is perhaps not the cheapest collection you might buy, given the excellent playing and classy vintage sound, it certainly comes highly recommended. However, if you are prepared to work a little harder to create your own drum mix, the multitrack version is certainly the way to go. For those wanting a convincing analogue vibe to their drum tracks, Drumdrops are hard to beat. *John Walden*

£ 16-bit Apple Loops, REX 2, and WAV DVD-ROM, £75; 24-bit multitrack AIFF 3-DVD-ROM set, £99. Prices include VAT.

T Drumdrops +44(0)1273 553106.

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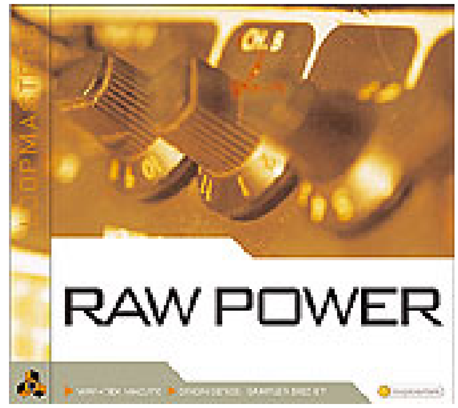
Raw Power ****

MULTI-FORMAT



This title from the Loopmasters Origin series comprises a total of 650MB of sample data in various different sample formats. All the original recordings are in the 120-150bpm range and are presented at 16-bit/44.1kHz resolution. The CD liner notes suggest the contents are suitable for anything from rock to indie, blues, goth, hip-hop, or metal. Having auditioned the loops, I'm less convinced about the last two genres. While there are some loops that might fit those styles, *Raw Power* is not the most obvious collection for fans of Eminem or Linkin Park. However, I would add punk (perhaps with a nostalgic '70s edge) and grunge to this list.

The loops are split into five main folders: New York Underground, Alternative USA, England's Dreaming, Artskool, and New Wave, with subfolders for drum (patterns and fills), guitar, bass, and synth loops. Within each main folder, the loops are all presented at a single tempo, and the collections within each folder contain loops that would obviously work well together in terms of style. Amongst the bass and guitar loops, there were some obvious groups, with four or five related loops that contained similar phrases and/or styles of playing, and this made it easily to build a coherent musical section with enough variability to sound 'real' rather than obviously 'looped'. It was perhaps a shame that this process wasn't more fully implemented within the drum loops — while these certainly contain plenty of excellent material, both in terms of performance and recording quality, there is less in the way of related loops. That said, it is not too difficult to build the basics of a complete drum track by mixing and matching within and between the various folders. If you need lots of flexibility here, you're likely to find yourself reaching for your favourite beat-slicing software.

In piecing together some brief demos, I found myself thinking of a number of specific bands. These would include some of the 'inspirations' listed on the liner notes — the Sex Pistols, Blondie, and Nirvana — but also other bands such as The Strokes and Queens Of The Stone Age. Much of this was down to the guitar and bass loops — while these covered all sorts of styles and sounds (clean and overdriven), the playing was grungy and not too technical. This is a compliment not a criticism — they're just right for that punk/indie/grunge vibe.

If there is a minor downside to the collection, it is that perhaps the library tries to cover a little too much musical ground in a single CD — other 'inspirations' listed on the liner notes include Beck, Oasis, The Pixies, Blur, Muse, Radiohead, The Beatles, and Jimi Hendrix! Even with more than 900 WAV loops, this is a pretty diverse musical palette to cater for. Still, if you like your rock to have character and a rough edge or two, *Raw Power* represents good value for money. *John Walden*

£ WAV, REX 2, and *Reason* CD-ROM, £39.95 including VAT.

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W www.timespace.com
W www.loopmasters.com
Published in SOS August 2005

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SOUND ON SOUND
The World's Best Music Recording Magazine

Studio Essentials

Accessories & Books

Published in SOS August 2005

Print article : Close window

Reviews : Accessory

ART Cleanbox II Transformer Isolator

Paul White

There are some ground-loop problems that are very difficult to cure, and sometimes the age-old solution of using an isolating transformer to ensure total electrical separation between the source and destination is the simplest and most effective solution. The ART Cleanbox II is a simple and affordable stereo isolation box containing two isolation transformers designed for line-level use, with quarter-inch jacks in and out. Both the inputs and outputs are unbalanced, but are wired to accommodate balanced or unbalanced connections on TRS jacks. Because transformers are passive devices, and because no other active circuitry is needed, there is no need for batteries or a power supply, but there is a small insertion loss of three or four decibels in typical applications.

I had the perfect opportunity to test this device in my own studio setup, where I use a small analogue submixer to combine the outputs from my various synths before feeding them to a spare stereo input on my audio interface. No matter what I did, there was always more hum on the mixer output than I felt there should be, even when using balanced leads. I suspected that metal-to-metal contact via the rack was causing multiple ground paths, but putting the ART Cleanbox II between the mixer and audio interface produced the best result I've had to date. There's still some hum from the modules themselves, but at least this goes away when the mixer gain channels are muted, suggesting that any remaining hum originates from within the modules themselves. Prior to fitting the ART Cleanbox II, there was always a noticeable amount of hum at the mixer output, even with the channels muted or the inputs unplugged.

After fitting the device, the level drop was inconsequential and there was no subjective difference in the sound quality of my synths, not even at the low end where the effects of low-cost transformers are often most evident. These claim to work from 10Hz to 50kHz (-5dB) and the quoted distortion is low, even at fairly hot signal levels. There are many studio applications for this little isolator, and though you should only resort to adding gadgets like this when all conventional approaches have failed, I have to say that this does the trick very nicely and for very little money in the UK. And of course I'm going to have to buy this one because I don't want my hum back!

information

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- W www.artproaudio.com

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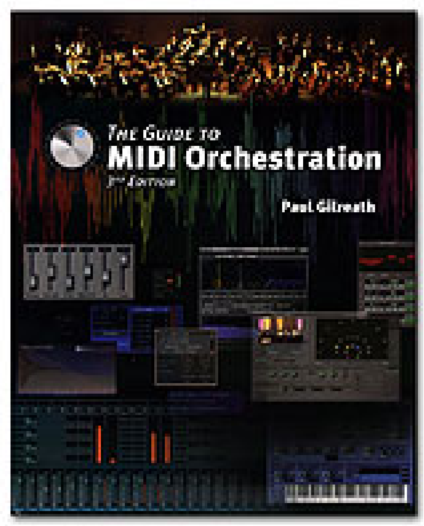
Book Review: *The Guide To MIDI Orchestration*

John Walden



With orchestral sample libraries getting ever more sophisticated, composers now have excellent tools for putting together convincing sample-based orchestral arrangements. These tools have many obvious applications — from the teaching of orchestration through to creating demos of compositions during the writing process. While a MIDI orchestration is never likely to better the results from a well-recorded live orchestral performance, budgets for TV and film music are becoming so tight that a sample-based orchestral score is sometimes all that is feasible. Although there are a good number of formal text books on the subject of writing and arranging for an orchestra, Paul Gilreath's *The Guide To MIDI Orchestration* approaches this subject in a fashion that is not only very accessible, but also more likely to appeal to the average SOS reader, insomuch as he discusses the hardware, software, and sample libraries required to do the job well.

The book is massive — it runs to over 700 pages, split into some 25 chapters — and the materials fall into three main sections. Chapters one to seven provide an introduction to the orchestra and deal with each of the major sections in turn. For those with no formal background or training in orchestral music, this offers a simple introduction and helps put much of the later material into context. For those wanting to learn how to arrange for the various instrument sections within the orchestra, chapters eight to 13 form a second section and provide sequencing examples to illustrate the most common ways to use, and write for, each major section. For example, chapter nine discusses how to write convincingly for the various elements of the string section, including how to build chord structures.

The final section — chapters 14 to 25 — is perhaps more familiar *SOS* territory. It covers the various equipment (hardware and software) required, the use of reverb, software samplers, and effects plug-ins amongst other things. This section also includes a comprehensive round-up of the current crop of orchestral sample libraries although, of course, this material will have a tendency to go out of date a little faster than the regularly updated coverage provided by *SOS*! This said, it is good to see direct comparisons of the major products all in one place. Also tucked into this final group of chapters is a small collection of interviews, mainly with composers and sample-library developers (the latter including Eric Persing and Gary Garritan).

Throughout, this book is beautifully presented, and Paul Gilreath writes with an easy, amusing, and informative style. It ought to appeal to those with a knowledge of music technology who also want an accessible introduction to the craft of orchestral composition — there is enough theoretical groundwork laid down here to get people started. Equally, those with a formal training in orchestral work wanting to learn more about the technology are also well catered for, and plenty of the advice will apply whatever brand of software sequencer you might be using. Overall, this book offers an easy route into MIDI-based orchestral composition, and at just over £30 in the UK it is also very good value for money.

information

£ *The Guide To MIDI Orchestration* by Paul Gilreath (ISBN 0964670534), £32.76 including VAT.

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SOUND ON SOUND
The World's Best Music Recording Magazine

Tannoy Precision 6 £399

pros

- Very detailed sound.
- Exceptionally well engineered.
- Wide sweet spot.

cons

- The Precision 6 may not have enough real low end for some tastes, in which case the Precision 8 might be a better choice. Alternatively, you could add a subwoofer to the system.

summary

Every speaker is a compromise, but Tannoy have got a lot right in the Precision series without it costing the earth. The low end lacks a degree of depth, but then this is the smallest monitor in the series.

information

£ Precision 6, £399 per pair; Precision 8, £499 per pair. Prices include VAT.

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Tannoy Precision 6

Studio Monitor

Published in SOS August 2005

Print article : Close window

Reviews : Monitors

Tannoy's new Precision monitors bring their celebrated dual-concentric driver and Supertweeter technology to the project-studio market.

Paul White

Photos: Mark Ewing

The Precision series sees Tannoy reverting to the dual-concentric designs they're well known for. However, seemingly in contradiction to this, there is still a separate tweeter, but in this design it is a so-called Supertweeter that extends the upper frequency range from where the centre horn of the dual-concentric driver leaves off, pushing the frequency response well beyond that which accepted wisdom would have us believe is adequate for reproducing music to humans. In this range of speakers, the response doesn't tail off until after 50kHz, but Tannoy claim that listening tests confirm that this does make a subjective difference when listening and also ensures greater transient accuracy. Two cabinet and driver sizes are available in the range, with the Precision 6 being reviewed here being the smallest. Where a little more level and low-end extension is required, the Precision 8 should fit the bill.

In designing this range, Tannoy's engineers made extensive use of advanced measuring equipment (laser scanning interferometry, acoustic CAD simulation, and so on) as well as conducting numerous listening-panel tests to see if adding those extra-high frequencies really made a difference. Their aim with the Precision, as with any good speaker, was to deliver wide bandwidth, low distortion, smooth frequency response, accurate phase control, and high sensitivity. Certainly extending the high-end response should improve the phase accuracy at the top of the audible range, so that's probably the main reason for any subjective improvement.

Tannoy's unique selling point has always been their dual-concentric driver design, which has the clear advantage of presenting a point source across the frequency spectrum. The tweeter unit is neodymium powered and ferrofluid cooled, and is positioned on the back of the low-frequency driver so that the sound emerges in the centre of the woofer cone rather than from a separate location. A practical benefit of this is that you get a wider sweet spot than with conventional speakers, but there are a lot of design challenges to get a dual-concentric speaker sounding right. It is intriguing, therefore, that they have combined their dual-concentric approach with a separate Supertweeter. How well this works remains to be seen, because coincident positioning of the drivers becomes more important at shorter wavelengths (higher frequencies) and it's clearly not practical to put the Supertweeter at the centre of the main horn.

Today it seems there is an appetite for active speakers, but with passive models such as these you at least get a choice of which amplifiers to use. With a frequency response of 64Hz-51kHz and a sensitivity of 91dB/W at 1m (no maximum SPL is specified), the Precision 6s don't need a huge amount of amplifier power to generate plenty of sound, but in order to allow adequate headroom for the handling of transients, an amplifier of between 60W and 120W continuous-sine-wave rating is recommended. My tests used a rather nice 100W per channel AVI integrated amplifier, and you need a decent amplifier because, although the speakers have a nominal 8(omega) rating, they are actually closer to 6(omega) in practice.

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Construction

The Precision 6s are a departure from Tannoy's usual restrained cosmetic styling and feature an oval brushed-aluminium plate inset in the baffle. A nice grey textured finish is used for the baffle, which is also sculpted to avoid the 'box' look and to help reduce cabinet edge diffraction. The reflex port is located on the back panel, along with the chunky connection terminals (no bi-wiring options) and both the dual-concentric driver and the Supertweeter are set flush into the baffle and secured with cap-head bolts. The dual-concentric driver is a nominal six inches in diameter and uses a paper-pulp cone. Flux control rings are fitted to the LF driver section with eddy-current damping provided via copper pole caps.

At the centre is a one-inch titanium-dome tweeter feeding the horn and the coverage is specified at 90 degrees in both the vertical and horizontal planes. The Supertweeter is also one inch in diameter and uses a titanium diaphragm. In traditional Tannoy fashion, the cabinet is very solidly built and well braced with a massively thick (40mm) front baffle and proper tongue-and-groove joints securing the front and rear panels. Lossy coupling, (which I take to mean some kind of resilient gasket arrangement) is used to prevent speaker vibrations making their way into the cabinet structure through direct contact. The crossover frequency is quoted as being 2.5kHz, but there's no mention of a second crossover frequency between the dual-concentric horn and the Supertweeter, so it's not entirely clear what if any frequency division is employed between the two tweeters. Magnetic shielding is provided for the benefit of those users who haven't yet given their 'glass and copper' CRT monitors to the science museum, and the overall size is just 272 x 440 x 287mm. The sturdy construction is reflected in the weight of almost 10kg per speaker.

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Session Impressions

Correctly mounted and positioned, the Precision 6s deliver a fairly well-balanced sound with plenty of transient detail, though for my taste the high end gets just a little too splashy and forward sounding at higher listening levels. This isn't serious, but I also found the bottom end to be a little weak at normal listening levels, with little real punch or depth. I expect the 8s to be better in this respect, but with today's technology it should be possible to get a reasonably satisfying and solid low end from smaller speakers without compromising too much on accuracy. Of course the method of mounting and the position within the room will also affect the low end performance, so if you have a room that normally produces an overblown bass sound, these monitors might suit it very well. If not, I'd go for the Precision 8s, which go down to 56Hz and are a couple of decibels more sensitive. Stereo imaging is good, and the dispersion characteristics of the speaker provide a usefully wide sweet spot. The sound can get a bit confused when the speakers are driven very hard, but at sensible nearfield monitoring volumes the level of clarity and detail is good.

I don't know how much influence the Supertweeter has on the subjective sound, as there's no way to turn it off, but to my ears, the Precision speakers have more of a conventional two-way character about them, while retaining the wide sweet spot and even dispersion of classic Tannoy dual-concentric designs. My observations regarding the forward sound of the titanium tweeter are personal, as I know many people like their tracking monitors to sound slightly larger than life, but to get a properly balanced low end in my studio I'd probably opt for the Precision 8s or add a subwoofer to the 6s. In other studios, they may well work fine as they are with some careful positioning.

I've never been a huge fan of dual-concentric speakers, but I didn't feel the Precision 6s had the somewhat recessed mid-range that I've noticed on some dual-concentric models. In this respect they probably combine some of the best characteristics of both design philosophies, which makes them well worth investigating for applications where you need a wider-than-typical sweet spot and a good tonal balance across the important mid-range and high-end frequencies. Of course, performance also has to be balanced against price, and the Precision 6s aren't unduly expensive in the UK, yet they are very well engineered. Compared with similarly priced passive monitors, the Precision 6s actually stand up very well and only the lack of low-end weight works against them, especially for those working on dance music where the low end is critical. In all, the Precision 6s turn in a very good performance within their size and price range — as long as they're teamed with a decent amplifier with adequate power to spare.

SOS

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SOUND ON SOUND
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In this article:

TFPro P4

Dual Recording Channel

Published in SOS August 2005

Print article : Close window

Reviews : Processor

In the red corner, weighing in at 0.8kg, is the latest addition to the TFPro range of outboard — a dual mic preamp with integrated dynamics processing.

Duncan Williams

Regular readers of *SOS* will be familiar with the distinctive muted red of Ted Fletcher's designs, and equally familiar with his greener designs from the early Joemeek processors. The neat little P4 is the latest addition to his own range, aimed squarely at the small-broadcast and DAW markets.

Thinking Outside The Box

The P4 comprises a pair of preamps with limiting or compression on each channel, and occupies half a 1U rack space. Arranged sensibly across the rear panel are two XLR microphone inputs, with their accompanying balanced line inputs and outputs at +4dBu, and paralleled phono outputs at -10dBV. The inclusion of phono outputs allows for easy connection to most consumer-level soundcards, giving straightforward integration of the unit within an existing DAW setup. The manual suggests that the paralleled outputs could be used for zero-latency monitoring, though this would require the preamp to be used in conjunction with a separate mixer. Unsurprisingly, given its petite stature, the unit is powered by a rather chunky external 12V AC power supply.

On the front of the unit are discrete controls to adjust the input level and the threshold of the dynamics stage, respectively. Each channel has switchable 48V phantom power (strangely enough this is linked with a -20dB pad) and a 75Hz high-pass filter, with an LED indicator for each. It is worth noting that, although the combination of phantom power and a pad on one switch is rather unusual, it does follow that condenser microphones will generally deliver higher input levels

TFPro P4 £299

pros

- Excellent-quality preamps for the price.
- Well-designed layout.
- Highly portable.
- Effective dynamic control.
- Easy to operate.
- Distinctive sound.

cons

- Simplistic metering.
- No power indicator light!
- Requires a mixer for monitoring.
- Chunky external power supply.

summary

A neat, user-friendly box which will appeal to those looking to upgrade the quality of their recording path at a reasonable cost. However, it is by no means the only option at this UK price point.

information

£ £299 including VAT.
T TFPro +44 (0)1803 316599.
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than dynamics, so should phantom power be required for a condenser, the corresponding drop in input gain will probably be necessary anyway. The high-pass filter is a nice touch for removing unwanted rumbles in the jungle, particularly when used in conjunction with shockmount-free dynamic mics. Finally, a simple four-stage LED display provides basic, but welcome, signal metering, and the dynamics stage of each channel is switchable between compression and limiting presets.

An interesting feature of the design is the P4's revised approach to limiting, which TFPro have christened 'asymmetrical' limiting, which they claim improves headroom when used in conjunction with digital recorders. Details of how this is achieved are not specified, but an audio demo can be found on the company's web site if you want to have a listen.

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On The Session

I tested the P4 on a voice-over session for a radio advert, with both Neumann U87 condenser and Shure SM57 dynamic microphones, recording into Emagic *Logic* v6 on an Apple Powerbook, via a small mixer for monitoring purposes. The first thing that struck me when powering up the P4 was that, owing to the omission of a power indicator light, if none of the front panel's switches are depressed it is impossible to tell whether the unit is actually switched on!

Nevertheless, having overcome this small hurdle, using the P4 was simplicity in itself. The built-in limiter relieves some of the inherent cautiousness one would usually apply when setting the input level on the preamp, and the deceptively simple appearance of the dynamics section belies the quality achieved at lower threshold settings, which was wonderfully transparent.

With the limiter driven hard the P4 is brutally effective, creating a substantial difference in the relative level of the vocal, although it was prone to distortion at high input and limiting levels with the U87, particularly on plosives. Due to the simplicity of the controls, however, tweaking the input and limiter settings is extremely quick, and the P4 quickly yielded a consistent, healthy level into the recorder with few compromises. In fact, what the dynamics section lacks in terms of control, (with no attack, release, or make-up gain), it more than compensates for in user friendliness, and can certainly deliver the goods. With the clock ticking on a session, the elimination of unnecessary tweaking is a credit to Ted's 'designed for operation' ethic, especially with live broadcasting in mind, where fine-tuning the make-up gain of a vocal on the fly could be dangerous!

The quality of the mic preamps is excellent value for money, delivering a sound which, it has to be said, has a lot in common with Ted's earlier Joemeek designs such as the VC3 (reviewed in *SOS* June 1999), meaning the P4 would certainly be a wise choice for anyone with a small mixer (such as a Spirit Folio, Behringer Eurodesk, or Mackie VLZ), looking to improve upon their existing preamps. Indeed, given its backpack-friendly size and weight, it's a shame that there is no headphone output included, as this would allow for zero-latency monitoring without the need for a separate mixer — location recording with the P4 could be a breeze. Even so, with processing power ever on the up, this will be less of an issue for those with very low-latency Core Audio, ASIO, or WDM drivers running.

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P4 Pronouncements

The P4 is a quirky box with a unique character and an unusual feature set for a portable unit in this price bracket. However, in a market already saturated with preamps and voice channels, today's UK buyer is certainly spoilt for choice, with products like the M-Audio DMP3 at £199, through to the more expensive Focusrite Twin Trak Pro costing £399. The P4's lack of fashionable tubes, combined with the limited controls of the compressor/limiter, may also put off DAW users looking to add 'warmth' to their tracking — they might want to consider boxes like the Presonus Blue Tube DP at £165, or the Aphex 207 at £409. That said, this unit isn't strictly aimed at home or studio recording, and it is to its credit that it can adapt to almost any situation where a good, clean, microphone level is required. So if you're looking for a convenient dual preamp with intuitive dynamics processing, then the P4 is an excellent choice. SOS

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In this article:

Waves *L3* £400/£799

pros

- Easy to use, yet the *Multimaximizer* version offers lots of flexibility.
- *Multimaximizer* is, in effect, a mastering EQ as well as a limiter.
- Excellent interface design.
- Can be more transparent than wide-band limiting, but can also offer some interesting effects.
- Free to existing *Diamond* & *Platinum Bundle* owners who subscribe to the Waves Upgrade Plan.

cons

- Won't always give noticeably better results than *L2*, and when it does, the improvement is often incremental rather than radical.
- This sort of processor rarely makes music sound better!

summary

If you have to fight the loudness war, Waves' *L3* will give you an edge.

information

£ TDM version £799; native version £399.50. Prices include VAT.

T Sonic Distribution +44 (0) 1525 840400.

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Waves L3

Multi-band Mastering Limiter [Mac/Windows]

Published in SOS August 2005

Print article : Close window

Reviews : Processor

If you want your mixes to go to 11, Waves' new mastering limiter might be the weapon you need.

Sam Inglis

The *Multimaximizer* version of *L3* can act as a mastering equaliser as well as a multi-band limiter. Here, I've gently boosted the upper-mid frequencies and cut the low mids.

Sometimes new technology gives us new ways to screw things up. Take, for example, the digital mastering processors that appeared in the '90s. In the right hands, they could add that last bit of sparkle and polish that would make a finished mix sound like the work of a professional. In the wrong hands, however, they could turn a bad mix into an unlistenable master, eliminating any trace of dynamic variation. And when record company executives got their heads round the idea that these devices could be used to make their CDs louder than the competition, even the best mastering engineers found themselves under pressure to deliver flatlined mixes.

In this quest for ever-louder CDs, two of the most popular weapons have been Waves' *L1* and *L2* peak limiters, which the company call 'Ultramaximizers'. *L1* and *L2* have two basic aims in life. The first is to increase the subjective loudness of incoming digital audio, without introducing clipping or other obvious distortion. The second is to dither the final output from your digital audio workstation to 16-bit for CD mastering (see box).

The genius of *L1* and *L2* lies in their ease of use. There are only two major controls: Threshold sets a level below 0dB to which the input signal is limited, and automatically applies the same amount of make-up gain. Output Ceiling then www.waves.com

Test Spec

- Waves *L3* v1.0.
- Inta Audio Centrino laptop with 2.0GHz Pentium-M processor and 2GB RAM, running Windows XP Home Edition SP2.
- Tested with Digidesign *Pro Tools M-Powered* v6.8.

'scales' the results so that the output signal never exceeds the Ceiling value. The net gain in loudness is thus equal to the difference between the Threshold and Out Ceiling settings. Waves say that with most material, it should be possible to achieve at least a net increase of at least 6dB before distortion and other side-effects become obvious. I've sometimes found this claim a bit optimistic, but even so, there's no denying that *L1* and *L2* can achieve some impressive results. Equally, there's no denying that they can be pushed much, much too hard.

top

The Next Level

The hardware L2 and the software *L1* and *L2* have been around for quite a while now, and have been facing increasingly sophisticated competition. Waves' answer is *L3*, a clever multi-band implementation of the same concept. Like almost all Waves processors, it's available both as a TDM plug-in for *Pro Tools* (in which case you need to run it on an Accel card) and a multi-format native plug-in, in each case supporting sample rates up to 96kHz.

In *L3*, the incoming audio is divided into five frequency bands. Linear-phase crossovers are used to do the splitting, so unless you actually apply limiting, the output signal is identical to the input signal, albeit delayed by the limiter's lookahead value of 80ms. The main controls are still called Out Ceiling and Threshold, and they still apply globally: whatever you do to the individual bands, their combined level is limited to the Threshold value, before being scaled to hit the Ceiling.

The clever part of the design lies in the relationship between the peak detection and the attenuation. Conventional multi-band limiters split the signal into separate frequency bands and then limit each band independently. Here, however, the level in all five frequency bands is summed using a patent-pending algorithm Waves call the Peak Limiting Mixer. If, at a given instant, this sum exceeds the Threshold value, *L3* works out the amount of attenuation that is needed and intelligently distributes it across the different frequency bands. By default, the bands that are attenuated the most are those that contain the most energy, but it's also possible to instruct *L3* to concentrate the gain reduction in frequency bands where you think it will be less noticeable. The result, at least in theory, is that you can apply more overall limiting with fewer audible consequences. Waves say that we can expect their PLM technology to be put to different uses in future products.

The more basic *L3 Ultramaximizer* lacks the detailed control offered by its bigger brother.

There are two versions: the simplified *L3 Ultramaximizer*, which is now included in both the *Platinum* and *Diamond Bundles*, and the full-blown *L3 Multimaximizer*, which is a *Diamond*-only affair. (Buying *L3* separately gets you both versions.) *Ultramaximizer* won't frighten off anyone who's used to the simplicity of *L1* and *L2*; apart from the Threshold and Out Ceiling sliders and the dither settings, the only user controls are a pop-up list of preset Profiles and a single Release control. *Multimaximizer*, by contrast, adds a graphical display and a set of controls for each of the five frequency bands, along with a pop-up list of Master Release characteristics.

Even by Waves' high standards, the ergonomics of *Multimaximizer*'s interface are impressive. Despite the number of parameters on display and the unfamiliar nature of settings like Priority and Separation, it's easy to take everything in at a glance, and nearly all the parameter changes you might want to make can be achieved using a single mouse movement in the graphical display.

The display itself is a model of elegance and a very useful analytical tool for fine-tuning your settings. It is animated to show the gain reduction as it is applied across the frequency spectrum. An orange line follows the gain reduction in real time, but with most material, this moves too fast to give you a clear idea of what's going on, so Waves have added a 'smoothed trail'; this is a blue shaded area that follows the orange line, but decays much more slowly, so you can see where gain reduction has been applied. There's also a dark red peak hold line, which shows the maximum level of gain reduction that was applied at every point in the frequency spectrum.

Superimposed on these traces are visual representations of the band-specific controls. Each band features a coloured diamond which can be moved in both vertical and horizontal planes. Horizontal movement adjusts the crossover frequency between a band and its immediate neighbour, whilst vertical movement adjusts both the Priority and Gain of that band.

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All In A Dither

Both the *Ultramaximizer* and *Multimaximizer* versions of *L3* include dithering features that are exactly the same as those in *L2*, except that the options are chosen from pop-up menus rather than by clicking to cycle through the choices. You can set *L3*'s output resolution to 24, 22, 20, 18 or 16 bits, choose either no dithering or two types of Waves' proprietary IDR algorithm, and select the amount of noise-shaping that the dither noise should have. It's all exactly as straightforward as it sounds.

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Priorité À Droit

The concept of Priority is at the heart of *L3*'s design, and it's these controls that allow you to shape the psychoacoustic element of the process. By increasing the

Priority of a band above zero, you're telling *L3* that the material in that frequency range is important, so when it encounters a signal that exceeds the Threshold, it will apply proportionately more gain reduction in the other bands and less in that band. Conversely, if you give a band a negative Priority value, *L3* will take the opportunity to dump proportionately more of that band's content when it needs to achieve a certain overall gain reduction. The Gain controls are more conventional, doing exactly what you'd expect — increasing or decreasing the level of each frequency band in a fixed rather than a dynamic fashion, allowing the *Multimaximizer* version of *L3* to act as a five-band, linear-phase equaliser as well as a peak limiter. Moving the coloured diamonds in the graphical display, or the double-arrow controls in the pane beneath it, adjusts the Gain and Priority together.

The Separation control allows you to vary the behaviour of the peak detection algorithm. At the default of 100, it operates as a fully multi-band device, where each band's peak-detection sensor receives only the signal from that band. As you lower the Separation, each band's detector begins to receive signal from the other bands as well, and at a value of zero, each band's detector receives the full-bandwidth signal. This makes *L3* behave more like its single-band predecessors, although you can still adjust the Priority settings in order to distribute the gain reduction unequally across the different bands.

Each of the five frequency bands also has its own Release time parameter, which is a big help in achieving more transparent results. Typically, you might want a slower release time for bass frequencies to prevent distortion, with a tighter response higher up the spectrum to allow *L3* to react more naturally to transients such as snare hits. You can manually set the release time for each band, but for most material, you're better off choosing one of the four programme-dependent settings from the Master Release pop-up. These all scale the release times across the frequency spectrum, usually with a slower release at the bass end, but also vary them to a certain extent in response to the programme material. Of these, ARC is similar to the auto-release algorithm used in *L1* and *L2*; Warm is more programme-dependent at the lower end and stays closer to the manual settings in the higher bands, while Scaled does the opposite and Aggressive provides the tightest of the four release characteristics. The choice of Master Release algorithm makes more difference to the sound of the results than almost any other single parameter in *L3*. For rock and pop music, I usually preferred the ARC or Aggressive settings, while softer material often suited the Warm preset.

Another feature that's intended to help fine-tune the settings is the ability to solo individual bands. This allows you to check which instruments in your mix fall into which frequency bands, which can be handy.

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System Requirements

Waves recommend the following minimum spec for any machine running their version 5 plug-ins (which includes both versions of *L3*):

- Mac: 867MHz G4 or better, 512MB RAM, Mac OS 10.2.8 or newer, 800 x 600-pixel display.
- Windows: 1GHz Pentium III, Pentium 4 or Athlon, 256MB RAM, Windows XP SP1 or 2000 SP4, 800 x 600-pixel display.

Whilst you can get away with running some of the simpler Waves plug-ins on lesser machines, *L3* puts obvious demands on the CPU and I wouldn't want to run it on any computer that was showing its age.

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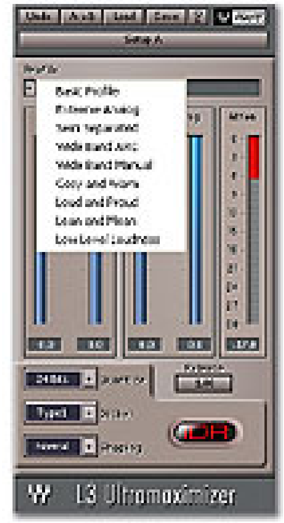
From The Get Go

Despite the welter of new controls, getting started with *L3* is just as easy as it was with *L1* and *L2*. You set the Out Ceiling control just below 0dB (allowing your signal to peak at 0dB risks the possibility of inter-sample peaks exceeding 0dB, which will clip when cheap D-A converters attempt to reconstitute your music) and bring down the Threshold control until you start to hear audible side-effects. At that point, you back off the Threshold a little, before getting into more detailed editing if you feel the need. One of the great things about all the L-series processors is that it's easy to A/B the processed and unprocessed signals at the same volume; you simply bring the Out Ceiling slider down to the same level as the Threshold slider, whereupon the perceived loudness should be the same as with *L3* bypassed. (This would be even easier if Waves added a dedicated button.)

I suspect that many users will be perfectly happy with the simpler *L3 Ultramaximizer*, but those who take their mastering seriously will find the full *Multimaximizer* does things that can't easily achieve in other ways, and will be particularly useful in tackling problematic material. For instance, suppose you have a track that's based around a very solid bass sound, and you find that conventional wide-band limiting is causing the low end to pump in an undesirable way. This is no problem for *Multimaximizer*: simply set the range of the low or low-mid band to encompass the bass, and whack its Priority up. Now, when the peak detector encounters a big snare transient or guitar chord, it'll leave the bass alone and achieve the necessary limiting by cutting the other bands.

Likewise, wideband limiting sometimes results in a situation where slow release times leave obvious 'holes' after snare hits, but a faster release causes the bass to distort. *L3*'s adaptive release algorithms allow you to use an aggressive fast release in the mid frequencies and be more gentle on the bass. *L3*'s flexible design also allows you to process your mixes in ways that are more characterful than transparent. The available Profiles include settings such as 'Extreme Analog', which uses fixed rather than adaptive release times to create a deliberate pumping effect that could well suit some types of music.



Although you can't edit the characteristics of each band in *L3 Ultramaximizer*, you can choose from a selection of preset Profiles.

In general, I think it's reasonable to say that compared to its predecessors, *L3* will often give you either an extra couple of dBs of gain reduction for the same perceived level of distortion, or the same amount of gain reduction with somewhat fewer side-effects. However, the loudness race has got to a point where there really isn't all that much room left for manouvre. For example, where snare drums are prominent in a mix, I've always found that you can't push *L1* or *L2* all that far before these start to acquire a papery, thin quality, and that this tends to be the factor that limits the amount of the process you can apply. I was interested to see whether *L3* would make it possible to achieve more gain reduction whilst retaining a solid snare sound, but I had only limited success. Turning up the Priority in the mid and high-mid bands did seem to give the snare more body, but only at the expense of obvious pumping in the bass and low-mid bands. I didn't find this any more pleasant than the papery snares I got with full-bandwidth limiting, but I suppose at least it's a different compromise, which might be more appropriate in some situations.

One of my other test tracks was a laid-back folk tune where the scope for wide-band limiting was restricted by the prominent female vocal, which had to be treated carefully to prevent it distorting very obviously on loud consonants. Again, dumping some of the gain reduction into the bass and low mids helped somewhat, but the results were still less than perfect.

I had a lot more success with rocky tunes containing plenty of distorted guitars, perhaps because these were masking some of the adverse effects. Even where the tracks were pretty heavily squashed already, *L3* squeezed another 3 or 4 dB of gain out of the mix without making it noticeably worse, and some interesting pumping effects could be achieved. Where the test tracks featured contrasting sections, such as a sparse verse followed by a busier chorus, it was invariably helpful to use different settings in different areas, but then this is true of many mastering processes.

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Conclusions

However much we might bemoan the self-defeating quest for ever-louder CDs, it's now a fact of life, and few of us can afford to put out discs that are 12dB quieter than everyone else's. In that context, Waves' *L3* is an impressive piece of software design. It allows you to achieve a bigger boost in levels with fewer adverse side-effects, it's powerful yet easy to use, and the *Multimaximizer* version has the flexibility to tackle mixes that would be a headache for other processors. The lack of 192kHz support might be an obstacle for those working with high-resolution playback formats, but I would hope that audiophile material wouldn't be treated to this kind of processing in any case. Price-wise, it's far from an impulse buy, but compares well to rival products such as TC's *MD3 Mastering Tools*, and owners of the *Platinum* or *Diamond Bundles* who subscribe to Waves' Upgrade Plan should be able to get it at no extra cost. Now, if I can only squeeze another 2dB out of my track, it's sure to be number one by the time you read this... SOS

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Yellow Tools MVIs

pros

- Simple to use.
- Fabulous-sounding libraries of high-quality samples.
- Very playable, with just enough tweakability.

cons

- Can take a while to load samples.
- Key-based security a pain, in my experience.
- Require a lot of hard disk space.

summary

Huge collections — mind-bogglingly so in the case of *Majestic* — of well-recorded, thoughtfully organised, playable samples. There may be only subtle differences between some of the sample sets, especially considering the size of the collections, but there's no denying the quality.

information

£ Full versions, £279.99 each; Loyalty Editions, £199.99 each. Prices include VAT.

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Yellow Tools MVI

Virtual Instruments [PC/Mac]

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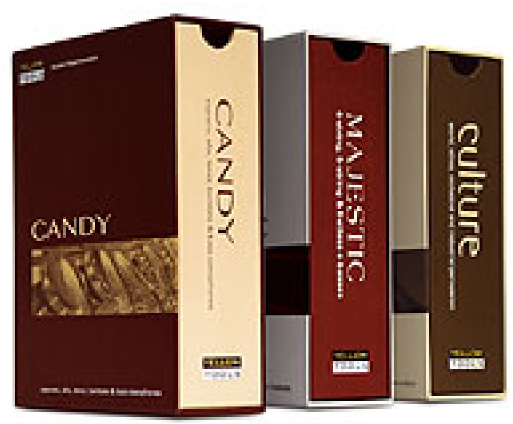
Reviews : Software

German company Yellow Tools started off as sample-library developers, but like many others, they've moved into the world of virtual instruments with the lovingly crafted MVI range. We check out all three...

Derek Johnson



Many of you will already know the name Yellow Tools — or more correctly, you'll probably be familiar with the German company's *Pure* series of sample CDs (released under the Best Service banner).

Pure Drums, *Pure E Basses 1 & 2* and *Pure Guitars* (the last reviewed in *SOS* August 2001) have been gaining a positive response since shortly after the company's inception in 1999. In fact, the libraries have been re-issued in single-set, multi-format 'New Editions', on DVD-ROM. Ultimately, though, Yellow Tools' development team found working for existing samplers restricting, and these days, the logical step for sample CD producers is to move into the development of dedicated sample-playback platforms. And this is how Yellow Tools' Modular Virtual Instruments — or MVIs — came about.

The first of these was *Culture*, a large, varied percussion instrument released in 2001. A couple of years later, the bass instrument *Majestic* followed, and more recently there's been *Candy*, which deals in saxophones. We'll be having a look at all three instruments in this review.

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www.yellowtools.de

Test Spec

- 3.06GHz Pentium 4 PC with 1.25GB of RAM running Windows XP.
- Steinberg *Cubase SX* v2.2.0.33, Mackie *Tracktion* v1.5.1.16.
- Yellow Tools MVI versions reviewed: *Culture* v1.7, *Majestic* v1.7, *Candy* v1.0.

Eye On The MVI

Before I get on to talking about the libraries, I should cover the structure of each MVI. First of all, the engine has been designed for efficiency and to have as small a load as possible on the host software. So it is that MVIs will run on comparatively venerable machines, according to Yellow Tools — the minimum system requirements are 400MHz G3 Macs or 400MHz Pentium, Celeron or Athlon PCs, with just 256MB of RAM (although more is definitely a better idea, as there are a lot of samples in each MVI). The instruments are supplied on DVD, so you will need a DVD drive and lots of free space for each MVI you plan to load — 16GB in the case of the *Majestic* bass. You also need a free USB port for the key-based Yellow Tools copy protection, of which more in the box over the page.

Part of the reason for the relatively light load the MVIs impose on their host computers can be understood by having a closer look at the screenshots in this article. You won't see any DSP-heavy additions such as resonant filters or built-in effects in any of the instruments, and this approach in itself almost ensures low CPU load. I wouldn't have minded a filter or two to let me get a little more creative with the *Culture* percussion set, but Yellow Tools obviously wish to let their samples speak for themselves. The MVIs' parameters are focused mainly on performance issues, with just a handful of subtle sonic-tweaking tools. Similarly, Yellow Tools have ensured that as many potential hosts as possible can accommodate MVIs. Practically anything running on PCs or Macs (even under OS 9) will do, whether it's a VST-, RTAS-, Audio Units- or DXi-compatible host — and the instruments can even be used in stand-alone mode.

Culture here being hosted by Steinberg *Cubase*; the pattern in the background comes from Yellow Tools' extra *Culture Groove Pack* (see the box on page 141 for more on this).

As already indicated, the MVIs potentially handle a lot of high-quality samples: the library is supplied as 24-bit/44.1kHz audio files, and up to 16 velocity-switched samples are used per note with some of the multisamples. What's more, disk streaming isn't used here, so you might find the load times of complex material to be rather long. On the plus side, if you want to load a sample set twice, the second load is instant, since it refers to the samples already loaded into RAM. And you can save on hard disk space by copying individual instrument sample sets, as long as you remain aware that some preset material won't be available if the necessary samples are missing.

It seems a small issue, but what passes for patch information is quite independent of the central sample collection of each instrument. This data is very compact and can be easily transferred between platforms. The sample collection for each MVI is fixed, so that every user of a given Yellow Tools plug-in has exactly the same collection, and no incompatibility issues can arise. The opposite side of this coin is that the sample collections are not expandable, least of all by the user. This isn't such a problem, though, since users won't be buying these MVIs as generic sample-manipulation tools.

The 'modular' bit of the MVI engine's name sounds good, but I couldn't see how the three instruments were supposed to fit together in any way at first, or how they could fairly be regarded as modular parts of, well, anything very much. That, though, was before I heard about Yellow Tools' new product *Independence*, which was announced at this year's Frankfurt Musikmesse, and was about to be released as I was finishing this review (perhaps predictably, Yellow Tools were planning its release day for July 4th!). It would seem that just as instruments based on libraries in Native Instruments' *Kontakt* format can be dissected down to sample level and made fully editable if you have the full version of *Kontakt*, so Yellow Tools' MVIs will be fully accessible and editable within *Independence*. As it's still unreleased at the time of writing, though, a review of *Independence* will have to wait for another issue of *SOS*! There's more on the new software in this month's News section at the front of the magazine. Even without *Independence*, though, the MVI engine does offer the user some opportunities for custom sound creation within the context of each plug-in's sample set.

Each plug-in looks largely similar, barring overall colour, and has all its functionality in one window, with no hidden operating levels. That functionality is similar across all three MVIs, with only minor differences.

The top level of operation is a Multi, which consists of up to eight Layers; a Layer would be what in synth-like terms we might call a single patch, and is a complete set of samples that replicate an instrument. Of course, that collection might be a couple of claps in *Culture* or three octaves of individual bass notes (with eight levels of velocity per note) in *Majestic*! Layers can also have a suite of audio, mix and control parameters applied to them.

The Keygroup is another level in the Yellow Tools sample hierarchy and consists of the sample (or set of velocity-switched samples) assigned to one note. The playback of each Keygroup can also be customised. However, as it stands, it's not possible to break down a Layer/instrument any further: the sample level is in the MVI somewhere, but you can't access it.

The Samples in each MVI tend to be unlooped, and Yellow Tools' minimum quality standard of at least one sample per note ensures that no artefacts will be introduced by transposing samples (though pitch-bend can be used, and Keygroups transposed if you wish). Looping is used rather a lot on *Majestic*'s basses. Sometimes, though, turning the looping off on bass Keygroups reveals natural endings to notes, but this is not always the case.

Where there is no looping, you'll only ever be able to hold a note for as long as the sample itself. In these cases, the sample length is dictated by the musical and physical context. The samples are as long as a lung-full of air allows in the case of *Candy*'s saxes, or as long as it takes for a struck or plucked note to naturally decay to complete silence. Naturally, this means that a 'lung-full' will produce shorter notes at the low end of the range of a bass sax than in the middle range of a soprano, and that notes sampled with different dynamics will have different lengths.

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The MVI Basic Editor

The MVI window is divided into two editors: Basic and Pro. The Basic (upper) part of the window lets you load and save Multis and Layers, and edit and customise Layer playback, one Layer at a time. Your eye will initially be drawn to the Layer display itself. If you instinctively try to click on a Layer name or space in the Layer display itself, or access editing for that Layer, nothing happens: you need to click on a button in the 'Select' column. Layers can be soloed, and each Layer's MIDI input and/or audio output can be muted.

So, you've loaded a Layer and you'd like to tweak it. What next? The rest of this section is sub-divided into smaller chunks covering parameters relating to audio, key range, velocity range, MIDI response, and polyphony. The audio section is essentially one channel of mixer: all three MVIs have level, pan and output routing controls. Eight stereo output pairs are available, if your audio host can handle them.

An 'Alternate' parameter also appears on all three plug-ins, and ensures that when you're playing two notes successively with the same velocity two different samples will be used, for greater realism.

The remaining audio parameters differ between *Culture*, the percussion MVI, and the other two. *Culture*'s last parameter in this area is labelled 'pre silence', and is a simple but effective option that lets you delay the onset of a Layer's notes. The most obvious use for this is to replicate the effect of several percussionists playing at once: they'll seldom hit the same notes at exactly the same time. But then when you're playing several Layers from one keyboard, neither will you! *Majestic* and *Candy*, on the other hand, benefit from a 'legato' button here which enables a monophonic playback option that's customised in the Pro editor, of which more in a moment.

Majestic's huge library isn't even hinted at when you've just loaded a Multi such as this upright bass example.

The remaining parameters customise playback in a big way. Layers can have key and velocity ranges, with scaling for the latter that lets you access the full velocity range in the plug-in from a limited range on your controller keyboard. It's also possible to enable and disable Layers from assignable keys (outside the playing range, of course!), and switch between two layers with the sustain pedal, complete with a crossfade value. This kind of instant switching between related Layers in a Multi doesn't impede your playing, and helps you create a more natural, smooth and authentic performance (depending on the Layers, of course!).

Manipulating polyphony within an MVI is part of deciding how an instrument will play. A small section on the right of the Basic editor provides amazing levels of control over polyphony, both for performance and system load purposes. The polyphony being used is reflected in a column in the Layer display: being able to see how many notes are playing at a given time will let you more easily make adjustments, and monitor how much load a given sound might put on your system.

Although the feel of these MVIs is very much of dedicated single instruments, each Layer can be assigned its own MIDI channel. Thus, true multitimbral operation is eminently possible; this may not often come into play with the bass instrument, but I can see users of the other libraries accessing Layers in this way.

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The MVI Pro Editor

The remaining (lower) portion of the MVI display is dedicated to the Pro Editor. What this provides is control over how individual Keygroups play back. You can work on one, several or all the Keygroups in the currently selected Layer/instrument. If the thought of adjusting individual playback parameters for hundreds of samples — which is what Yellow Tools' multisamples contain — sounds daunting to you, it is. But the way they've been organised makes this mass of material easily accessible.

As already explained, a Keygroup in the MVIs is one note's worth of samples, which can be up to 16 samples at different loudnesses, triggered by different MIDI velocities. Individual Keygroups are selected in the display below the on-screen keyboard (shown above). The manipulation parameters are fairly simple, as was the case with the Layer level of editing, but they're exactly what you need to customise the playing characteristics and basic sound of the supplied sample sets.

The most synth-like parameter group here is contained in the Envelope section. The envelope controls are pretty excellent, and in a signal path without a resonant filter this may well have the most impact on the final sound of an edited Keygroup. It's a five-stage affair — attack, hold, decay, sustain and release — with a 'velocity attack' parameter that lets you modify the attack portion in response to incoming velocity. If you'd like a softer attack at low velocities, set the Velo Attack parameter to a lower value than the attack parameter itself. If you don't want an envelope at all, you can switch it off.

All three MVIs offer level and pan for each Keygroup, plus a 'note off' switch. This option is ideal for triggering a hand sample or fret squeak in one Layer after a note has played, adding to the illusion of reality. Keygroups can be reversed, and a 'skip' parameter lets you move a sample's start point, under velocity control. You'll need this when reversing samples that are very long and end in silence. Keygroups can also be tuned by ±50 cents, and another 'skip' parameter lets you offset the pitch change so that the initial attack of the Keygroup is retained.

Majestic and *Culture* also have a ±12 semitone pitch offset, again with Skip option. *Majestic* has a Keygroup-level Legato control that provides fine control over how one note blends into the next, emulating legato as it would appear on a real bass. *Candy* also has a customisable aftertouch-to-volume parameter section that helps simulate the level changes a real sax player can achieve within a single note. There is no breath control option for MIDI wind controllers, though.

All in all, each MVI offers a decent level of detailed editing options. But what do they actually sound like...?

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Software Protection

I'm not a great fan of security keys — dongles — but I admit that developers who use them have good intentions. At least the USB key chosen by Yellow Tools is compact — and it's also yellow, which is kind of cool! One key is supplied with each MVI, but there is a way for you to avoid this duplication — buy one ordinary Yellow Tools virtual instrument, which will include a key, and then buy the more affordable so-called 'Loyalty Editions' of the others. All your authorisations can then be stored on the one key, saving you 80 quid on each Yellow Tools purchase you make after your first.

One good thing about the Yellow Tools key is that you can move it between computers — your licence lets you install the plug-ins on multiple machines with the one authorisation working for each.

Sadly, that's the easy bit over. I encountered a couple of hiccups when trying to authorise my MVIs and get the key to work — and it would seem, having heard the stories of other users, that I'm not alone. I had the Kafka-esque experience of knowing perfectly well that all the key-related files (and some Java gubbins that's also needed) were on my hard drive, yet the on-line checker insisted they weren't. When I booted up a plug-in, the software kept telling me the key wasn't ready.

I eventually got it sorted out by generating various ancillary files manually, with the help of someone at Yellow Tools, but it wasn't ideal. Eventually, though, I stopped moaning and was able to start playing!

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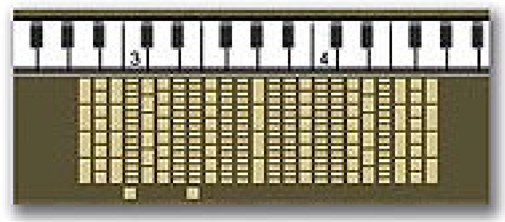
Culture — Percussion

Yellow Tools' first MVI collection is dedicated to percussion. There are two DVDs of samples (plus the main installation CD) in this set, making for a detailed 9GB of sonic material — over 25,000 individual samples. Many of the 'instruments' have different sample sets for left and right hands, and velocity-switched Layers up to 16 levels deep. Some of the instruments have over 1000 samples each. There is no mention of who played this prodigious amount of percussion, but I'll bet their arms are sore!

This massive collection omits kit drums and electronic hits and instead majors on ethnic, world, orchestral and 'industrial' instruments. I counted 318 individual instruments or instrument sets, divided into 61 groups. The world and ethnic selections include Latin, African and a few Asian instruments: djembes, darbukas, shakers, frame drums, tablas, taikos, talking drums, udus, timbales, congas, cajons, bongos, agogo, guiros, rainmakers and much more. And there may be no surprises amongst the orchestral instruments — snares, cymbals, bass drum, temple blocks, thunder sheets, tambourines, toms and so on, including such simple sounds as finger snaps and claps — but there is a terrific variety in terms of size and playing style.

'Industrial' is an intriguing (and not wholly accurate) descriptor. The samples here are of a curious selection of objects being hit: flightcases, watering cans, plastic tubes, dustbins, that sort of thing. Of course, you may have found yourself generating rhythms or fun noises with any or all of these, but when have you found time to sample them? Now you don't have to!



A close-up of a typical Keygroup display from *Culture*, showing a number of Keygroups with four, eight and 16 levels of velocity-switched samples.

Some percussion instruments are very simple — just a couple of hits, perhaps, with velocity-switched layers — and that's all you need with claps, really. But the hand drums offer a great deal more, and reveal quite a lateral approach to mapping. For example, a hand drum would normally be played by two hands... and that's what you do with the *Culture* equivalent. Yellow Tools have adopted a certain amount of common sample mapping throughout their percussion sets, particularly the hand drums.

The note G# is taken as a central point — it'll often have a hand stroke effect assigned to it. The left-hand samples are then arrayed to the left of this note and the right-hand ones are assigned to the right.

Symmetry is the key here, with related hits assigned to ascending or descending semitones either side of the G#. For example, a Djembe starts with the G# hand effect. Moving in semitones in either direction, your fingers will find left- and right-handed varieties of bass tone, open tone, mute, slap, finger, close slap, fingered flam, open flam, bass flam and slapped flam.

The snare instruments are even more detailed, with over 700 samples, and some Latin Layer sets offer three drums, each appearing in a different octave, but with the same symmetrical arrangement. So rather than feeling as though you're trying to play drums with a chromatic keyboard, you get the impression that the keyboard has been subverted into something more like a percussion controller, and you play patterns in a more drummer-like way than would normally be the case, without the 'choking' or 'machine-gunning' effects that often result from playing rhythm on a keyboard. All hits are allowed to breathe, and in some ways it's almost like drumming patterns with your fingers on a table, except you're triggering rich, expressive samples of real percussion.

The set contains a few disappointments — I was sad to see that 'gamelan chimes asian' simply contains long and mute chimes, and something equally gamelan-like is promised by 'gong set Burma', but frustratingly, it's been sampled in equal temperament. Perhaps the collective heads of Yellow Tools' developers would have exploded at the thought of sampling a typical Javanese gamelan, or similar gong, bell, drum and chime ensemble from other cultures — but in fact I think their meticulous nature and lateral approach to keymapping would have suited the job.

These niggles aside, Yellow Tools' approach has resulted in a percussion collection of massive appeal. So many users will appreciate the depth and weight of the samples — you often feel as if the drums really are in your room, a feeling helped by the way all the sampled hits have been allowed to decay to silence. It's easy to feel you're actually playing the drums with *Culture*, and not just triggering samples, thanks to the thoughtful multisampling.

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Culture Groove Pack

If you're keen on the percussive opportunities offered by *Culture*, but know that rhythm programming is not your strong point, Yellow Tools have the answer. Patterns haven't been built into their plug-in — an approach taken by many other developers — but registered users can download the *Culture Groove Pack 1* from the Yellow Tools web site. This collection gives you loads more Multi and Layer patches, plus over 1100 MIDI Files. These files can be up to two minutes long, and offer breaks, fill-ins, fill-outs and so on in a wide range of styles. The playing is natural and seems unquantised, and really shows off *Culture* remarkably well.

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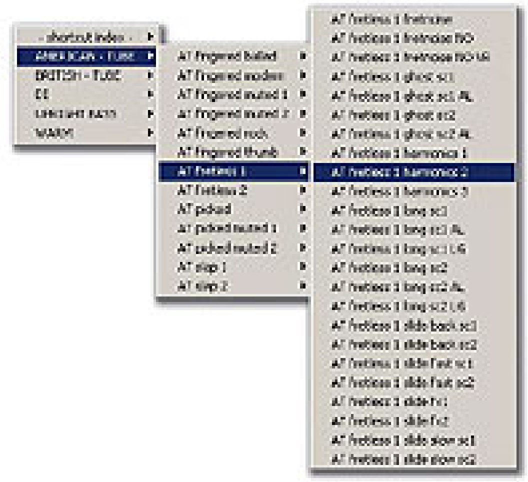
Majestic — Basses

Majestic is the largest sample collection of the three MVIs, taking up no fewer than four DVDs. This entire review could be taken up with listing the variations available: there are thousands of samples here, and every one of them was played by German session bassist UMBO (or Unidentified Moving Bass Object), the player on Yellow Tools' original *E Basses* collections.

To summarise this set, let's just say that UMBO plays a number of excellent (but equally unidentified) basses though a number of amps. There are four broad categories of Multi and Layer: American tube, British tube, DI and 'Warm' (the last collection is only available for download to registered users). The names speak for themselves, though some more detailed indication of *which* American and British hardware was used would have been nice. 'Warm' is a catch-all for fully produced bass sounds processed by EQ and compression to suit a range of specific styles.

Each set contains the same collection of somewhat curiously titled playing styles, (fingered ballad, fingered modern, fingered muted 1 and 2, fingered rock, fingered thumb — sic! — fretless 1 and 2, picked, picked muted 1 and 2 and slap 1 and 2). The choices within each sub-category are also duplicated. The exception here is the Warm set: the collection is organised by style, with categories like 'American ballad', 'American hip-hop', 'British ballad', 'European rock', and so on. There are 26 styles, each with a number of choices.



An example of the nested menus that pop up when hunting through the vast number of bass instrument Layers within *Majestic*.

The individual Layers, or instruments, come in a variety of flavours, so you can access harmonics, slides, tapping, hammer-ons, fret noise and special effects. The sample sets start at a low 'E', or 'D', or 'C' (so five-string instruments are part of the set), and range just over three octaves, with multiple samples assigned to velocity splits for each note. Looping is used in this plug-in, but with no unwanted artefacts.

Amongst all the electric bass, there's an unexpected bonus in the shape of a meticulously sampled century-old upright bass. Truth be told, it sits a little uncomfortably in the collection, but it is definitely worth having. Again, various playing styles, tricks, harmonics and noise are supplied. It's been played by fingers all the way, with no bowing, but that suits jazz and pop contexts better anyway.

It can be fairly mind-numbing to listen to this collection all the way through; a certain 'sameyness' does creep in when listening to the repeated styles and effects. But then this is a four-DVD set of samples for the same price as the other two collections, so I think I can forgive that!

There is no denying the quality of the samples, and you *will* find something suitable for nearly every electric bass (and some upright) requirements. As with the *Culture* set, there's a fabulous depth to the plug-in's output. Clever Multis let you effortlessly switch, crossfade or otherwise access multiple basses to create a finished performance that could have been achieved on a real bass, moving from mute notes to long sustains, and fret noise on a note off to convincing slides and hammer-ons — the works. On the subject of tuning, any Layer can be 'moved' along the lower keyboard display, making for instant transposition, so if the lowest note in your current selection isn't low enough, you can just drag it; transposition artefacts don't really kick in for a couple of semitones.

There will always be something a little odd about playing bass guitar from a keyboard, as much as there is about playing percussion in the same way. Yellow Tools have organised the sample collections to gently manoeuvre the player into the right mindset, and the audio feedback tends to make you play more like a bassist. It helps if you actually *play* bass, of course, but the sounds and subtle touches do tend to move you that way, even if you don't.

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MIDI Automation

Many MVI Keygroup and some Layer parameters can be automated from external MIDI control surfaces or your host sequencer. This aspect of the MVIs isn't as well documented as it might be, and is a little confusing. What's not made clear is that if different Layers are set to respond to different MIDI channels, then each Layer's parameters can be controlled independently via those MIDI channels. The documentation seems to indicate these are global assignments.

The Keygroup parameter assignments *are* global, with one MIDI controller-to-Keygroup parameter assignment tweaking that parameter in all Keygroups in a Layer. A list of controller assignments is provided, so you'll be able to set up your host sequencer or external controller accordingly. But be prepared for a surprise: some parameters that do not graphically appear on screen are still accessible over MIDI. *Culture*'s 'pre-silence' parameter is one example, and can also be tweaked, if you need to, on *Candy* and *Majestic*. The Keygroup level semitone pitch parameter can also be accessed in this way within *Candy*, though the AT Volume parameter knob actually moves on screen. Clearly, a little more polishing is needed, but the basic implementation is sound.

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Candy — Saxophones

The name of Yellow Tools' most recent MVI apparently refers back to Dutch sax whizz Candy Dulfer, although all the saxes sampled on the library were played by German session saxophonist Ralph Gundel. His was a new name to me, too, but on this evidence, he plays really well, with an excellent tone. On *Candy*, he airs every modern member of the sax family — soprano, alto, tenor, baritone and even bass sax.

Having a look at the plug-in's content list reveals that a total of nine instruments were used in the *Candy* sessions, from such noted manufacturers as Yanagisawa, Keilwerth and Selmer, and with a range of mouthpieces and various reed options. This level of detail makes it doubly strange that the equivalent data wasn't provided for the bass collection! Each instrument has been sampled playing long and short notes at several dynamic levels, with and without vibrato, but there are also honks, key slaps, falls, growls and other effects in the set. Despite this typical attention to detail, *Candy* remains the smallest set of the current MVIs: everything, including the 8GB sound library, fits onto one DVD.

I was looking forward to reviewing this MVI least of all — in my experience, saxophones and sampling are not usually a happy combination. If the bass performance is an expression of an individual's connection with their instrument, it's even more the case with sax. There are so many performance nuances available to sax players — and that's without considering the effect of changing mouthpieces and reeds — that sampled saxes can often just sound like polyphonic harmonicas in the wrong hands. Plus, it has to be said, sax can be so cheesy in so many pop contexts.

But a pleasant surprise awaited me. As I was expecting by now, the recordings were exemplary, and even though *Candy* features the smallest collection of samples of the MVIs on review here, there is still a great deal of variety. Timbrally and stylistically, the performance bias is jazzy, but in a classy rather than obvious way. Solo sax parts worked much better than expected from the keyboard, and even more so from my old Casio MIDI horn.

Saxes galore with *Candy*; this is me experimenting with a small sax ensemble.

In ensembles, too, *Candy*'s box of saxes surprises and pleases — there's enough here to attempt big-band horn sections, and Phil Spector's wall of sound (just add reverb and brick-wall that compression!). To challenge the plug-in even further, I approached it as a saxophone quartet, and I was surprised to find that my efforts largely paid off! The eight-Layer Multi suits this approach, since I was able to double each instrument, and switch different timbres by velocity or MIDI keys. This way, I could have a genuine mezzoforte alto sax crossfading or switching with a full-on forte variant, for example, making for a more dynamic 'performance' without working too hard. Perhaps not enough timbral variety to ape a group as *avant-garde* as the Rova Saxophone Quartet, but enough for most normal situations! And I loved having bass sax as part of the ensemble: if your arrangements don't seem anchored enough with the baritone sax, then nothing comes more solid than this. It's not a sound for every occasion, but it's worth having.

Instrument ranges are pretty closely adhered to: the lowest note of the sampled instrument is the lowest note of the instrument being played, and the highest note tends to be a couple of semitones higher than the real instrument's nominal range. Saxes usually have a range of about two octaves and a fifth, but experienced players can often move into the 'altissimo' register, up to a sixth higher than that. *Candy* plays it safe, but not *too* safe!

I liked Ralph Gundel's playing a lot. I get the feeling he was really focused during the recording of all these long notes (and key slaps and mouthpiece pops). He has a really warm sound, with an almost oboe-like feel to his soprano playing. This feel helps when trying to use the instruments in a solo context. Once again, the keyboard seems to take a back seat as you play sounds which drag you into playing with an eerily sax-like articulation.

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Conclusions

Summing up a single piece of software you like can be tricky, but juggling three is even more so. Broadly speaking, these are well-recorded sample collections married to a tidy, sophisticated, easy-to-use playback system. They're not the cheapest instrument-based libraries around, but I'd say their quality makes them worth it.

There's no denying that although Yellow Tools' MVI engine is very efficient, the sample-heavy nature of nearly all the Layers will make demands on your computer that will only be met if you have sufficient RAM — and having all three accessible simultaneously might well mean you need to increase your hard drive capacity. But that's it for nit picking. It's the sound that counts, and these MVIs are quality products. The depth and presence of the percussion and bass sets has to be heard to be appreciated. If your monitoring includes a subwoofer, be prepared to notice some serious floor movement! The saxes are also excellent, ear candy helped along by some splendid playing.

Finally, there's that elusive playability: the MVI experience somehow massages you into playing your keyboard as if it *were* a real drum, bass or sax. And what's *that* kind of magic worth? SOS

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SOUND ON SOUND
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Q. Is it worth isolating my speakers and other equipment?

Published in SOS August 2005

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Sound Advice

I would like to know how much benefit can be gained by isolating my speakers and other gear from their supports — so-called 'seismic isolation'. I recently saw a forum posting suggesting that passive monitors have an advantage over active ones as, in the case of the latter, vibrations from the speaker can affect the components of the built-in amp. The benefits of decoupling equipment using springs that have a very low resonance frequency so that it 'floats' was also discussed. Apparently many things can benefit from this technique — not just speakers but CD players and studio gear also. Improvements to the stereo field and depth are said to be quite noticeable. I would like to know if the £1300 I spent on an active Blue Sky System One (which I like very much) would have been better spent on passive speakers and an external amplifier. Can you shed any light on all of this for me please?

***SOS* Forum Post**

Technical Editor Hugh Robjohns replies: With regards to what you read about internal vibrations in monitors, in general neither the active or passive form has an advantage in this regard, and both potentially suffer exactly the same problem.

Clearly, there is a lot of sound energy inside most loudspeaker cabinets, and if that energy is allowed to impact on electronic circuit boards it is possible that some components might resonate and vibrate, eventually resulting in damage to the solder joints or the components themselves, and possibly such mechanical resonances might affect the electrical signal passing through the components. However, this would apply equally to passive crossover boards as much as active amplifiers.

With properly designed and constructed

In 30-odd years of playing around with loudspeakers in many and various forms, I can't say I have ever found this to be a real problem. I have occasionally come across speakers that have suffered component or solder joint failures, but in all cases the causes have been traced to faulty production or failures in quality control. When the faults were fixed properly, none recurred as far as I am aware — even though you would expect them to if the sole cause was sound vibrations within the cabinet. So I am confident that this argument can be set aside as a popular but completely unfounded myth.

monitors, whether active or passive, you needn't worry about internal vibrations damaging electrical components.

Mechanical isolation of speakers or other devices from their supports can be used to advantage in certain situations, but it is a complex subject and it is easy for the inexperienced to make the situation worse with inappropriate decoupling systems. In my experience, most equipment works best when mounted on solid, heavy supports — there is nothing as effective at controlling vibrations as a lot of mass.

However, sometimes it is necessary to come up with some form of decoupling to prevent vibrations generated in one source from entering an adjacent surface. The classic example is that of placing nearfield speakers on a desktop, when the inherent speaker cabinet vibrations will often cause the desktop to vibrate and resonate, resulting in unwanted rattles. In this situation, placing the speakers on some form of decoupling medium can improve matters — something like the Auralex Mo-Pads, for example, are very effective. However, far better results can be obtained by removing the speakers away from the table top completely and mounting them properly on solid, heavy stands placed directly on the floor.

As far as equipment is concerned, I don't subscribe to the view that properly designed and manufactured amplifiers and other electronics should be decoupled to improve stereo imaging or anything else. However, when it comes to systems involving some mechanical element — like record players, CD players and so forth — unwanted vibrations entering the mechanical system can cause problems.

Most people are very well aware of the susceptibility of record players to external mechanical or acoustic vibration. The required tracking precision in CD players and DVD players is many orders higher, and mechanical vibrations that reach the mechanism will affect the accuracy of the tracking. Potentially, this will cause the tracking and focus servos to work harder, forcing greater current flows at higher frequencies through the motors. In cheaper designs, this may well affect the power supply's stability and result in noise currents reaching other parts of the circuitry. Reduced tracking precision can also potentially result in a greater uncorrected error rate and far more jitter. Cheap and poorly designed players are likely to suffer these effects to a much higher degree than properly engineered equipment, which will usually incorporate properly decoupled drives, effective de-jittering circuitry, and so on.

Auralex Mopads can be used to isolate monitors placed on a desktop, but heavy-duty floor stands are best of all.

It's a familiar scenario in the hi-fi world — people discover that badly engineered equipment reacts 'unexpectedly' to different cables, mechanical decoupling, or painting with a green pen — all of which bestow a 'miraculous' benefit to the sound... and then declare (from no scientific basis whatever) that all vaguely similar equipment will behave the same. It's just not the case.

As to whether you would have been better off buying passive monitors and an amp, the answer is probably not. I can think of some excellent passive monitor and amp combinations for the rough cost you mention, and in direct comparisons I dare say some people would prefer a passive speaker and amp configuration over your Blue Sky System One. But it comes down to personal preferences regarding sound, convenience and styling, and how the system works in a given room. I think you can continue to enjoy your Blue Sky system and completely disregard any *faux* concerns raised by the technical myth-spreaders! SOS

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Q. Should I sync my MIDI gear with my multitracker?

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Sound Advice

I recently bought a Yamaha AW4416 digital multitracker and am in the process of getting to grips with its features. I'm going to be recording some vocals and guitars to the hard drive, but I use a lot of MIDI-sequenced sound modules for backing, so I am considering the option of sync'ing my sequencer with the AW and running the modules in time with the recorded material. Nevertheless, I'm still not sure if it would be more sensible for me to record the outputs of the modules to the hard drive. What are the pros and cons of each method?

Chris Duerden

***SOS* contributor Tom Flint replies:** First, let us deal with the MIDI synchronisation option. External sequencers can be sync'ed to the AW4416 via MTC or MIDI Clock. There are several advantages of working in this way, the main one being that the sequenced part of a composition is always editable, right to the point where everything gets mixed down to stereo. This means that a basic MIDI arrangement can be developed hand-in-hand with recorded audio parts. Indeed, it's often the case that once a vocal or guitar arrangement is added to a composition, some part of the MIDI backing becomes redundant and needs to be removed or changed.

As far as audio routing is concerned, unless you want to use an external mixer, the setup requires the sound modules to have their own set of input channels and sockets in the AW4416, which can get complicated if you are regularly using the inputs to record multiple instruments to the hard drive. If you have a number of MIDI sources you might consider buying an analogue input card so that the modules can be permanently connected without obstructing the inputs you typically use for recording. For example, Yamaha's MY8AD and MY8AD24 YGDAI expansion cards (the latter being a 24-bit version), provide eight balanced TRS jack inputs, allowing four stereo modules to be connected and routed to input channels without troubling any of the standard ins.

Yamaha's AW4416 can sync external MIDI gear via MIDI Clock or MTC.

The main problem with the MIDI sync approach becomes apparent if you have to take your recorder out of the studio to a session, to record a vocalist or instrumentalist, for example. If none of the sound module parts are recorded, you would have to either take the entire MIDI rig to the session, including sequencer, effects and modules, or you'd need to bounce all the audio sources to a spare couple of tracks in order to create a guide part (this is done by routing all the relevant input channels to a pair of busses, and then assigning them to a pair of record tracks).

One further drawback is that there are more chances for a non-recallable setting to get altered accidentally. For example, you could spend ages carefully setting up EQ, level and processor settings for each mixer channel, only to find the mix balance completely altered by a nudge of a sound module's volume fader, or tweak of a global parameter setting. The problem is likely to be compounded further if you continually have to unplug sound modules from the inputs so that vocals or guitars can be recorded. In such circumstances, the AW4416's preamp pots will need to be reset to their mix position after every recording session. Problems such as this are definitely worth taking into account if you are someone who takes equipment out on the road regularly, if you are likely to adjust the settings of certain modules from one song to another, or if you intend to use the AW for a variety of jobs.

I use an AW4416 sync'ed to a MIDI setup, and at the moment I record my MIDI gear to the AW because most of my songs have MIDI arrangements that were carefully prepared as backing for live gigs. I like the fact that once the sound modules are recorded they can be disconnected, freeing up AW4416's input sockets and channels for other uses, and it's comforting to know that a composition will not be damaged if I adjust a few module faders or preamp levels. I've also found it useful to have been able to take the AW to recording sessions across the country without needing to do any submixing or take MIDI outboard with me.

The recording method's most obvious drawback is that it uses up valuable audio tracks that could otherwise be use for massed backing vocals or other parts, although by making good use of virtual tracks and the bussing structure it is still possible to comp multiple backing parts — see the Yamaha AW4416 User Tips article in *SOS* June 2005 for more on how to do this (www.soundonsound.com/sos/jun05/articles/yamahaawtips.htm).

Another problem arises if the recorder and sequencer have not been synchronised and a MIDI part needs modifying late on in the recording process. This is because the replayed MIDI part is likely to be out of time with the rest of the composition. Fortunately the AW's onboard editing tools make it possible to move audio in time.

The AW4416 User Tips features didn't explain how to do this, so here's an example, assuming that the part about to be replaced is a sequenced performance played from a stereo drum machine.

First off, record the new part to a pair of unused tracks or virtual tracks, but leave the old drums intact. Once recording is complete, the exact relative position of the old and new recordings needs to be established. To do this, stop the recorder on the original drum track's first distinct beat, navigate to the waveform page and use the data wheel to scroll to the beginning of the wave. Identifying individual beats visually is easier when the waveform display is set to a low resolution, and therefore has small vertical (Amp) and horizontal (Time) values. A low Time number also makes scrolling to the correct point considerably faster. Nevertheless, once the position marker is at the start of the beat, a finer resolution needs to be selected for precise positioning. I rarely use a time resolution smaller than 'x2048' for this purpose.

The Yamaha MY8AD expansion card adds a further eight balanced audio inputs to the AW4416.

Next, press Locate and note down the exact time value. I like to double-check the position by looking at the waveform of the other track in the stereo pair The two should match, but off-centre panning can alter the point at which the waveform is visible, so it's best to be sure. If all is OK, the same beat-locating procedure can be used to determine the corresponding start point in the newly recorded drum track. Once again, note its position down exactly, making sure that the figure is taken when Time and Amp settings have the same resolution as was used previously.

Next it's time to start editing. For the sake of this example, let's say that the original drum pair began at 05.240 but the new drums start later at 05.365. It's then simply a matter of subtracting 240 from 365 to leave 0.125. Therefore 0.125 needs to be cut from the start of the new tracks. Before deleting anything, it's worth pairing the drum tracks so that any editing work done to one half of a pair is automatically applied to the other. Track, as oppose to channel, pairing is done by clicking on the relevant broken-heart symbol in either the TR Editing screen or the V Track page.

To make the edit, select Part rather than Track or Region, and choose to 'Delete' the relevant track numbers. Set the start point at zero and end point to 0.125. After the edit, check that the position of the waveform matches that of the old audio, and do a listening test to ensure the work has been successful. If you are playing back the new tracks next to the old there should be a degree of phasing and cancellation all the way through, thus proving that the new parts are about as close as they needed to be. The old drums can be deleted as soon as the new parts have been saved to disk.

On other occasions you may have to insert space rather than cut it to align the parts, but process is identical, except that, from the Edit Part menu, Insert needs to be selected rather than Delete.

So as you can see, both methods do have pros, cons and workarounds, the choice just depends on how you prefer to do things. SOS

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Q. Should my Valve Mic be this noisy?

Published in SOS August 2005

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Sound Advice

I have just got hold of a CAD M9 valve mic and am concerned about the noise level from it. When I switch it on it chuffs and farts a bit, much as my Fender valve amp does, then settles down. That seems OK. But once it has warmed up, the background noise level seems high compared to my other condensers. If I have a vocal take at normal levels being recorded at about -6dB peak, then the noise level is registering at -38dB. The vocal sounds fine, but the noise seems high. Is this what I should expect from a valve mic? Should I try another valve in it?

The self-noise figure of valve mics such as the CAD M9 is generally slightly higher than that of their solid-state equivalents, but should still be well within usable limits.

***SOS* Forum Post**

Technical Editor Hugh Robjohns replies: Valve mics are generally more noisy than solid-state condensers. The M9 is specified with a self-noise figure of 15dBA, which is roughly 8dB higher than the best of the large-diaphragm solid-state designs — the Neumann TLM103 has a self-noise figure of 7dBA, for example.

However, while it is possible that your mic is faulty or requires a new valve, the high noise floor you describe could also be down to poor mic technique.

With vocals peaking at -6dBfs, a noise floor of -38dBfs does seem poor. The question is, how much of that noise floor is due to the mic, and how much is due to the recording environment? Are you recording a low-volume source at a considerable distance, or in a noisy room, or with a poor-quality mic preamp, for example?

If you have access to another large-diaphragm mic, I would suggest you rig that alongside the M9 and adjust the gain to get the signal peaking at the same level for both mics, and then compare the background noise floors. If both mics deliver similar noise levels, then the room or your technique are at fault. If the M9 is more than a few dBs noisier than a solid-state large-diaphragm mic, then the M9 is in need of repair.

It could be that the valve is faulty or worn out, and certainly that's the easiest thing to replace yourself. However, there could also be a problem in the power supply or elsewhere in the mic's output circuitry, which would require a return to the supplier to be fixed. SOS

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A faulty valve can easily be replaced; more serious repairs should be carried out by an experienced technician.

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Q. What is different about the many varieties of Dolby noise reduction?

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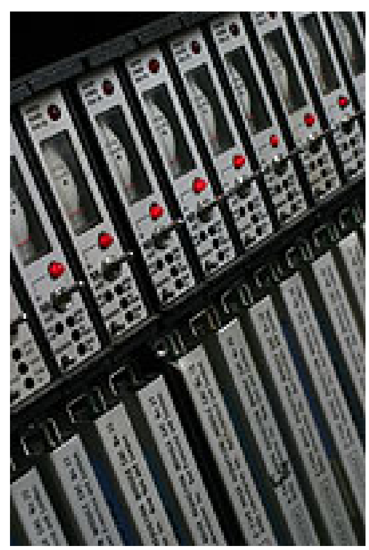
Sound Advice

I never did quite understand the subtle differences between all the different variants of Dolby — A, B, C, HX and SR. Could you explain them to me? Are there any others I've missed? What are Dolby Labs doing these days? I guess they've undergone some 'reduction' themselves...

***SOS* Forum Post**

Technical Editor Hugh Robjohns replies: Dolby A was the first professional noise-reduction system — launched in 1967 if memory serves — and it used four separate frequency processing bands. You can think of them crudely as bass, mid-range, treble and high treble, with the top two overlapping so that the 'hiss region' was processed more heavily than the rest. Avoiding line-up errors between encoding and decoding was crucial, so the infamous Dolby warble tone was used to identify encoded tapes and to allow accurate replay alignment. Dolby A was originally used to get respectable audio performance out of early professional video recorders, but was later adopted for multitrack recording and cinema optical soundtracks.

Dolby B was a very simple domestic system intended to improve the performance of compact cassette recorders. It was also used on some later domestic quarter-inch machines. Dolby B was a single-band system affecting only the high end, with very modest compansion. It had no facility, or indeed any practical need, for replay alignment.



A rack of Dolby A NR modules, Dolby Labs' first professional noise reduction system.

Dolby C was a much more aggressive multi-band version originally intended for small-format professional videotape systems and narrow-gauge semi-professional studio multitrack recorders. It was very sensitive to mistracking, but was unfortunately designed without any line-up tone facility to calibrate playback levels.

In the professional market, Dolby A was superseded by Dolby SR, which was Dolby's most sophisticated multi-band noise reduction system. This employed 10 bands altogether, some operating at fixed frequencies and others moving automatically to suit the material, and allowed the user to get a signal-to-noise ratio of around 90dB from analogue tape. However, although it was a very clever and effective system it arrived just a few years too late and the digital revolution effectively eclipsed it. Dolby SR used a modulated noise signal for identification and replay alignment.

Finally, Dolby S (one you missed off your list) was a last-ditch attempt aimed at semi-pro and domestic recorders, and was a halfway house between Dolby SR and Dolby C. It still had no built-in line-up facility, though. It was used on some semi-pro narrow-gauge multitrackers and the last of the high-end hi-fi cassette recorders.

Dolby HX is not a noise-reduction system at all — it is a clever system to avoid over-biasing on analogue tape machines using high-output tapes. This system was used on some high-end domestic cassette recorders and the last of the professional analogue two-track machines, such as the Studer A807. Dolby HX is a once-only process that needs no decoding. In essence, it reduces the bias level if there is a lot of high-frequency content in the audio signal, thus preventing over-biasing and the noise artefacts and frequency-response errors that go with it.

Dolby Labs still make Dolby SR and A systems for analogue multitrack and cinema applications, and I guess they are still collecting licensing revenues from the other systems when they are used on domestic cassette recorders and the like. However, most of the company's efforts these days are geared towards digital data-reduction systems, which are based entirely on the frequency-masking principles first exploited by Dolby's analogue noise-reduction systems. That is why Dolby AC3 has always been amongst the best of the data-reduction codecs for a given data rate — the company had a major head start on the rest of the field. SOS

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SOUND ON SOUND
The World's Best Music Recording Magazine

Business End

Reader Tracks commercially evaluated

Published in SOS August 2005

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People : Miscellaneous

Business End enables you to have your demo reviewed by a panel of producers, songwriters, musicians and managers. If you want your demo to be heard by them, please mark it 'Business End'.

Shared Nightmare Club

Liz McCudden (LM): "So is he a one-man outfit? I can't make out if it's real musicians."

Kevin Paul (KP): "I think it's electronic drums but real guitar and bass."

Joff Gladwell (JG): "I think this is all right but I think it's more of a vibe thing than a song thing. The second track, 'Who's That Girl', has got quite a hooky chorus though."

KP: "It's got quite a trashy sound hasn't it?"

JG: "Yeah. I really like the recording on the second one — it's kind of like a lost psychedelic record from the '60s or something. The third one's got a similar feel to it as well, I like that.

"I do think it's probably better for the vibe than for the songs but I do like the vibe."

Joe Vanags Fleming (JF): "I think I feel the same. I found the arrangements really strange though, you just feel like it's a bit random and you don't know what's going to happen."

JG: "The first one, particularly, had a weird arrangement."

LM: "I wouldn't have put that one first, not on a demo. The rest of it's so different from the first track and you might not get past that to hear it."

KP: "The first track seems to suggest a certain style but then he never comes back to it. You know — it's quite electronic and quite dark but then it goes all rocky and even a bit psychedelic. His musical direction was... Well I don't quite know what it was."

LM: "I think if you can't hit high notes you shouldn't try. I think it's the third song which is worse for that. You need to stick to your range."

KP: "The recording's pretty good though, it sounds quite good — nice use of effects in places. I think maybe on the fifth track the vocals are a bit quiet but that's quite a minor thing really."

JF: "It all seems a bit simple to me, like he was trying different ideas for different tracks but then he didn't really take them very far, like he didn't really let them develop. It's a bit like each song is a different idea."

LM: "I think he's showing his age with his influences. It really makes me think of the Stone Roses or the Charlatans, you know, that baggy sort sound. Actually, he says in his letter that he's been making music for 17 years and I think that shows a bit."

KP: "I reckon he just needs to work on his songwriting a bit. The songs sound a bit like they're jams at the moment, he just needs to work on the structures because there's some good stuff in here — the chorus on the second track especially. Maybe he should get together with someone else, it can be hard writing stuff on your own. Actually he mentions that he's been swapping ideas with someone, maybe they should get together and make it more of a band."

LM: "I thought some of the lyrics were a bit mundane. It's like 'mowing the lawn, la la la', you don't need to hear about that — you should put music on to think about things other than everyday life I think."

KP: "You get the impression that he knows what he's doing but he's not quite sure what he wants to do. It doesn't really take you anywhere because the style of music keeps jumping around. It's definitely a good effort though, he just needs a bit more direction and a bit more focus."

JF: "Yeah, that's right, he basically needs to hone his ideas and work out exactly what he wants to create."

JG: "And don't put 10 tracks on a demo, just put three good tracks on it — people can always ask you for more if they want to hear some more. We get about 20 demos every day and it would be impossible to listen to 10 tracks on every one. And always put your best track on first. If this guy had put the second one on first you'd be intrigued to hear more — it's pretty poppy and accessible, it's got a hooky chorus and the production's quite interesting as well."

LM: "You can't afford to make people wait for the good tracks when you're sending a demo, just put on your three best ones."

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Fluid

KP: "Seventeen tracks."

JF: "Seventeen? And after what we've just been saying about 10 being too many!"

KP: "This sounds like the sort of thing that was being done in the late-'80s, early-'90s. It sounds a bit like early Future Sound of London or something like that. I think the name 'Fluid' suggests that sort of Black Dog-style techno and that's exactly what it is.

"What do you think you're supposed to do to this sort of music? I mean are you you supposed to go mental and rave to it or are you supposed to get really stoned and chill out? I mean it's not really dancing music."

JF: "No, it's too stop-start for that."

LM: "Armchair disco."

JF: "It's definitely getting better as the CD goes on. It's a shame this is number 15, we easily could have missed it."

JG: "He could have put the CD in reverse order."

JF: "Or just put the second half on it."

JG: "I just think this sounds like it's from a different time really. It sounds pretty old and I think the sounds he's using seem quite dated. It's kind of like a weird cut-up, it never really gets into a groove."

JF: "Yeah, it's quite disjointed."

JG: "Yeah, but it's not disjointed in an interesting way, it's disjointed in a kind of arbitrary, lazy way."

LM: "It's not very fluid."

JF: "Despite the name."

KP: "He says in his letter that he's been making music for a year and that shows a bit I think. The stuff towards the end is a lot better though. A lot more listenable."

JF: "They're a lot more polished, aren't they?"

KP: "There are better people doing the sort of thing he's doing out there though."

JF: "There were better people 10 years ago."

KP: "There's some pretty tough competition with this kind of music. You really need an original take on it to give you an edge. His ideas are only half there and with electronic music you do need to be a bit more creative and inventive.

"The drum programming was a lot better towards the end, a bit simple maybe but it was actually cohesive and it made some sense. I think it's the 'Amen Brother' breakbeat on track 15 — that's a classic breakbeat and the track was pretty interesting. The mix on number 16 was actually pretty decent, especially compared to the earlier tracks where you couldn't really hear the synths because the drums were too loud."

LM: "Maybe he's been a DJ for years."

KP: "What, and gone deaf? No, I'm joking, if he's only been doing this for a year and the stuff at the end is the new stuff then he's not doing badly at all."

LM: "There's just nothing about this which is different from other music like it."

JG: "I think he needs to listen to more music, especially more modern music. Maybe we should send him some Mute compilations."

JF: "If this CD is actually everything he's ever done in chronological order then he's definitely improved as he's gone along."

KP: "It seems like the early tracks are a big learning curve, at least with the later ones you get the feeling he's got a better idea about what he's trying to make."

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Ascoltare

KP: "After just one minute this is the most interesting thing we've heard so far. I always quite like this style of music anyway. I'm always quite interested to know how people make this sort of music. You know, just the mind-set you have to be in to make it. I'm quite curious about it, we get to hear quite a lot of it at Mute.

"The mix is quite good, it's a bit lo-fi but that doesn't really matter with abstract stuff like this. I reckon he could get someone to put this out actually."

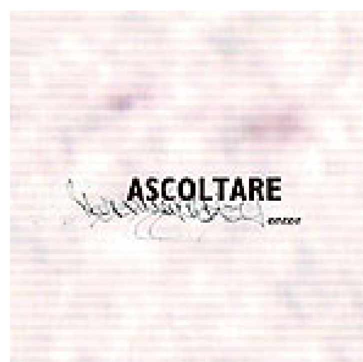
JF: "How far are we in?"

KP: "9:50 out of 10:30, we've listened to the whole thing, — that's the first one."

JG: "This guy certainly doesn't need any Mute compilations."

KP: "I think it's quite cool just putting one 10-minute track on a CD and sticking it a brown envelope with a web site address and a phone number on it. With this sort of thing that's all you need really. It's quite curious music anyway — you just go and check out the web site."

LM: "The thing I don't like about this sort of music is that there's never anything to look at at gigs. It's just a man behind a laptop."



JF: "Sometimes they have lights or video projections, I think that can work quite well."

JG: "I think it's good. We get a lot of this sort of thing at Mute though, we've got a big bag of them downstairs. I do think it's good but I don't know how special it is compared to other stuff. It's inventive though, he's got some good sounds going on — there's one bit where it sounds like a music box going through a photocopier or something. It's got a really pretty sound but it's really mechanical as well — I like that. I'm sure there's loads of labels out there who would be happy to put this out and sell a few hundred copies."

KP: "I don't think he'd have any problems getting someone to put this out as it is. This sort of thing's quite relevant at the moment and lots of people are doing it, labels like Mille Plateau for example."

JG: "I think with the other demos we've heard so far the people were a bit unsure about what they were trying to do, but you get the impression that this guy to know exactly what he's doing and what he wants. He's pulled it off really well."

JF: "It's hard to distinguish between songs like this. It's difficult to say what makes the difference between something being released and not being. I don't really know enough about the field to say how good it is compared to other things but I quite like it."

LM: "It doesn't sound like a demo really."

KP: "It sounds like a record."

LM: "If someone stuck that on in a set I wouldn't know that it was a demo."

JF: "You could probably get this on an advert — surreal, abstract music always does quite well. It's not going to get on Capital Radio or anything but there's definitely a market for this sort of thing."

LM: "As long as when he's playing live he doesn't scare people with that interference noise at the start. At first I though 'oh no, it's one of them' — try and drive everyone out of the room and then feel really smug when it's empty. But he wasn't like that, he just has to be careful playing live."

KP: "The fact we managed to listen to all of it tells you it must be pretty decent. I like the brown-envelope style packaging. Minimal record — minimal packaging."

JF: "It gives it a certain aura of mystique. Well, assuming he's not just a lazy bastard."

KP: "I'll definitely go and check out the web site."

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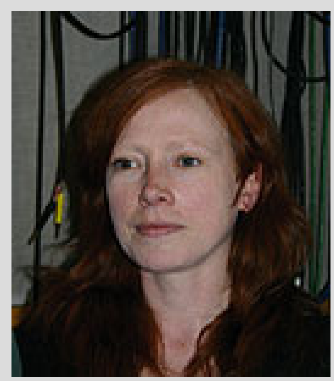
This Month's Panel



Kevin Paul began his career in audio engineering as a tea-boy at Ray Davies' Konk Studios. He joined the Instrument at Mute in '94 and became Head Engineer there in '97. Over the course of his career he has worked with David Bowie, the KLF and, of course, a host of Mute artists, including Goldfrapp and Nick Cave and the Bad Seeds,

Recently Kevin has been concentrating on 5.1 mixing and is about to release the debut album by his band AGK (www.agkmusic.com).

Joff Gladwell is an A&R scout for Mute Records. His daily rountine involves listening to demos, going to gigs and talking to people about bands — which is very nice work if you can get it.

He's probably earned it though: Joff started his own record label, Bad Jazz, when he was 16 and spent the next seven years releasing diverse and eclectic music by acts like My Morning Jacket, Lone Pigeon and James Yorkston. He joined Mute in his current position in 2002.

Joe Vanags Fleming is Assistant Publisher at Mute Song, Mute Records' publishing division. After graduating from university Joe worked for the Performing Rights Society (PRS) before going on to join Mute.

His job as Assistant Publisher involves liasing with Mute's international sub-publishers, processing royalties and occaisional A&R work with urban and dance music.

Joe is also a DJ and has recently began producing his own music.

Liz McCudden is Product manager at Mute Records, a job which involves organising the scheduling, production and promotion for the many different acts on the label.

Liz's career in music began in the live industry where she worked as Stage Manager at Shepherd's Bush Empire and in some of Londons' other larger live venues.

After a spell working for record label City Slang she went on to join Mute in 2003.

Many thanks to MJ and The Instrument at Mute Records (www.the-instrument.com) for organising and hosting the session.

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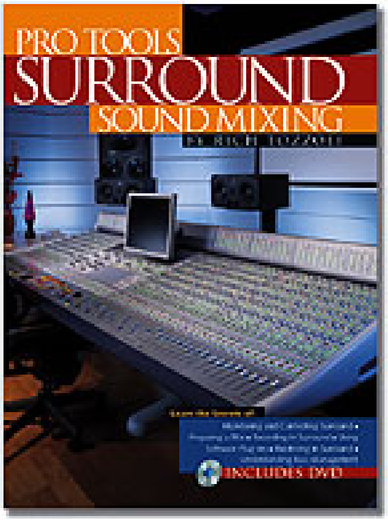

Crosstalk: Your Feedback

Response to your letters, emails and calls

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People

Source Of Contention

In the July issue's Leader column, I think Paul has unintentionally misrepresented the views of free software developers producing audio and music applications for Linux and other platforms. There are people who think software should be free, but they're not demanding that proprietary software houses give away their creations.

Instead, they are writing their own software and giving the source code away, which of course they are perfectly entitled to do. It has practical benefits because software needs active maintenance to remain useful, so the more people who have access to the code, the better.

The argument that software and physical property, such as the stuff in your house, are equivalent will always fall on its face. At the end of the day, software is just a bunch of numbers. If I create a piece of software, I can burn it on to CDs any number of times for a few pence each or, even more cheaply, put it on a web server. The people I give the software to can pass it on further, until between us we've created millions of copies at negligible cost.


leader

The Soft Sell

Unfortunately, what's true for software binaries is not true for the atoms in physical property. I can't take my guitar and make thousands or millions of perfect copies of it using my computer. In fact, each copy I make could cost at least as much or more than the first one, assuming I actually had the skills required to copy a guitar. So while I'm happy to distribute free software, I wouldn't let just anyone borrow my guitar — and there's nothing illogical about that.

A problem is created when proprietary software companies try to have it both ways. On the one hand, they want to be able to use CD-ROM and Internet technologies to mass-produce software at a very high profit margin. On the other hand, they pretend to be upset when other people use exactly the same technologies to copy the software without paying for it.

Proprietary software companies moan about so-called 'piracy', but in truth they'd rather have an unpaid-for copy of their own software running on your machine than a paid-for copy from their competitor. It's how a particular application becomes a *de facto* standard. Every time someone passes a installer CD to their friend, they are participating in viral marketing for that company.

Hardware prices have fallen dramatically, but proprietary music software is as expensive as ever. I mean, £1450 for a single copy of *Nuendo*, with known bugs? I'd resent having to pay for bug-fix updates too. Is it any wonder that students and musicians are writing their own software?

Daniel James

Editor In Chief Paul White replies: *I don't think I ever intimated that people shouldn't be allowed to develop and distribute free software — I'm all for it where people have the time, patience and altruistic drive to do so. As to your observation that software is just a bunch of numbers, that's also perfectly true, but then a book is just a collection of letters and a top-of-the-range BMW is just a bunch of atoms. The hard work comes in arranging the numbers, letters or atoms in the right sequence to make a product, and whether that product is hardware or software, somebody has to use their time to create it. After all, atoms are pretty much as free as numbers and they've been lying around almost as long as time itself, so I really don't see the distinction.*

I know that mechanical products have to be manufactured rather than simply stamped onto a CD-ROM, but in most cases, the real cost is in the development of the product, the manufacturing plant and setting up the machinery. Once the production line is ready to run, what falls out of the other end often costs the manufacturer little more than the raw material cost, but they still have to charge a lot more to cover the setup costs. Software is much the same — the material cost of the CD-ROM, manual and so on may be negligible, but the software company's profit margins are still wafer-thin because they have to cover the costs of development.

As to the retail cost of proprietary software, I agree that some of it is expensive and some is rather more buggy than it ideally should be, but the choice as to whether or not to purchase it is entirely yours. If you can get by using free software written by musicians and students, then that's great.

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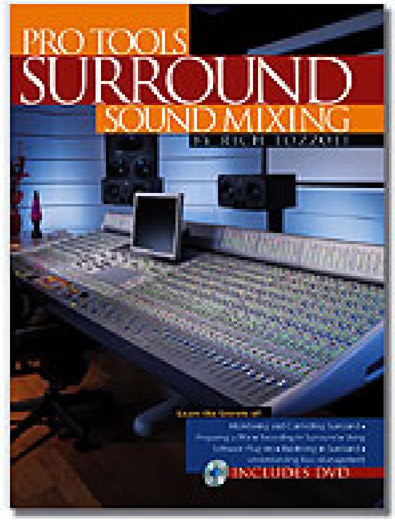
Rich Pickings



As a longtime *SOS* reader, I was happy to see my book, *Pro Tools Surround Sound Mixing* [*Backbeat Books/www. backbeatbooks.com*], mentioned in Mike Thornton's column [*see* Pro Tools *Notes,* SOS *June 2005*]. However, he said "The book is called *Pro Tools Surround Sound Mixing*, and is written by Rich Tozzoli, who describes himself as a 'surround sound guru'." Ouch! I certainly would never call myself that, as my experience is garnered simply from hard work, experimentation and making mistakes.

Rich Tozzoli

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More DPS24 Tips

I read with interest the letter from Rupert in Bath (surely a dangerous place to use electrical kit) concerning his problems getting signals to tracks on the Akai DPS24 [*see Q&A,* SOS *June 2005*]. It got an exellent and informative reply.

I think Rupert needs to do two things. Firstly he should try the 'tracks direct' patch presets, and more importantly, check out a really great forum — DPS World [http://dpsworld.vibestudio.net]. Here he can share the angst that is the natural result of any Akai manual, and get great advice from experienced DPS24 users. There's also a rather wonderful quick-start guide. The site is entirely independent, and populated by great people.

Greg Tempest

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In this article:

Jeff Wayne's Musical Version of The War of The Worlds

5.1 Surround Remix & Remastering Project

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People : Artists/Engineers/Producers/Programmers

Jeff Wayne's classic adaptation of *The War of The Worlds* was a technical *tour de force* and an enduring commercial success. A quarter of a century after its release, it has undergone a complete rework for 5.1 surround.

Sam Inglis

On June 9th, 1978, music journalists were invited to the London Planetarium by CBS Records for a lavish album launch. If they knew the name Jeff Wayne already, it was as the producer of teen idol David Essex; yet what they were presented with that day was a million miles from 'Rock On' or 'Hold Me Tight'. Here was a musical, but one that had never been seen on stage. Here were rock stars in abundance, but as well as singing, they were taking parts in a Victorian science fiction drama. And at a time when punk rock ruled, here was a 90-minute concept album fusing lengthy rock instrumentals, classical orchestration, a hint of disco, and the fruity narration of Richard Burton.

Photos: Richard Ecclestone

Jeff Wayne's Musical Version of The War of The Worlds, to give it its full title, might not have seemed like an obvious hit. Yet it remained in the British album charts for six years, with the best estimates available putting sales so far at a staggering 13 million. That figure is likely to rise even further this year, thanks to a lavish re-release incorporating both a new stereo mix and a 5.1 surround version. More than two years in the making, the reissue was put together at Wayne's Hertfordshire manor house by engineer Gaëtan Schurrer, and mixed by

Schurrer and Gary Langan, engineer and founder member of the Art Of Noise.

Jeff Wayne (left), Gary Langan (below) and Gaëtan Schurrer.

"Back in 1978, Jeff was pushing technology to the limit, and he's doing it again now," says Gary, and it's hard to argue with either claim. When it was made, *The War of The Worlds* was probably the first large-scale project ever to be recorded to 48 tracks, at least in the UK. "At the time I knew that I was doing a 48-track production, I was dead lucky that the studio I was working at [*Advision in London*] owned a distribution company, and had just taken delivery of this device called the Maglink, which was the first device that linked machines together," explains Jeff. "I knew that I was going to struggle with 24-track, and I didn't quite know how I was going to resolve that. I was doing something that no-one had yet done in production terms. It didn't mean it was going to be a better composition, or a better piece of work, but technically we were pushing the limits."

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Making Arrangements

By the mid-'70s, Jeff Wayne had his own small studio and was an accomplished synth programmer and engineer, but the music for *The War of The Worlds* was never demoed "in the sense of a record, where you'd do demos and see how it sounds". Nevertheless, he had undertaken a lengthy and meticulous process of pre-production. "It was scored out at the composition and orchestration stage like I would score a movie, to the fraction of a second, and our scriptwriter had to work to those timings. So not only did she have to be creative, but she had to be a technician to work to a given moment, because you may have songs, or you may have lead instrumental sections, or our journalist telling the story. When does that person enter or leave? So you have all these elements that had to be timed very precisely.

"I was trained to write out everything. Whether I work on an electronic score or an orchestral score or whatever, I compose at an instrument, and when it comes to the arrangement and orchestration, that I do away from any instrument. I find it's more free, but I'm from an era when orchestration was taught to me.

"There was a period of scriptwriting, then I did all my arrangements in readiness for the band sessions, which were the first tranche of recordings, and they were recorded as masters. Because some of these tracks were 10 or 12 minutes long, inevitably, what we played as a band had to be built up upon, and where there was a thematic piece, either I or one of the other musicians would play a guide melody on an instrument, just to show the band where the melodies were going to go. Where there was going to be singing I brought in people who were excellent session singers in their own right, and they sang the songs to give us an idea, and to present to the people who were the potential guest artists.

Jeff Wayne at Advision Studios during the recording of *The War of The Worlds*.

"Most of what we recorded in the band sessions ultimately became the roots of the entire master recording. It's only recently I've come to realise that most people didn't realise it was a live performance piece. It wasn't built upon one musician at a time playing to a click track, or anything like that. It started with a full band playing it live, and then, yes, I brought in my strings, but they were played on top of a live band performance. The guys who played on it were all pretty guv'nor musicians, and we could all sight-read, so no matter what you'd throw at them, they were there."

"The way that Jeff tackled it was that he'd have bass, drums, two guitars, a percussionist, Ken Freeman on synths, Jeff sitting at a piano, and a metronome," adds Gary. "And Jeff would have set the metronome and said 'We'll have a go at this tempo.' No clicks or anything. The amount of tempo changes, kids growing up today would cringe. The bpm's all over the place, but because you've got a cohesive bunch of people all moving together, it sounds enormous. And because the chemistry between those people, as somebody starts to pick the tempo up slightly as they move into another part of the arrangement, maybe it's Jeff who starts pushing them, because they're all working together, they'll all move together and it sounds cohesive."

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The Prof

Jeff Wayne, Gary Langan and Gaëtan Schurrer are all effusive in their praise for the musicians who played on *The War of The Worlds*. From Herbie Flowers's rock-solid bass to the guitar antics of Chris Spedding and Jo Partridge, the performances are all impressive, but the greatest praise is reserved for synth wizard Ken 'Prof' Freeman.

Ken 'Prof' Freeman (left, with ARP Odyssey), with guitarist Chris Spedding and Jeff Wayne.

"Ken Freeman was a complete genius," insists Langan, and when you discover exactly how much he contributed to *The War of The Worlds*, it's impossible to disagree. "There's a lot of orchestration that Jeff wrote for orchestral sounds — brass, horns, flutes, piccolos, oboes — and all of those are synthesizer sounds made by Ken," says Schurrer. "The only real orchestra on there is the strings, and some of the strings are actually string machines.

Assistant Ken Thomas sets up an ARP 2500 synth.

"He started mainly with a Minimoog and an ARP Odyssey, and then later on in the production when they were nearing completion he got a CS80, and they re-overdubbed a lot of the horns and things. I've actually got the Arturia CS80 plug-in, and tried to emulate some of those sounds that Ken came up with, and I fell well short. I think he had an understanding of real instruments and orchestration that gave him the edge as to how to make the synth respond like an instrument would."

"He could create drum sounds," adds Langan. "In 'The Artilleryman And The Fighting Machine', there's the sound of military drums. When you listen to the mixes from 1978 you'd probably swear that it was a military side drum and a military bass drum, but it's not — it's Ken doing jiggery-pokery on his synthesizers."

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Beyond Our Ken

The elaborate overdubs included Ken Freeman's multitracked synth parts, the voices of all the actors and singers, a live string section, solo instrumental parts and sound effects. On some of the recordings, Wayne and the band crammed almost 100 different parts onto the 48 tracks available to them.

"From a composing point of view, I was trying to create melodies, hooks and riffs, but from a sound point of view, as orchestrator and producer, I wanted to subtly create this feeling of a ping-pong match," explains Jeff. "When you're hearing the story through the eyes of humanity you hear this big, symphonic string section. When it's Martian and aggressive, it's the electric guitars, it's the synthesizers that create all the atmospheric sounds, and I deliberately kept true to that throughout the whole 95 minutes."

Percussionist Roy Jones with some of the instruments played on *The War of The Worlds*.

Creating the right sounds for the Martian themes required considerable ingenuity. Much of this, again, came from Ken Freeman. "He was able to take things that were scored out, and put a sound into them just by us talking," says Jeff. "He would take a look at the score, and separate from the notation, I might have a sound on it that says 'A snowflake'. What is a snowflake? Well, we had worked enough together to work out a shorthand where he'd start by trying to interpret it, and this was the era when you were dealing with sound waves — sawtooths, sine waves and square waves — and he would mix them together and create the sound from the synthesizers that he was playing. You couldn't store them in memory, they were monophonic so you had to build them up, and eventually we'd get to the point where we had something we were both very pleased with."

The only challenge Freeman couldn't meet was to interpret the voices of the Martians themselves, and their war cry of 'Ulla'. "The notation was there, the chords were there, but not the actual sound, which I wanted to be unique. It had to be gigantic when the Martians were terrifying the Earth, but at the very end when they're dying, the same vowels, 'ulla', had to sound like the Martians themselves were dying," says Jeff. "I asked Ken Freeman if he could devise a synthesizer that could produce the word 'Ulla' and play it from a keyboard. He was not only a brilliant musician, but a craftsman from the technical side. He came up with this little box that I recall, and it could go 'ooh', but it couldn't get the 'la', so it was just 'oo-ah'."

The box was jettisoned, and instead, guitarist Jo Partridge multitracked the 'Ullas' using a Peter Frampton-style talk box.

Elsewhere, *The War of The Worlds* was also heavily laden with foley effects, most of them created by Jeff Wayne's wife Geraldine. "In those days you had to go to sound effects libraries, tell them 'I'm looking for this', and they'd give you everything they had,' explains Jeff. "You'd try all the libraries, listen to them all, and pick the best of what you had, and very little manipulation could be done with them. The ones that wound up on the record, with one exception, were made. The one that always comes up is the sound of the cylinder unscrewing [*when the Martians emerge from their spaceship*]. The truth of is that it is me with two kitchen saucepans, turning them and scraping them together. Our engineer Jeff Young heard the idea, amplified what was on microphone, and there's this giant sound. He put it into stereo so it could really move around, because he knew the idea I was trying to create, and what the script said. There I am on acoustic saucepan!"

Although the lengthy pieces on *The War of The Worlds* were all performed in their entirety, there was also a lot of editing before, during and after the original mix. "They mixed it in sections," says Gary. "You couldn't mix it from one end to the other, it was impossible. They probably mixed 30 seconds, a minute, two minutes, whatever that bit required, and there would've been two or three of them. Which is how I started — as an assistant you were given the not important faders, and as you got better and better, if you were really good you got the drums."


David Essex played one of the major dramatic roles in Wayne's musical.


Jeff Wayne conducting the string section for *The War of The Worlds*.

"The original work that I recorded was just over two hours, so there was a lot of editing and restructuring to get it down to what it became," says Jeff. "It had to fit on four sides of black vinyl, but also to not bore everybody, because it was just too long at that length."

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The Eve Of The War

Jeff Wayne comes from a musical family, and the *War Of The Worlds* project owes much to his father Jerry. "My Dad kept saying 'You set out to be a composer who arranged and produced,' and I was on a long run of producing artists and doing shorter pieces, like film scores and TV themes and music for advertising. I said 'Let's have a go at finding a story that could, from a composing point of view, challenge me.' And that's really how it started.


Richard Burton recording his narration.

We read lots of different books, and it came down to a few that I remember — *The Day of The Triffids*, by John Wyndham, Aldous Huxley's *Brave New World*, and HG Wells's *The War of The Worlds*. But with *The War of The Worlds*, I can always remember, I could hear sound on one read-through. It wasn't a very long book, about 150 pages or so, and maybe that's why it was easy to hear. It turned out there was an agent who represented the estate of HG Wells, and we convinced the estate that we wanted to be true to the story in creating this musical interpretation, and we did a deal. That was 1975, and the whole writing, orchestration, scriptwriting, paintings, recording sessions, everything to do with it, took the better part of two and a half years, between early '75 and June '78.

Jerry Wayne, Jeff's father, was a well-known stage actor, and it was his encouragement that prompted Jeff to write *The War of The Worlds*.

"I still have the original book with all my underlinings and scribbles about things that motivated me to compose something or to chat to the guys that were the lyricists, or our scriptwriter. The truth of it was that I somehow thought I was going to do an instrumental album, no guest artists, no roles being played, virtually all instrumental, no paintings, no script, and a budget that was in one ballpark. And it turned out to be quite the opposite.

"By the time we started approaching the artists, the characters themselves had built up. Almost all of them really emanated from the HG Wells novel, so by the time I turned it into a musical piece, I knew of the type of vocal sound that I wanted. It was a British work, and I always wanted it to stay British in that sense."

The most important vocal part was that of the journalist who narrates the story. More in hope than expectation, the Waynes approached Richard Burton, who was at the time appearing in a play in New York. "People say 'How did you choose so and so?', but the truth is you don't choose people of that stature. You hope that they're going to be attracted to work on your project. That's how it came about with Richard. We did have a list of people on the expectation that Richard was going to say no, and we expected that somewhere down the bottom somebody would say yes. To our great surprise, Richard was attracted to it, and we were thrilled, because his voice was magnificent. He liked the script, liked the idea of it and said yes immediately."

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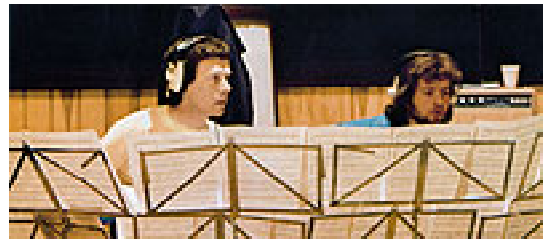
Forward In Time

A quarter of a century later, when Sony approached Jeff Wayne and his team with the idea of reissuing and remixing *The War of The Worlds*, the scale of the project quickly became apparent: they opened the tape archive to discover no fewer than 75 two-inch, 24-track masters and a staggering 372 quarter-inch masters. These had to be transferred to a digital format, before Gaëtan Schurrer could begin the painstaking process of identifying takes, sync'ing slave and master reels and reconstructing track sheets (see 'The Big Jigsaw' box). This, in turn, necessitated a comprehensive refit of Wayne's studio, involving the installation of a top-of-the-range Pro Tools HD rig and an M&K surround monitoring system (see 'High Density' box). Once the multitracks had been pieced together as *Pro Tools* Sessions, Gary Langan was brought in for the mixing stage. Langan and Schurrer's job was, first, to recreate the original stereo mix for release in high-quality SACD format, and second, to reinterpret Wayne's original vision in 5.1 surround.

"You've really got to understand the work of music that you're dealing with," explains Gary, "and the best way for me was to mix it in an environment that I knew and understood, and where it originally came from. So the mixing of the stereo was a learning curve for me. It was a way of getting myself into the album. A project like this is a bit like running a marathon — you've got to go through the pain barrier. It's an hour and a half long, you've got to be able to listen to it three or four times and be able to concentrate the whole time that you're listening to it. Mixing the stereo enabled me to go through that pain barrier, so when I got round to the real crux of why I was here — to mix the 5.1 — I knew every overdub, I knew how absolutely everything worked, how it was constructed, Jeff's thinking behind the overdubs, why this is this and that's that, which is so important.



With some songs exceeding 10 minutes in length, the musicians had a lot of sight-reading to do.

Drummer Barry Morgan at Advision.

"The stereo mix had to be a faithful reproduction of what they did in 1978, but using today's tools. So I limited myself to two reverb plates, one delay, a Harmonizer, a phaser, and that would have been about all they had during the '70s. Advision, I think, had two EMT plates, and you would have had one long and one short. You would have had a Revox or another stereo machine that would have given you delays by using the repro head and creating a delayed plate, and that would have been it."

Gary and Gaëtan were able to reproduce nearly all of these effects to their satisfaction using plug-ins within *Pro Tools*; likewise, the EQ that would originally have been applied at Advision's API console.

"We found a lot of the analogue hardware emulations in plug-ins, like all the Bomb Factory stuff, really useful," says Gaëtan. "When we first started working on Richard Burton's voice we made an effects chain using the original hardware Urei compressor and the Pultec programme EQ, and then we recreated the same thing in *Pro Tools* using the plug-ins, and it really was very close. There was just a little bit of extra 'oomph' using the hardware that wasn't there using the plug-ins, but we found that by putting a *Mic Modeler* plug-in on the voice before them, and using a little bit of that valve saturation parameter that it's got, it just gave it the oomph that was missing. That's how we went with most of it, but there were just a few synth sounds where we couldn't get the EQ that we wanted out of any of the plug-ins that we have — and we've got a lot of them. We couldn't get a certain sheen that we were looking for, and we ended up getting that from a little rack of original Amek EQs. We also used the old Dbx compressor on some of the bass lines; it just sounded right, so we used that and printed it. In both the stereo and 5.1 mixes we used an SSL compressor quite a lot as well. In the stereo, it was mainly over the whole mix, and in the 5.1 mixes we used it quite a bit as well, but mostly on the drum master — we'd have a group for all the drums, and we found the SSL compressor very useful for getting them to sit right in the mix. Other than that, it's all plug-ins."

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The Big Jigsaw

Before any thought could be given to creating a 5.1 mix of *The War of The Worlds*, the multitracks had to be transferred from two-inch tape and assembled in *Pro Tools*, a job which was, in itself, pretty daunting. "The multitracks were pretty well conserved, " says Gaëtan Schurrer. "They had been in secure archives for many years, and stored up here on Jeff's grounds. We decided to have them baked anyway, so FX Rentals baked them all and transferred them at 96k, 24-bit into a Pro Tools system. We ended up with eight Firewire drives, 120GB each, full of the multitracks.

"I needed to make decisions as to storage, because you can't chain eight Firewire drives and work from them — I think the limit is currently five. I worked out that about a Terabyte would be enough, so I bought three Lacie 320GB drives and chained them over Firewire, and then I started copying all the relevant takes to the Firewire drives I was going to use for the project. I actually split them into 16 tracks, so I would have tracks 1-16 of the master multitrack on the first drive, tracks 17-24 of the master multitrack on the second drive, tracks 1-8 of the slave multitrack also on the second drive, and tracks 9-24 of the slave on the third drive. So when I ran 48 tracks I knew that there were, at most, 16 tracks running from each drive. It's worked really well. We haven't had one problem to do with the disks not being able to cope.

"Greg Brooks, who's the *War Of The Worlds* archivist, listened through all 75 multitracks with me. We made track sheets, because there were none in existence any more, and from them we started deducing which we thought were the masters and slaves. Because it was a 48-track production, we had to find out which tapes were the masters and which tapes were the slaves, and which tapes were actually relevant to the double album and which weren't, because there was all sorts — there were out-takes, there were foreign-language versions, there was everything in those 75 multitracks.

"The slaves were actually quite easy to identify, because I only found one multitrack for each that was obviously the slave — it had the timecode on it, and the strings were only recorded on slaves, so I could always tell. With the masters, on the other hand, there were always two, three, four, five tapes that could be the master, different tapes that seemed to have all the parts. Some of them were copies, some of them were earlier, some of them were later, and it was trying to find the one. The only way to do this was to listen to every track, listen to every part and compare it to the original album. It was a big puzzle, and it took me a few months until I had the masters and the slaves for every track.

"We wanted to have a full Session in *Pro Tools* for each major piece, ready to mix. The album plays continuously, but it's 13 different tracks in terms of multitrack space. In the day, they'd only mixed in small sections — everything was mixed in two-minute sections and edited on quarter-inch and put together that way. What was on the multitrack was what had been written originally, but by the end everything got chopped up and edited, so on multitrack, the structure didn't always reflect what's on the album.

"Even before I could do the structural editing, I had to resync the multitracks together. They were sync'ed with Maglink code, which was the first timecode available. It wasn't very accurate, and there is no machine that exists today that can read the timecode. Jeff actually bought a Maglink machine at an auction for 50p, but we couldn't make it work. So having found the right master and the right slave, I had to put them in time, and invariably I found that the slave was always drifting later and later. Then I found that by looking at the timecode on the waveform display in *Pro Tools*, the timecode actually had a shape, which looked a bit like Morse code. It had all these little blips in it — there'd be one, then four, then two, then five — and you could actually match the tape between the master and the slave. So I created two groups of 24 tracks, one for the all the master tracks and one for all the slave tracks, then soloed the timecodes, found roughly the place where they started, lined them up together, and then by listening to it I could hear when it was going in and out of phase as it was going in and out of time. I went through every multitrack like this, doing a cut every three or four seconds so it would get in phase, then it would start to drift out, then I'd do a cut and move the slave back by about 100 to 150 samples — bearing in mind that we're working at 96kHz, so there's 96,000 samples a second. Then it would come back into phase, start to drift again, then I'd cut again — and I did that for the whole album.

"On the multitrack there was also a lot of stuff that never made it to the album, which they'd just muted at the time they were mixing. But I had everything there, and by listening to the original album and comparing to the multitrack, I had to decide what used to be in and what wasn't. Sometimes it was really obvious, but other times you can't tell, especially because they used to double and triple and quadruple-track a lot of stuff. A lot of the synth sounds, for example, were made with monosynths, so every time they wanted a big sound, they would play the same part again and again and track it five or six times, and then bounce that as the sound. Some of the tracks only had the one track of the bounce, and that was it, but on some of the others there'd be five or six tracks of it, and it would be like 'Did they use all five, or only two or three?' There was also one part in the third track, a sort of voice-box guitar, that was recorded right through the multitrack, yet listening to the album I found that it was only actually used twice — once for about 16 bars and towards the end for another four bars."

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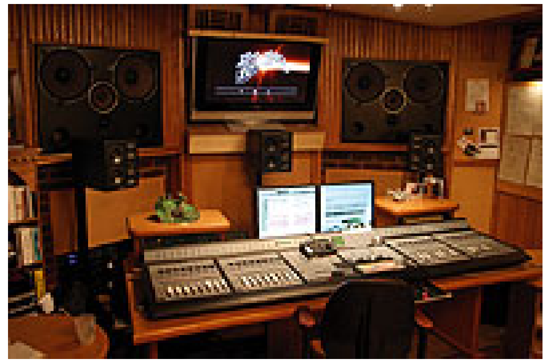
Spreading Out

The task of recreating the original stereo mix left little room for experiment, but the same was not true of the surround version. Many classic albums have had the surround treatment in recent years, but *The War of The Worlds* provided more options than most. This was partly because it had originally been recorded to 48 tracks rather than eight or 16, and also because of the dense, cinematic nature of the music itself. Gary Langan and Gaëtan Schurrer approached the job with one eye on the creative possibilities offered by the 5.1 medium, but without losing sight of the principles that had guided the original mix.



The studio at Jeff Wayne's Hertfordshire house where *The War of The Worlds* was remixed for surround, with the large Digidesign Pro Control setup that was used.

"One of Jeff's theories is that as one instrument leaves, the incoming one must be slightly more powerful, to take over from the one that's going," says Gary. "It didn't matter whether it was Richard's voice, a piece of narration that was leaving you and a guitar line that was coming in, or whether it was a vocal line that was going and a bass line that was coming in, they always had to take the same place, so that there was always this role of a lead going throughout the whole project. It was slightly more difficult [*in 5.1*] because I've got three times as much space, and although it's a great album to mix in the surround environment, there were occasions when there wasn't the support from the musical production.

"It was an over-produced stereo production, so when you listen to it in stereo there's always something going on, there's never one second where you're left alone. But then you spread that out into six speakers, and sometimes it's a bit of a desert, so I'd employ some tricks to make something *faux* 5.1 as if I'd recorded it in a 5.1 environment. Like the 'Ulla', which is the sound of the fighting machines — I turned that into what they call a *faux* 5.1 so it always comes at you from four speakers, but I've employed delays to give it a big spread. I've had to employ tricks like that where the reinforcement isn't quite there.

"I use mainly surround reverbs, but for special effects I might let something have its own discrete reverb, say something like the Heat Ray [*the Martian weapon, represented by Jo Partridge's distorted guitar*]. For the most part of the album, while they're conquering the Earth, they live in the back speakers with their own reverb and their own delay. And then, when the Martians have conquered the world, at that point, as a triumphant signal they all move to the front speakers, and again, they come with their own reverbs and their own delays, rather than sharing the 5.1 reverbs that might be going on the drums or the strings."

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High Density

The War of The Worlds was a groundbreaking recording project in 1978, and even in 2005, running a 48-track mix at 24-bit, 96kHz required cutting-edge equipment. "Our setup is actually the biggest setup you can get from Digidesign at the moment," explains Gaëtan Schurrer. "We've got a Pro Tools HD rig with seven HD Accel cards, which is the current limit. We already had Pro Control with three fader packs, so we got a G5 and a new expansion chassis, got an HD Core card and seven Accel cards, and added an Edit Pack to the Pro Control to do the 5.1. We got four 192 I/O interfaces and fitted them all with the extra eight analogue inputs, so we would have all the analogue I/O we needed for the outboard. So we've got a system with something like 64 analogue inputs and 32 analogue outputs, and a Sync I/O looks after the whole thing.

"In terms of plug-ins, we got pretty much everything that Digidesign distribute, we bought the Waves *Platinum Bundle* and *Surround Tools*, and we've also upgraded quite a few other plug-ins that we've had over the years, like all the Sound Toys stuff, which has always been one of our staples. The original *Timeblender* and *Pitchblender* was in every mix we've ever done here, because there's nothing else that really does what that does. Then they came up with these new ones called *Phase Mistress*, *Tremolator* and *Filter Freak*. They're unbelievable. The other thing that I've always really loved is *GRM Tools*, which do a bunch of things that no-one else does. I realised a bit too late that *GRM Tools* TDM only supported up to 48kHz, which was a big problem, and the only way it would support up to 192 was RTAS, and I really didn't want to use any RTAS plug-in if I could help it, because I didn't want to put any load onto the computer other than what was needed. We did use quite a bit of the GRM *Doppler*, which has a 5.1 version. It's really amazing — you can put a mono or stereo sound in it, and it'll spin it at weird rates around all six speakers, it's a great effect. We did actually manage to use it on a few of the songs, and then we were getting all these strange crashes. We were thinking 'What is going on?' and obviously we'd forgotten the fact that that thing was not compatible — yet it still worked, so what we ended up doing was we managed to print those great effects and then we just used them as an audio track and disabled the plug-ins and moved on.

"While we were doing the project we went from *Pro Tools* 6.1 to 6.4 to 6.7, and each time there were really incredible additions that helped us out. The *Revibe* plug-in appeared early in the project, and that's a fantastic reverb plug-in that we've used throughout. In the stereo mix we used mainly *Revibe*, and a bit of *Realverb* and a bit of *TC Reverb*, and a few *D-Verbs* as well, because that's still a great old favourite for certain effects. But in the 5.1 mix we found that it's been a combination of the Waves *360* reverb and *Revibe*. The combination of the two gave us a slight difference in flavour that we needed. The *Impact* compressor came out, which is very much like the SSL compressor, and we ended up using that over the master for all the 5.1 mixes. We used a lot of the Focusrite suite of plug-ins, and *Smack!* is a really good compressor.

"Another plug-in we really liked for many years was Metric Halo's *Channel Strip*, and we bought that because they brought it out for OS X, only to find that it wasn't Accel-compatible. Same thing with *Mic Modeler* from Antares, and *Auto-Tune*. Unfortunately we used *Mic Modeler* and *Channel Strip* quite a lot, especially in the stereo mix, and when we went to 5.1 we realised that it was nothing but problems because they would only instantiate on the HD Core card. We would find we had loads of DSP left, yet it was telling us that we didn't, and it was because those plug-ins couldn't go on the right DSP.

"The way we went from stereo to 5.1 was we used the stereo mix as a base, so we could use all our edits and our original plug-ins, and that's where we had trouble: once we'd started doing the reassignment of the outputs, it started using a lot more DSP. The 5.1 mixer uses a lot more DSP, and also, on the stereo mix I was using the long delay-compensation engine, which is 8200 samples per channel, and that obviously uses quite a lot of DSP too. We found that in 5.1 it just wasn't going to do it, so we reverted back to the short delay-compensation engine, and then our troubles were over. I really like the way the plug-in delay compensation is organised in *Pro Tools*, because you can manually bypass it per individual track. So on Richard Burton, we found we had to use the *Nonoise* plug-in because some of his takes were really noisy from all the original comping and recomping, and *Nonoise* introduces a huge latency which can't be compensated by the auto compensation. But it's so easy — you just disable the compensation for this track and shift the whole track by the right amount of samples, and there it is, back in time and denoised. I think that's one of the most efficient ways of implementing delay compensation that I've seen."

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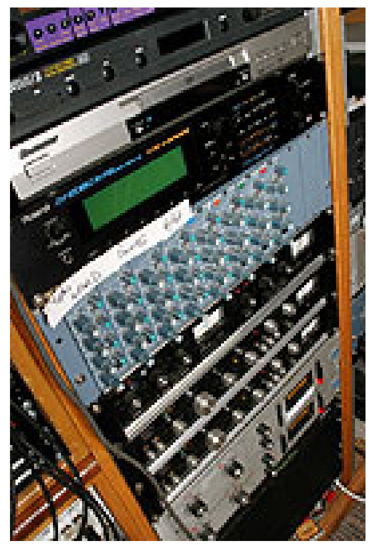
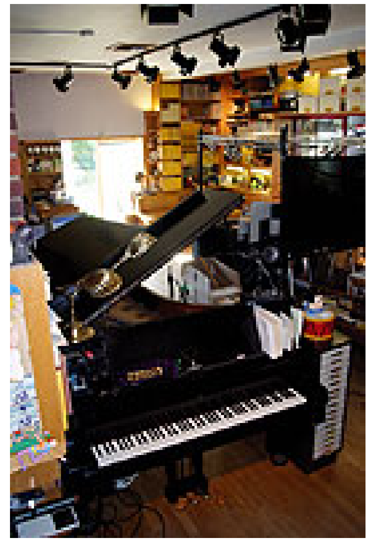
Journey To The Centre Of The Mix



Most of the original effects were recreated using plug-ins, but hardware Amek equalisers and Dbx compressors also played a role.

Langan has a few theories of his own, including a controversial view about the role of the rear speakers in surround mixing. "When you mix stereo, where do your drums and bass go? They sit in the middle. You don't put them to the left or the right. So when you're mixing 5.1, they go in the middle. I think this is a mistake that some people make when they're mixing 5.1. They think 'I've got the centre speaker, so I've got three speakers at the front and two at the back, so therefore I must have a front-to-back picture.' And I don't think you should at all. I think you should have a picture such that wherever you turn in that environment, you have the core instruments with you at all times — which is what happens in stereo. When you listen to a stereo mix, if you go over to the right-hand side, you might get more of the high backing vocal than on the left-hand side, but you're going to get the same amount of bass, drums and core rhythm instruments, because they've been mixed to the centre, so they're always going to travel with you. And that theory I think should be applied to 5.1. So in my mixing, the bass, the drums and anything that I call core rhythm instruments, driving instruments, support, stay in the centre of the quad. If I put the drums down in the front and you're standing at the back of the room it's going to sound awful for you, because you're not going to have any drums. They're going to be swamped by what's coming out of the rears. But if you put it all in the centre, it doesn't matter whether you go and stand at the back, it doesn't matter whether you go and hang out at the side."



The main work area in Jeff Wayne's house; the studio is to the left of picture.

His use of the word 'quad' is no accident, since the surround mix of *The War of The Worlds* is almost entirely a four-speaker affair. "In music production, I don't think you should use the centre channel. It's designed for cinema and it's designed to carry speech. The reason it's there is to help you hear the dialogue in a movie. It's so that the dubbing editor can turn up the dialogue over and above the underscore. Now that scenario doesn't exist in a record. OK, *The War of The Worlds* is slightly different because it's got a narration, but that aside, it's a work of music. When you set up your home theatre sound, you have the choice of turning up or down that centre speaker, and people do, because they can. So somebody might like the centre channel of their home system really loud. Now if I put Richard's voice solely on that centre channel, when you play the mix at home, because of how you like it, it's not how it should be. And also, it means that Richard Burton ends up being bare-arse naked, because if you switch off all the other channels, which you can do, you can just listen to the centre channel, and you can have Richard Burton or your singer or whoever, just there in the centre channel. He won't have any reverb because the reverb is on your phantom centre, so it just sounds awful.

"I think there's about 18 percent of Richard's vocal goes to the centre speaker, just so that there's something there, and then I put reverb there too, so that if you do happen to switch everything else off, what you're going to hear in my mix is a quiet bit of Richard with lots of reverb. It's mainly to keep the mastering people happy, because if I turn in the files and there's nothing on the centre channel, they'll 'phone up and say 'You haven't sent us all the information.'"

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The Martians Are Coming

The new surround version is indeed impressive, and the time put into its creation shows in the finished results. It was originally intended to celebrate the 25th anniversary of *The War of The Worlds*'s release, but the sheer scale of the project has but it back by two years. "We've created the first ever 27th anniversary edition," laughs Jeff. As it turned out, the timing has been perfect; through pure chance, the reissue will hit record shops just as an unrelated but massively hyped film of the HG Wells novel arrives in the cinemas. Wayne also has his own plans for a CGI animated movie, not to mention an ambitious theme-park attraction and a West End stage show. It seems we're not going to be able to get that infamous 'Da-da-daaaa!' out of our heads just yet... SOS

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The Wizard Of Oz

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People : Industry/Music Biz

Australian manufacturers Rode started off making their mics with imported Far Eastern components, but as *SOS*'s Editor In Chief found out on a visit to Sydney, these days their products are entirely conjured up in the land down under...

Paul White

Rode microphones are well known throughout the recording world, but a lot has changed since the company started out in business by teaming Chinese capsules with Australian electronics. Today, all Rode manufacturing takes place in Australia, the majority of it within Rode's own factory located in Sweetwater, a suburb of Sydney. Rode's founder and Managing Director Peter Freedman was kind enough to give me a detailed tour of his facility, and he told me something of the background to the company as we went.

SOS Editor In Chief Paul White and Peter Freedman of Rode checking out mics being tested at the Rode factory in Sydney.

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Rode Runner

"My father was an audio engineer; he designed products for companies in Sweden, and moved to Australia in 1966" explained Peter. "Electronics was always his hobby, so I was born into that environment, helping out with servicing and doing PA for bands."

During the 1980s, Peter picked up a sample of a cheap Chinese capacitor microphone, but didn't entertain the idea of marketing it until the home recording boom really took off in Australia on the back of the ADAT revolution in the early

1990s. He subsequently tested the mic and discovered that although the circuitry was noisy and unreliable, the capsule and the housing were basically OK. By designing a new circuit board for the microphone and using decent components, he was able to create what was then a very respectable microphone at a very low cost. At around this time, Peter met up with Englishman Colin Hill, who agreed to try to market the mic and they remember thinking that if they could sell 50 a month, they'd be on to a good thing! Peter and Colin took the mic around various outlets, including several in the USA, and sales were soon exceeding all of their expectations. "I managed to sell our new, improved model to a music store out there — and if you can sell stuff in America, you can sell it anywhere. At that time, we were buying the bodies and capsules from China and doing all our own electronics. We'd bring in capsules in bulk and about half would pass our quality control; the rest would be sent back.

"The NT1 we sold worldwide was made with a Chinese capsule, but we built the body and electronics in Australia. It was the same with the Classic mic — the body was entirely made in Australia, and that was the beginning of our metalworking manufacture with a partner company in Mudgee [*a town a few hours' drive from Sydney — Ed*]. We also looked at the way the capsules were made, and worked with our Chinese partner to improve the design. In this way, we built up quite a large line of microphones exclusive to us."

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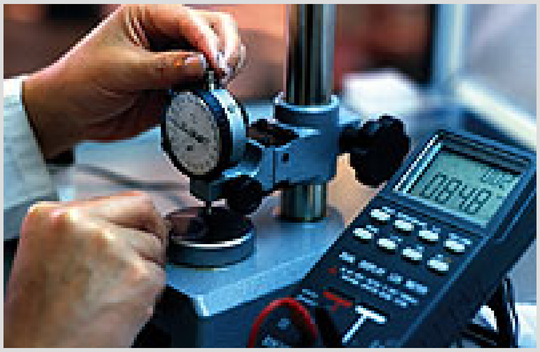
Rode Testing

As you would expect from a professional mic company, Rode mics and their constituent parts undergo extensive testing at various stages of their construction, as Peter Freedman explains. "We test all the way through trying to weed out any potential problems. While it's a costly thing to do in one way, it's cheap in another, as our failure rate is now so low. It starts with the raw materials — everything is tested and batch-tested for quality under the microscope. Because we have the test equipment, we can check components from different sources to find out what works best. If you don't have that facility and the knowledge, you have to go and buy guaranteed branded components, which may cost a lot more. If you buy only big-name parts when you don't need to, then obviously the price will be higher."



The capsule is next. According to Peter, since the company switched over from using Chinese capsules, the rejection rate has fallen from around 50 percent to less than three percent. Rather than test the capsules in a foam-lined box, Rode capsules are tested in free space using a window analyser, which takes a measurement before any reflections from the walls or floors can get back to the microphone. This, in effect, creates a virtual anechoic test environment down to around 150Hz, and Rode's engineers have found through research in university anechoic chambers that if the microphone capsule measures correctly above 150Hz, it will be in spec below that figure.

"Each capsule is tested before it goes on to a microphone, both acoustically and electronically. We test for sensitivity, and the 180-degree off-axis plot also tells you a lot, so we check that too. For the frequency response, we have a plus/minus limit outside which they fail, but because of the machining tolerances, failures are now very rare. We also do a 24-hour soak test, because the chances are that if a component is going to fail, it's going to fail within the first 24 hours."

As if that weren't enough, the finished mic is also tested. "When the tested PCBs and capsules are mated together, they are retested — but we also listen to every mic, because you can't always pick up rattles or crackles with test gear. Once you've done all that, everything that goes out is pretty good."

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Bringing It All Back Home

Within a few years, a number of companies discovered that they could buy in Chinese microphones to resell in Europe and the USA, and Peter had to raise his game to stay ahead. He soon realised that to improve the quality significantly above that of the competition, he'd have to bring the manufacturing to Australia.

Chinese microphones were (and still are) relatively cheap because of the low labour costs in that country, so to compete in Australia, where labour is considerably more costly, Peter needed to invest in expensive automated machinery that could make the necessary high-precision parts in high quantities, quickly and to a greater degree of accuracy than Chinese hand-made parts. "There is no book you can buy that will teach you how to make capsules, and nobody will show you. Instead, we had to spend a few million bucks on the machines, work from what we already knew, and from there it was trial and error. Fortunately, we had access to some very good physicists!

"I don't think we're talking about new capsules being so much better than they were in, say, an old U47 — some of those old mics sounded fantastic, but not all of them did. Now we have the consistency and the quality, but at a low cost.

There was little point in building similar products at a similar price to those being made in China, so Peter set out to built mid-priced microphones offering high-end specifications, and to date, his strategy seems to have paid off. "Anybody can make a good mic by spending a lot of money on the best parts. There's no magic in that, but if you can pull out an incredibly good mic at a low cost, that's something. That's what we do, and the only way to do that is to produce in volume."

The last Rode microphone to use a Chinese capsule was the NT2, recently discontinued in favour of the new NT2A, which is a completely different microphone rather than a revised design. Today, the only parts imported from China are the shockmounts for some of the cheaper microphones, and even these have new parts fitted to them to improve their performance.

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Rise Of The Machines

Rode have been in their current Sweetwater premises for around 18 months, and although it has been fully functional for some time, some new and highly advanced computerised metalworking machinery was arriving at the time of my visit. "When I first started looking at making our own capsules, I didn't have any machinery, and I got a quote back for $1000 for making just two backplates. Then I asked how much it would be for 50, and they told me it wouldn't really cost any more, because the cost is in setting up the machines. Back when microphones were produced in low qualities, it cost insane money to get good parts, but now we have a high-volume market, so the price can come down.

Diaphragm backplates following drilling.

"Now all the critical stuff is done with machines — the rest is just assembly. I don't want manufacturing by personal interpretation, where you may have one good capsule-assembly person; I want every mic to come out the same. That's why we use high-precision finishing machines and nine-axis computer-controlled metal lathes — CNC machines, as they're known — instead of hand labour. If I walk out onto the factory floor and I'm looking at two people working full time, every day, every year, just doing a process, I try to think of a way to do it with less labour. The final assembly of the one-inch capsules and the fitting of some large electronic components still have to be done by hand, but I'm looking at getting some through-hole insertion machines which will reduce the number of components that have to be hand-fitted."

The best way to find out how a Rode microphone goes together is to follow the production process, so Peter gave me a full tour of the facilities, although he declined to explain some of his new processes, as he didn't want to give away any of his ideas to his competitors!

"Today we have around 50 people in the company, with our own video and graphic design departments, and six people on full-time R&D. I'm involved in that as well; I come up with the concepts for the products, and the R&D guys come up with the way to do it. Initially, I don't think about price — I just look at whether we can make it work or not! Once it does, I have to think about whether we can sell it."

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Constructing A Capsule

The mic-manufacturing process starts with the mic capsules. The all-important diaphragm backplates (shown above) are machined and drilled using the CNC machines, and when you examine one of these parts, you see that there are around 50 precision-drilled holes that form part of the acoustic system of the microphone. According to Peter, some Asian microphone manufacturers still use manual labour to drill these holes by hand, which seems incredible given how tiny and precisely spaced these holes are. Next, a special insulating plastic is formed around the edge of the backplate in a high-pressure moulding machine, after which the backplate is 'lapped' to a precise thickness in a British machine that can process several backplates at one time. The lapping process essentially grinds the backplate between two flat surfaces using a fine abrasive paste to achieve the desired thickness and finish. A second, larger lapping machine was just being installed as I visited to cope with the increased demand.


Backplates on Rode's lapping machine after grinding.

Rode's production microphones typically use six-micron, gold-evaporated mylar diaphragms, but the company also has a sputtering room where prototype diaphragms or small runs can be manufactured. The gold-evaporated film is then tensioned before being fixed to rings prior to the capsule assembly, but Peter didn't want to show details of this part of the process, as Rode have found new ways to tension and age the mylar that improves both the precision and speed of manufacture. "As we have our own machines, I can try out these new ideas. Even though you know all the parameters, you might want to try a capsule using, for example, very thin mylar. Going to thinner mylar doesn't actually seem to make much of a difference, but you have to have the ability to try these things. I don't like to accept what I'm told at face value — I want to hear the difference for myself. That's why we have our own sputtering lab."

Capsule assembly takes place in a pressurised Clean Room (see overleaf). The room is sealed off from the outside world with HEPA (High Efficiency Particulate Air) filters, which let no particles larger than a third of a micron through, and are used in nuclear power stations to prevent radioactive contamination. Being rather larger than a third of a micron in size, I wasn't allowed to enter, but I was able to watch through the glass wall!

A typical Rode capsule has a 20-micron spacing between the diaphragm and the backplate, and even with the high maximum SPL these mics are designed to handle, the diaphragm moves less than one micron. The only way to get the diaphragm to hit the backplate is to subject the capsule to a strong gust of air, which is why it is imperative to use all capacitor microphones with a pop shield when recording vocals.

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Boards & Bodies

Capsule assembly in the Clean Room.

The automated component-placement machines used to manufacture the electronics boards for the mics.

While the capsules are being made and tested in one part of the factory, the microphone circuit boards are being built in another using high-speed automatic-placement surface-mounting machines where the components arrive on bandoliers that look like spools of cine film (see the picture on the next page). A flux paste is screened on to the boards, which also holds the components once they are placed, then the whole board passes through an oven to solder the joints. Each board actually comprises *multiple* identical mic preamps with slots milled between them to allow them to be separated easily after assembly. Custom microprocessor 'bed of nails' test jigs are used to test the performance of each board, including noise specs, prior to assembly. Afterwards, each mic is burnt in for 24 hours and then retested, including a critical listening test. And all this is just part of Rode's extensive testing regime (see the 'Rode Test' box on the previous page of this article).

According to Peter, the use of selected components, careful electronic design and surface-mount technology is what enables Rode to offer such impressively low noise levels, which he claims are 10dB or more better than those of their competitors in many cases. The use of surface-mount technology also eliminates a lot of wiring, and although some hand-wiring is still required to link the capsule to the circuit board, when you compare the inside of a Rode mic to that of a typical Asian microphone, there is indeed much less wiring, and the overall standard of assembly is also much neater. Apparently, much care is taken at the design stage to ensure that the metal parts of the microphone don't 'ring' and that the basket doesn't adversely affect the sound of the capsule within it. Many otherwise adequate microphones have been compromised by the housings used to protect them, which is why Rode tend to use solid metal mic bodies, either machined from solid metal or cast and then machined. Even the baskets are produced using custom presses that form both the inner and outer meshes in a single operation. The mesh is then fitted to channels in the rest of the metalwork using a high-performance adhesive.

"Some of the bodywork, such as for the NT5s, is made here on computer lathes, while the larger Classics and NT1s are made by the company in Mudgee. Some of the baskets are made here too, and I have some ideas for smaller mics that we'll also be able to manufacture here. With these new machines, you can do some amazing stuff because they can mill, drill, turn and pick up, all in one operation. A bar goes in at one end, and this incredible piece of engineering comes out of the other."

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Batch Production

Throughout my visit, Peter kept stressing the need to manufacture in quantity, so I was curious to find out how large the batches were that are run through production. After all, there aren't enough machines to make all the mic models all the time, so the production line must need to be switched from one model to another as required. He explained: "At the moment, we run two or three thousand of a particular model, after which we have to change the tools, set up the machines and change the CNC programs for the next model.

"However, for some of the products we're looking at, we envisage runs of 50,000 or even 100,000, because it's much more competitive to set up the machine once and then run it for three shifts. Do that for a couple of weeks and you're done for maybe a year's production on that model."

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Facing The Future

Now that Rode has a comprehensive range of studio microphones, the company is turning its attention to other areas. They now have a hand-held capacitor vocal mic for live use, and last year they developed a low-cost, high-performance short shotgun mic with integral shockmount for use with camcorders. Sales of this microphone have already exceeded expectations, so I'd be very surprised if we didn't see more live sound and audio/visual products joining the Rode line soon. Peter Freedman: "We haven't yet covered the live or broadcast markets fully, we haven't done installed sound, and I have a few more things up my sleeve on the recording and location sides. I also have some more ideas on stereo mics, but when it comes to large-diaphragm studio mics, I think we have the definitive line out there now."

Beyond the widening of the product line, there are also those new machines and continuing changes to the factory and production process. At a time when many companies are outsourcing more and more of their manufacturing processes to third parties, Rode seem to be going in the opposite direction, and Peter says this trend is likely to continue. "We will almost certainly bring the plastic injection-moulding in house — there are now some seriously sophisticated

NT2A mic bodies prior to assembly.

machines for just a few hundred thousand dollars. And we already have our own toolmaker, which saves a fortune in tooling costs. There are people who visit the factory and comment that it reminds them of the old vertically integrated setups of years ago. Some business people might think this is wrong. They think you should stick to what you're strong at; I think that you should have all your machinery under your control. But if the company was run by accountants, I'm sure it would look very different. We're enthusiasts — and in five years, we want to be world-beaters, the biggest in the industry." SOS

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SOUND ON SOUND
The World's Best Music Recording Magazine

SOS DVD 001

Paul White's Leader

Published in SOS August 2005

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People : Industry/Music Biz

The first-ever SOS DVD, distributed free with this issue, marks the start of a new chapter in the history of *Sound On Sound*, bringing the added dimensions of sight and sound to your favourite music monthly. The original brief we set ourselves when conceiving this disc was that the content and format had to augment and enhance the written word, not simply duplicate it in another medium. The DVD had to add real value to the magazine and cover disc package, and allow *SOS* to cover subject areas in ways that the printed magazine simply does not allow. But overall, we wanted readers to find the disc stimulating, entertaining and, most importantly, beneficial to their recording and music-making.

On our DVD-ROM, you will find over two and a half hours of video interviews, documentaries and tutorials, plus in excess of 150 minutes of music and audio examples. Significantly, over an hour of that music is made up of unsigned tracks from you, the readers, and this too was an important goal for *SOS* with this venture. There is nothing quite so motivating as knowing that your music will be heard to make you complete that lyric or finish that mix! Realistically, not every reader will win that elusive record contract or have the ability to drive thousands of listeners to their custom download site, but now there's a valuable promotional outlet for your music in the form of the SOS DVD. The best readers' tracks will be featured on each DVD we release and distribute to 43,000 potential listeners — greater exposure than you could realistically achieve from most Internet download sites and many radio stations! So send in your best tracks (see disc for details).

With each DVD, it is our intention to stimulate you into making music, and this month our Prodrive music-for-picture competition gives budding media composers a unique opportunity to break into this field. Watch the video clips on the DVD, listen to the client's brief, and deliver your music in time for the deadline, thereby experiencing the real-world pressure that professional media composers work with every day. You might win the trip of a lifetime in a Subaru world rally car, and you never know where such an opportunity might take your career...

You may well ask why we've waited so long before producing a cover disc. Quite simply we never felt a disc full of software and samples (however good) was a true representation of *Sound On Sound*'s editorial features or the breadth of equipment we review. Music software is only one aspect of the magazine's content, and we would be doing a disservice to the hardware reviews, workshop and tutorial articles if we'd simply delivered what most cover discs offer. So we waited for desktop video technology to mature, so that you'd be able to see *and* hear our content.

Now, with reasonably priced digital video cameras, powerful video-editing software, and today's generation of DVD-ROM equipped PCs and Macs, it is possible to record and play excellent-quality audio and video. This enables the SOS DVD to offer readers a truly interactive multimedia experience, seamlessly linking the content on the disc to current and previous articles on our popular web site, as well as those of the featured manufacturers. But that is only the beginning...

To help deliver a wide range of stimulating and informative content on disc, we've developed three custom-built media players — one for audio tracks, one for video, and the very useful SOS Photo Zoom player for static images. Check out the 18 multimedia product showcases on the DVD to experience the variety of ways in which these players combine to enrich your understanding of the products.

The *SOS* team has always striven to deliver a quality product. So rather than attempt to build a disc which performs across the widest range of computers, ie. for the lowest common platform, we have created a DVD-ROM which delivers high-quality interactive multimedia content for playback on computers running Windows XP and Mac OS X. Our readership survey shows that the majority of *SOS* readers are well equipped with powerful machines running the most recent OS, so we hope few of you will encounter any problems. We elected to use a web browser interface, as it is a familiar tool and provides the necessary flexibility for web integration, whilst having the advantage of making the DVD-ROM faster to use once images and text on the disc have been cached by your browser.

Finally, we'd really like to hear your thoughts about the disc, so join in the SOS Forum discussions or email dvd@soundonsound.com. We hope you will enjoy every minute, and look forward to our next disc in November with great anticipation.

Paul White *Editor In Chief*

Published in SOS August 2005

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SOUND ON SOUND
The World's Best Music Recording Magazine

Sounding Off

Big George

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People : Sounding Off

In music for film, the solution is to not be a problem.

Big George

It's been a while since I stuck my head above the parapet. In fact since I last graced the pages of this esteemed top-shelf periodical, I've found out exactly how boring being a TV presenter can be, discovered the joys of early mornings as an all-speech breakfast presenter for BBC local radio (www.bbc.co.uk/threecounties) and continued with my quest to provide 30 seconds of tosh for the front and back of a variety of TV programmes. I also composed a film score — short form, which was not so much 'low budget' as 'no budget', and managed to bypass the 'straight to video' stigma by never even making it to the duplication plant, hence saving me the indignity of being found out. And no, I'm not going to reveal the movie title!

Anyway, it got me thinking about how we, as producers and composers, go about making a buck from our art. The fact is, unless you're one of the lucky ones who is happy just to make music for the fun of it, or one of the confused who thinks their art transcends commerce, we're primarily service providers for an increasingly expanding (but ever more impoverished) industry. To that end, our main aim is to make the customer satisfied.

About The Author

Big George is currently doing anything and everything he can to avoid getting a proper job. He has had his mid-life crisis (namely a Gretsch Nashville semi-acoustic) and is looking forward to senility kicking in to relieve him from the tedium of giving a toss.

After years of ligging at showbiz Christmas parties, chance meetings in gear shops and a few days in court I've realised two truths about the music industry. Firstly, every successful operator reads *Sound On Sound* (from multi-millionaire lucky bastard to moderately successful genius) and secondly, cliché-ridden as it might sound, there really are only three steps to heaven.

1. Getting the job.

2. Doing the job.

3. Delivering the job.

Let's start in the centre of the action with 'doing the job'. This is where the vast majority of mistakes are made. The main one is thinking that composing and producing the music is the most important aspect of the procedure. It should be the easiest — in fact, it should take up hardly any of our nervous energy. Aren't we supposed to be adept at making music? Aren't there a zillion pieces of music out there for us to pillage and plunder? (What? You want to come up with something new, original, in your own unique style... you are joking aren't you?) Aren't we all blessed with facilities at home which 10 years ago would've cost more than a pound a minute to use?

The reality is that the opportunities are out there for everyone to get a credit — now! Although, to look at the prospects purely in terms of commerce, if you were a plumber the average rates for a broadcast job would buy between five and 25 hours of your time, in total. Yet I doubt there's anyone who wouldn't spend at least 100 hours on their half-minute masterpiece.

Now the hardest part, and something that needs careful, if not meticulous consideration: delivering the job. This, in truth, is the only part of the process that matters. If they like it, they'll use it, which in turn means they pay you, which will hopefully lead to you getting more work.

So deliver your best, but do not hand over the CD you've spent the last five weeks working on with miserable tales of woe about the sync'ing problems you overcame, the latency issues overdubbing the real tambourine caused or the latest saga in your broadband nightmare. They don't care!

They want you to be a problem they don't have any more. So when you deliver the job, be cool, be confident and make yourself believe that the hackneyed old nonsense you managed to cobble together at the last minute (and delivering last-minute is the best option — less time for panic changes) is the greatest slab of programme-defining audio ecstasy known to human kind. They'll love you for it, and if they do want changes, it's obvious you must have delivered too early, sucker!

Here's a quick tip if you want to get work in the broadcast industry — when asked, regardless of the question, the answer is always yes! You're also incredibly busy, but you can just about fit this particular job in. And this is by far the most important thing to remember — let them do all the talking and agree with every word they say, regardless of how stupid, irrelevant and ill-informed it may be. Until you have established a rapport, you must appear to take everything they tell you on board. Of course, as soon as you leave the office, you'll be free to go home and rewrite the same old riff you've been struggling to master for years.

Now then, having read this far, I think it's time you went and got yourself your own job. How? What? Where? Maybe next time!

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SOUND ON SOUND
The World's Best Music Recording Magazine

In this article:

Studio SOS

Brett Taylor-Homes

Published in SOS August 2005

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People : Studio SOS

Sorting out the monitoring in a studio with carpeted walls is the challenge for the *SOS* team this month.

Paul White

Brett Taylor-Holmes called *SOS* because he'd just finished building a studio in a single-floor extension to his Herefordshire home and had discovered that the monitoring acoustics weren't all he'd hoped for. He'd also bought a Mackie Control to go with his Apple *Logic*/G5 setup and had problems getting it to work. We arrived at Brett's rather nice period house near Malvern, which adjoins the local cricket green, and were served tea and Hobnobs while we asked more about Brett and his studio, which he wants to use mainly for composing film and TV music, as well as for video editing. Brett is currently taking a master's degree in film making and is also a performing drummer and drum teacher.

Paul constructs improvised bass traps by cutting an Auralex foam panel in half and then mounting small foam blocks to the halves to distance them from the surfaces to which they are to be fixed.

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Too Much Carpet!

As soon as we entered the studio, we could see some of the potential trouble spots. The monitors, which were Mackie HR824s, were set directly on top of a large, flat desk and were positioned well back in the corners where they were bound to generate strong reflections from the surface of the desk. Perhaps more seriously, Brett had used thin carpet to line the walls of his studio in the hope of providing a less reflective environment. In small areas carpet can be useful, but because it is so thin it only absorbs high frequencies with any degree of efficiency. Low and mid-range sound simply passes through it to the walls behind and then bounces back into the room, the result being a dry top end with a boxy mid-range.

He'd treated his live room in the same way with much the same result, and the door between the live room and control room was a lightweight interior door which didn't seal all the way around, so isolation between the two rooms wasn't great. This wasn't a problem when recording drums, as Brett uses his Roland V-Drum kit for that, but when recording vocals he'd had trouble with spill from the control-room monitors getting onto the recording. This was clearly going to take lots of Hobnobs!

Before looking at how to improve the acoustics, Hugh and I both felt the monitors should be brought much further forward to move them out of the room corners, bringing them in front of the computer monitor screens and helping to reduce the significance of desk reflections. We also positioned them on resilient pads to stop vibrations getting through into the desktop.

Listening to some pre-recorded Donald Fagen tracks from CD quickly confirmed that something needed to be done. Amongst the bits and pieces kindly provided by Auralex, we had a set of foam speaker pads which we used with their extra wedge pieces inserted to get the speakers roughly parallel to the desktop, as this gave the best tweeter height to match Brett's normal seating position. We brought the speakers forward to reduce the amount of desk area in front of them as well as to prevent diffraction from the LCD monitors. In doing so we had also moved them away from the corners, so we looked around the back of the HR824s to see what EQ switch settings had been used.

Brett had the monitors set up for 'quarter space' which was correct for working close to corners, but we switched them to 'half space' now that they were much further into the room. He'd also left the bass extension at maximum, which provided too much low end for such a small room — especially one with no real bass trapping. Switching to the middle 47Hz setting improved things, and listening to the same record produced a tighter, more even bass end, but with very muddled stereo imaging and still some residual lumpiness in the bass. We also noticed that the bass tended to disappear as you moved back from the listening position, something we've noticed in a lot of small rooms. Careful trapping can help, but in most cases you simply have to identify the rogue areas and keep away from them when mixing.

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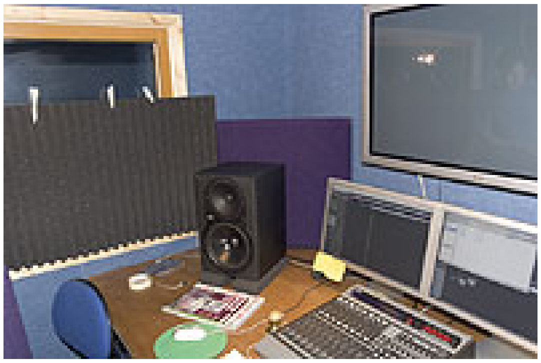
Improving The Stereo Imaging

To sort out the worst of the imaging problems, we used three panels of acoustic foam, one on either side of the listener at the 'mirror' points, and one on the ceiling. If you hold a mirror flat against the wall and mark where you can see the monitor from your normal listening position, that denotes where the centre of your acoustic foam panel needs to be. The trick works for placement of the ceiling panel too. Brett had placed a window through to the live room at the left-hand mirror point, and there was a large picture behind glass on the other, which was less than ideal! So after removing the picture, the right-wall and ceiling foam panels were fixed in place using spray adhesive, but the left-wall panel was temporarily fixed over the control room window using masking tape. Our suggestion for the long term was to glue this panel onto an MDF sheet and then hang it over the control-room window on hooks only when mixing. During tracking it could be left off to restore sight lines into the live room.


To get better response from Brett's Mackie HR824s, Paul & Hugh moved them away from the corners of the room, and placed them on some foam isolator pads. Acoustic foam was then placed on either side of monitoring position to reduce reflections which were muddling the stereo image — although on the left side the panel was only fixed temporarily so that it wouldn't obstruct the sight line into the live room while tracking.

The back of the room was another flat carpeted wall, so we decided to use one of the new Auralex four-inch panels that comes with thick foam spacing blocks which allow it to be used away from the wall to increase the low-frequency absorption. We put this right at the top of the rear wall, since you need to get the trapping into corners to have any impact on low end. Although we didn't have any corner bass traps with us, we told Brett that it wouldn't hurt to add some later — there was plenty of space to install something in the two rear corners of the room, which would be ideal. The panel kit we had used comes with a pair of polystyrene diffusers which we felt might also help if we fixed them to the rear wall below the foam to scatter some high end, rather than allowing the carpet to absorb it all. The idea was tested subjectively by temporarily propping them in place and conducting more listening tests. There was a small but worthwhile increase in liveliness which further countered the tendency of the room to sound boxy, so we decided to fix them permanently using the spray adhesive again. A word of warning here — polystyrene reacts very badly to some types of glue, so check the glue on a gash piece of polystyrene first to avoid having to watch your new diffusers melt away before your eyes!

We had one more four-inch foam panel left which we decided to cut in half and fix across the front corners of the room using the supplied spacing blocks, which happen to be cut at a 45-degree angle for exactly this purpose. We could have installed them directly behind the monitors or up against the ceiling, but we decided on the latter as it should make them more effective at low frequencies — and it also looked better.

While these little modifications weren't going to transform Brett's studio into a world-class monitoring facility, the subjective improvement was very significant. Now the stereo imaging was good and the room sounded much more neutral and far less boxy. It also looked good, which is important if you're spending long hours working in one place.

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Live-room Acoustics

Turning our attention to the live room, it became clear that the dividing wall was a simple stud partition, with a single layer of MDF on either side and no insulation between them at all. So sound isolation between the control and live rooms was very poor, and only a complete rebuild of the wall would help — which wasn't practical or necessary. We felt that the best approach would be to make the best of the situation by modifying the way Brett worked when tracking, and to improve the live room acoustics to help with vocal and acoustic-guitar recordings.

We suggested creating an area where vocals could be recorded effectively by fitting a curtain rail around one corner and then hanging a heavy drape or duvet on this. Our remaining piece of two-inch Auralex foam was stuck to the ceiling with the idea that the singer would stand under that with their back to the drapes to record. The rest of the room wasn't too crucial, as the V-Drums don't care about acoustics and any guitar amps would be close-miked. Acoustic guitars could be recorded satisfactorily in the vocal position. We didn't have the time or fittings to put up the curtain rail, but Brett said he could arrange that with no problem.

Another foam panel was placed on the rear wall of the studio, along with a polystyrene diffuser designed to combat the high-frequency absorption of the carpeted studio walls.

It turned out that Brett's monitoring difficulties arose when he fed the singer a headphone mix from his MOTU 828 MkII interface and then tried to juggle the headphone and speaker monitoring level using its front-panel controls. The MOTU 828 MkII isn't really designed to do this, and both Hugh and I felt that the only satisfactory solution would be to use a monitor controller that allowed the cue headphone and speaker levels to be controlled independently. Furthermore, some monitor controllers have two headphone outputs, each with its own level control, so the vocalist can have one feed and Brett the other while tracking. This would allow him to track vocals with the monitors turned right down, thus avoiding spill into the live room. An additional benefit with a monitor controller of this type is that there's a talkback facility built in so you don't have to yell through the control-room glass. Brett agreed that this approach made a lot of sense and put in an order for a Mackie Big Knob the same week.

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Troubleshooting Mackie Control

That left the Mackie Control to deal with. Brett's troubles with this had started when he bought two MIDI leads that were such a tight fit in the Mackie Control's sockets that he was almost afraid to pull them out again! I took a couple of MIDI cables along with me and they fitted fine, so I think he was probably unlucky and just happened to get a pair where the pins had been too heavily plated. Then again, this isn't the first time Brett has been unlucky with leads — when he bought his MOTU 828 MkII he couldn't get that going for ages, and the culprit turned out to be a faulty Firewire cable!

Once we'd switched on the Mackie Control, it was clear that it hadn't yet been set up to work as a Logic Control, but as I was unfamiliar with the unit I decided to break the habit of a lifetime and look in the manual. This showed which two buttons to hold down while turning the unit on to get it into the mode where you could choose what controller you wanted it to emulate. Pressing down the selection buttons on the first two channels entered the emulation mode, and then pressing the V-Pot on channel eight selected the Logic Control option, after which I configured it in *Logic*'s Preferences.

The final remaining foam panel was placed on the ceiling in the live room, above where Brett was planning to set up his mic for vocal recording.

All seemed well, with the transport controls and faders behaving as they should, but a few more tests showed that we still had problems. The Jog wheel wouldn't jog and the pan controls would only pan to the extreme left or right with nothing in between. No amount of resetting or trashing *Logic*'s Preferences would fix it, so I came to the conclusion that the most likely cause was a hardware or firmware fault, and the only way to prove that was with a replacement Mackie Control.

As Brett was a relative newcomer to *Logic*, I took some time to set up a default song for him with 24 audio tracks and 16 instrument tracks, which I thought would be more than enough for most of his projects. I showed him how to create screensets, then explained that the best way to deploy reverb plug-ins is to use them in busses, fed from post-fade aux sends, as that way all the channels can have access to the same effects.

If you simply drop a reverb plug-in into the insert point of every channel that needs reverb, you'll soon run out of CPU power, especially if you're using a power-hungry convolution reverb such as *Space Designer*. For most mixes two different reverbs will provide all the variety that's needed. Hugh and I also went through the basics of compression, as this is one area that a lot of people have trouble with.

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Summing Up

By the middle of the afternoon the studio was sounding much more neutral, Brett had a good idea what needed adding to his gear list and, aside from the Mackie Control problems, his system was up and running, requiring only a suitable duvet before he could begin making first-rate vocal recordings. Once again, it is demonstrable that, while it may look very nice, the overuse of carpet as an acoustic treatment can really mess up your room acoustics unless balanced by some high-frequency reflection and low-frequency trapping. Given the choice, I wouldn't have used much if any in this room, but as we were faced with a finished job we had to incorporate what was already there into the final design.

The Auralex panels mounted on blocks really seemed to do the trick in sorting out the lower mid-range, though Brett's room would benefit from more bass trapping, and a large void under his desk area was still acting as a resonator to some extent, as it was an empty space enclosed on all sides but one. Filling this with mineral wool, foam, or even old bedding would help damp it down and provide low-cost bass trapping. SOS

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Current Versions

Mac OS X: Apple *Logic Pro* v7.1

Mac OS 9L Emagic *Logic Pro* v6.4.2

PC: Emagic *Logic Audio Platinum* 5.5.1

Logic v7.1 Stability Issues

Logic Notes

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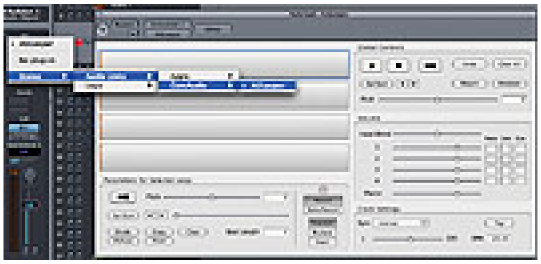
Print article : Close window

Technique : Logic Notes

We take a look at some of the stability problems still lurking in v7.1 under Tiger, and also reveal a couple of extra plug-ins tucked away in the new OS upgrade.

Stephen Bennett

Apple's *Logic v*7.1 is proving to be much more stable on my G5 and G4 Powerbook than its predecessor. I've finally managed to complete several whole sessions without a single crash. It does, however, still seem quite susceptible to problems with third-party Audio Units plug-ins — even those that pass the AU validation test. I've managed to make *Logic* fall over a few times when using the latest version of Waves' Waveshell and the *Logic* 7.1-compatible *Ivory* piano virtual instrument update. Unfortunately, the particular sequence of events causing these crashes doesn't seem to be easily reproducible.



The new *AULooper* plug-in can be found in the Tiger upgrade's Developer Tools.

The various *Logic* Internet forums are still quite full of reports of instability — albeit on a lesser scale than for version 7. On the plus side, Apple have re-introduced the helpful '*Logic* has unexpectedly' quit dialogue and updated it with a friendly Reload Program button. I've also discovered that *Logic* v7.1 now places a 'recovery' file in the Song folder. In a couple of cases I've managed to recover the Song right up to the point where *Logic* went down. In spite of this help, it's always sensible to save often. *Logic* has a feature whereby you can save up to 100 backups of the current song — in version 7.x it's set in the Global Preferences, on the Song Handling page. Using this, you can save often and be sure you haven't overwritten anything useful. Even with this feature, though, *Logic* sorely needs an 'automatic save' function.

I've still to install Tiger on my G5, as the current drivers for my UAD1 DSP card don't run under 10.4.1. I have, however, been using version 7.1 on my Powerbook. Here it seems extremely stable, though on this system I have no third-party AU plug-ins installed. There are a few extra Tiger-only features available to *Logic*, one of these being the *AUPitch* plug-in. This appears to be a plug-in version of *Logic*'s new time-stretching algorithms, and it's a further indication of how Apple are slowly building features directly into the OS. *AUPitch* is a pretty smooth-sounding plug-in, and I've been using it successfully to tighten up some fretless bass pitch discrepancies. Doing this on small sections of audio reminded me that *Logic* really could do with a facility to process regions with plug-ins, rather than whole tracks — and destructively if need be. In my example above, I have to create a region by cutting out the offending bass note, copying it to another track, processing the track with *AUPitch*, bouncing down the processed track, re-importing it, and finally inserting it back to its original position. A 'process region' feature could do all this in one mouse-click.

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Logic Tip

The project management introduced in *Logic* v7 makes it easier to keep track of Songs and their associated files. In earlier versions, saving a Song wouldn't automatically set the audio-recording file location to the Song folder. This often resulted in audio files being misplaced when backing up. Saving a *Logic* file as a Project rather than as a Song has several advantages. It places all the audio files in a subfolder of the *Logic* Song folder. You can choose to save other associated files into the same location; these include *Space Designer* impulse files and *EXS24* and *Ultrabeat* samples. You may want to do this if you're sending the Song to another studio or archiving a Song for backup and want to be sure these are available in the future. You should get into the habit of saving all new Songs as Projects — once done, you can just use the Save Key Command (Apple-S by default) to save.

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AULooper Plug-in

One undocumented surprise is the appearance of an *AULooper* plug-in which is only available if you install the Developer Tools from the Tiger install disc. You can then copy it to your Library/Audio/Plug-ins/Components folder where it will then be available along with other AU plug-ins. There are a whole bunch of musicians who use looping programs and hardware echo units, such as the Echoplex, to build up complex layers of sound in real time — often in live situations. This is quite different from taking a pre-recorded snippet of audio and looping it. It's about playing phrases, letting them loop and adding new ones on top — a modern 'sound on sound' technique, if you like. You could see at it as the 21st-century equivalent of Robert Fripp's 'Frippertronic' experiments in the 1980s. The Apple *AULooper* seems to have a pretty comprehensive range of features and, as one of the ways I use my laptop is for live looping, I'm keen to put it through its paces. If you're interested in the wonderful world of looping, check out the looping community at www.loopers-delight.com.

Speaking of Tiger, there's a Pro Application support update available at www.apple.com/support/downloads/proapplicationsupport30macosx104.html. It's an 'under the hood' update and is required for all of Apple's Pro applications, including *Logic*. It seems to have cured a few reported *Logic* booting problems under Tiger, but is otherwise pretty transparent to the user. SOS

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SOUND ON SOUND
The World's Best Music Recording Magazine

In this article:

CLASSIC TRACKS: The Bee Gees Stayin' Alive

Producers: The Bee Gees, Albhy Galuten, Karl Richardson

Published in SOS August 2005

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Technique : Recording/Mixing

Disco was an American phenomenon, but its greatest hits were recorded in France by an English band who were trying to play R&B...

Richard Buskin

Years after the '70s disco fad and subsequent backlash had subsided, Maurice Gibb told an interviewer that he'd like to dress up the *Saturday Night Fever* album in a white suit and gold medallion and set the whole thing on fire, such was the stigma that had been attached to him and his brothers Barry and Robin by press and public alike. One minute, they were the purveyors of 'blue-eyed soul', melding their pop roots, trademark harmonies and Barry's newly discovered falsetto with their love of early '70s Philadelphia funk, crafting heavily rhythmic dance music that was finding its way onto black American radio stations. The next, thanks to a soundtrack album that sold a then-record 25 million copies worldwide and topped the US charts for 24 weeks — where it spawned four number one singles, three of them their own — they were the Kings of Disco and all that encompassed, reaping the rewards and then the brickbats.

Still, as Barry later asserted, it did put food on the table, while the *Saturday Night Fever* album was a significant moment in the annals of pop culture; a moment when a trio of white Englishmen almost single-handely ignited a widespread mania for the disco music that had previously been the domain of the black and gay sub-cultures in America, and had been superseded by punk in Europe. In addition to the Bee Gees' recordings of 'Stayin' Alive', 'How Deep Is Your Love', 'Night Fever', 'More Than A Woman', 'Jive Talkin' and 'You Should Be Dancing', the two-LP set contained their compositions being covered by the Tavares ('More Than A Woman') and Yvonne Elliman ('If I Can't Have You'), alongside lesser material by the likes of Walter Murphy, David Shire, Ralph MacDonald, MFSB, the Trammps, Kool & the Gang, and KC & the Sunshine Band. Yet it is 'Stayin' Alive', which played over the movie's opening credits while John Travolta's Tony Manero strutted down the New York streets in his polyester suit, that best evokes the era and its promotion of sex, drugs and breathless boogying as some form of decadent compensation for a humdrum daily existence.

Photo courtesy of Dick Ashby

The Bee Gees: from left, Maurice, Robin and Barry Gibb.

It is therefore interesting that this track, along with 'How Deep Is Your Love' and 'More Than A Woman', was originally penned for a completely different project: the follow-up to the *Main Course* and *Children Of The World* albums that had seen the Bee Gees reinvent and reestablish themselves as a chart force, thanks to hits such as 'Jive Talkin', 'Nights On Broadway', 'You Should Be Dancing' and 'Love So Right'. And there to assist them as a co-producer and engineer was Karl Richardson, a man whose impressive track record is one of the industry's better kept secrets.

A Miami native, Richardson began his studio career there in 1969 at Criteria, mastering a wide variety of mixes and popular hits of the day, not least because Atlantic Records block-booked Studio B for an entire year. So it was that he cut his musical teeth listening to Aretha Franklin, Dr. John, the Young Rascals, Brook Benton and whoever else was on the Atlantic roster at that time, as produced by Jerry Wexler, Tom Dowd and Arif Mardin. It was quite an education, and soon Richardson also found himself assisting as a recording engineer on albums by Eric Clapton, the Average White Band and Lynyrd Skynyrd.

"It was a lot of ear training," he says. "We would try loads of different things with a lot of different microphones before using them on a client. Ron Albert and Chuck Kirkpatrick were already staff engineers at Criteria alongside [*founder*] Mack [*Emerman*], and I was the third guy."

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Voices From On High

It was while recording 'Nights On Broadway' for the *Main Course* album that Barry Gibb had discovered his falsetto voice, urged on by Arif Mardin who was looking for a chorus part to spark the song. "He wanted somebody singing up high or maybe screaming," Barry would later tell *TV Guide*. "So, I went out in the studio, and I found that not only could I scream in tune, I could sing a whole song in falsetto."

Thereafter, this piercing sound would become his — and hence the group's — trademark, along with the silver *lamé* shirts and toothsome smiles. Much imitated, often derided, it was undeniably a remarkable vocal attribute, and one that served the Bee Gees well on the *Saturday Night Fever* album and subsequent projects.

"I miked him with a U87, and I just remember at that point Barry was so confident in his falsetto, the sound of his falsetto, because he had more control over it," says Richardson. "He had great mic technique — he's as good as Michael Jackson any day of the week, and I've overdubbed both of them — and although Barry was always a little shy about his tenor voice, he had this command of the falsetto."

Chateau d'Herouville studio assistant Michel Marie recalls being amazed by the way the Bee Gees recorded their harmonies. "When I learnt they would be three to sing, I set up three different headphones and three different mics. They said 'No, no!' and asked me for a single mic, and no headphones — just a little speaker close to them. They sang together around the same mic, looking at each other. And when I heard them, I knew what 'good singing' meant : even on the first take, they were perfect, in tune, in the rhythm... I was not used to that from French singers!"

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Hands Too Full

Criteria had a rich history of technological innovation. Opening as a three-track facility in 1958, it boasted an eight-channel console as early as 1964 and four-track recording a couple of years later. Studio B was constructed in 1967, Studio C was added in 1972, and D and E were built towards the end of that decade, by which time everything was 24-track. Richardson, in the meantime, became Criteria's Senior Engineer and ran its disc mastering plant *en route* to amassing credits that include Barbra Streisand, Diana Ross, Dionne Warwick, Lee Ritenour, Peabo Bryson, Michael Jackson, Andy Gibb and, of course, the Bee Gees. It's with the brothers Gibb that he's enjoyed his greatest success, earning nine Grammy nominations and winning two: Producer of the Year and Album of the Year for *Saturday Night Fever*.

With Karl Richardson behind the board, Arif Mardin had produced 1975's *Main Course* album, but Mardin was out of the picture once Bee Gees business manager Robert Stigwood ended his RSO label's distribution deal with Atlantic and went over to Polygram. Thereafter, the band, bolstered by Alan Kendall on lead guitar, Blue Weaver on keyboards and Dennis Byron on drums, ventured out to California for the *Children Of The World* project, and when things didn't work out with a new producer they turned to Richardson and asked him to collaborate with them back at Criteria.

Photos courtesy of Dick Ashby

After the success of the *Saturday Night Fever* soundtrack, the Bee Gees built their own Middle Ear Studios in Miami, where this shot was taken.

"While I did all of the engineering, I wasn't really up to snuff in terms of my musical knowledge and how to communicate on that level," he admits. "My hands were too full, so I called my friend [*keyboard player and producer*] Albhy Galuten, who was in England at the time, working on an album for Dick James Music, and I said 'Albhy, listen: I'm sitting here, alone in the studio with the Bee Gees, and I'm trying to tell the drummer to not be so busy on the hi-hat, but he wants to know about eighth notes.' Albhy said 'Well, that's an amazing coincidence, because I've just listened to the final playback of the album I'm working on. I'll be on the next flight.' That led to the beginning of Karlbhy Productions and our association with Barry.

"Albhy had gone to the Berkeley School of Music, and his input about chord changes and so on would be invaluable — he was the only one of the team who could write string lines. Barry could sing them,

Photo: Franck Ernould

The Chateau d'Herouville as it looks today. This is the left wing.

but Albhy could play piano and Albhy could play guitar, so he knew what would and wouldn't work. I, meanwhile, would take care of the engineering and I'd constantly be challenged by the guys to 'Make it sound better, Karl.' What's 'better' mean? The console doesn't have a 'better' knob. Still, the combination of Barry, Albhy and I worked in the control room, and the rest, as they say, is history."

This history would, in addition to the Bee Gees' *Children Of The World*, *Here At Last... Bee Gees... Live*, *Saturday Night Fever*, *Spirits Having Flown* and *Living Eyes* albums, also take in Andy Gibb's *Flowing Rivers*, *Shadow Dancing* and *After Dark*, Barbra Streisand's *Guilty*, Dionne Warwick's *Heartbreaker*, and the 1983 soundtrack to *Fever* sequel *Staying Alive*. All featured hugely successful Gibb compositions, yet back in 1977 the guys felt under pressure to deliver the goods for their follow-up to *Main Course* and its three hit singles.

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Funky Chateau

"For tax purposes they needed to make records outside the UK and the United States," explains Richardson. "So, we were looking at Canada and their [*personal*] manager Dick Ashby was looking at other places, and Dick eventually booked a month of studio time at Chateau d'Herouville, where Elton John had recorded *Honky Chateau*. It was in a funky little village about an hour and a half north of Paris, and it was in a funky state, with a 24-track Studer tape machine, serial number 001 — the thing was barely on its last legs. In fact, after Elton John left there, the owner of the studio had smashed it to smithereens and rebuilt what was left of it in a second-storey loft within this castle. Well, it was ungrounded, and when I arrived there was a terrific buzz on everything, so I spent the first two days grounding the place.

"The console was an old API 550 on which someone had coloured all the EQ knobs with nail polish, and the other fun thing about it was that the faders had worn contacts in them, and there were holes where the audio would completely drop out. So, I had to mark all the faders with tape: 'Don't move across this line if there's any audio.' Even the monitors were horrible. We used headphones and a little set of Auratones."

Photo: Franck Ernould

The right wing of the Chateau.

All of which begs the question: Why stay there?

"Management had already committed us," comes the reply. "Of course, we showed up there in the winter, and there was an outside window behind us in the horizontal control room, so we'd put the Heineken bottles there to keep them cold. It was great, and another fun thing about the expedition was the fact that, while some of the guys brought their wives, there were only two showers on the premises that worked. This meant everybody had to line up for the bathroom, and the result was that we'd probably start recording at about two in the afternoon and stay up until about three or four in the morning. It was really stupid, but that was the only way we could get all of the showers done in the morning."

The control room overlooked a live area whose ceiling had less-than-ideal wooden beams, and it was there that Karl Richardson recorded the four-piece setup of Maurice on bass, Dennis Byron on drums, Blue Weaver on keyboards and Barry on acoustic guitar.

"Afterwards, Maurice might do a couple of touch-ups on the bass," he recalls, "Alan Kendall would overdub a guitar track and Barry would make me double up the acoustic."

Richardson can no longer recall exactly how the drums were miked, but he does know they were condensed to four tracks — bass, snare and a stereo track for everything else — because this was his preferred method for many years. And he also has a clear recollection of how he first came to hear 'Stayin' Alive': "I distinctly remember Barry saying 'Boy, Karl, have I got a song for you,' and sitting down to play 'Stayin' Alive' on an acoustic guitar. It was like a chant and it was unbelievable. I said 'Barry, don't forget that rhythm. That's a number one record.' I knew, five bars in, no questions asked. You couldn't get past the intro without knowing it was a smash."

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The Sound Of Disco

Together, the Bee Gees, Karl Richardson and Albhy Galuten created what most of us think of as the sound of disco, but it was not their intention to invent a new style. "We never knew about disco and we didn't think about disco. We thought we were making rhythm and blues records. It was all about R&B. We loved all that stuff, we just couldn't figure out how to do it! 'Oh man, that sounds great, but it sounds like a room full of studio musicians.' 'Yeah, well...' The sound we came up with was therefore our sound, and for the *Saturday Night Fever* album we recorded all of the tracks in France, and overdubbed and mixed in Miami, while overdubbing in LA too. 'More Than A Woman' was given to the Tavares brothers, and their track came in sub-standard so we were asked to go out to LA and finish the Bee Gees' own demo. That's what we did, adding congas and strings before mixing at Criteria.

"'Night Fever' was cut live in France. Maurice was playing DI'd bass with his pick, Dennis Byron was playing drums, Blue Weaver was playing keyboards, Alan Kendall was playing rhythm guitar, and Barry was playing rhythm and singing the pilot vocal. The drums were the only thing retained from this live track — it was a complete take, not comped — and all the other parts were overdubbed, like the keyboard part that was carefully crafted. I mean, many parts weren't there from the start. Blue, Barry, Albhy and I would sit down and say 'That chord sounds great there, but how about when the guitar player goes "dang, wa-tang"? Do you want the seventh in the chord or do you want to leave that hole there?' Those were the kinds of things that had to be worked out.

"It was all very orchestrated. It was a process and it was all about 'head charts'; creating in the studio. You know 'Gee, OK, that's the part of the verse for the keyboards.' Then we would go for the performance. All of the arrangements were done on the spot and then the performance was executed until it felt good. That was the standard. It didn't matter how we got there — whether something was thrown together or it was one take — our concern was that it felt good, that it made a statement. How it's done, I don't know. I mean, how do you make a Mercedes-Benz? Do you start with the tyres? All I know is the end product. If that's accepted, then how it came to be is just detail.

"We had no guidelines. The only rule was there were no rules, so we could do anything. It didn't matter if it was a bass drum or a synthesizer sound — we would talk about it and say 'Well, why don't we do this?' And I can't recall anything specific because this took place almost on every song. Plus the fact that everything was at least second-generation, most of it third. On the next album, *Spirits Having Flown*, we discovered 48-track, so everything at that point was multitracked, Dolby, bounced, bounced, bounced, bounced, bounced, whereas the other stuff was 24-track. However, to get it to 24 a lot of it was hand-sync'ed and we'd overdub forever. Again, those sounds were probably limited to what we had available at Criteria in terms of reverb chambers, processing, MCI consoles. Who knows?

"I do know this: 'Night Fever' is the rough mix. We mixed that song in 10 minutes. We had overdubbed all these synthesizer pads, extra guitar notes, little percussion instruments and so on, and we kept mixing it again and again and again, and then finally we played the rough mix and everybody said it felt better. You see, it was all about feel at that time. It wasn't about trying to impress people. And that was the key to the music. As a matter of fact, we had a demo of 'How Deep Is Your Love' from France, with the brothers singing, Blue playing keyboards and Mo playing bass, and right up until the final mix we would play that rough mix from France to use as a guide, because the feel was everything to us. "

One thing that distinguished the Bee Gees from traditional R&B was their characteristic rhythms. "A lot of that was Barry's right hand," Richardson says. "I mean, every one of those records has some form of acoustic guitar with Barry going ching-ching-ching. Whether it's hidden or not, it's there, driving the track along."

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A Low-Budget Movie

Barry and Robin had written the song while they were staying at Robert Stigwood's house in Bermuda, and it was shortly after the composition of 'How Deep Is Your Love' and 'More Than A Woman' at Chateau d'Herouville that Stigwood called to inform his boys that these songs weren't for a studio album after all. Instead, they would form part of the soundtrack to a low-budget movie that he was producing, *Tribal Rites Of The New Saturday Night*, based on a New York magazine article penned by British journalist Nik Cohn.

'Stayin' Alive' was tracked live in the studio, vocals were added and a rough mix was done, but nobody was overly impressed with the end result. "It didn't sound steady enough," Richardson recalls. As a result, 'Night Fever' was quickly penned and recorded with a view to it being the film's banner waver and accompanying album's first single. "Everbody was real happy with the way 'Night Fever' turned out," Richardson states. "It had spark and it sounded wonderful."

However, when the Hollwood honchos were not quite so enthused, attention was refocused on 'Stayin' Alive'. Unfortunately, by this time Dennis Byron had had to fly back to England when his father passed away just days into the sessions, and with no replacement drummers happening to wander by the remote studio in the middle of winter, Barry asked if there was any way they could use the rhythm machine inside the facility's Hammond organ to make the track sound more steady. In the days before Linn Drums, this was quite a request, prompting Karl Richardson and Albhy Galuten to suggest that it might be workable if augmented by Barry's own rhythm guitar.

"We were able to get a 4/4 beat out of the Hammond, but when Barry played along to it we didn't like the result," says Richardson. "Then Albhy and I came up with the idea of finding two bars [*of real drums*] that really felt good and making an eternal tape loop." In fact, the engineer's initial intention had been to take two bars of the four-track drums from 'Night Fever', re-record these about 100 times and then splice them together in order to create a new track. It was only while he, Galuten and the band listened and listened and listened to the song over the Auratones in order to find the best couple of bars that Richardson then decided to copy these onto the half-inch tape of an MCI four-track machine and create the aforementioned loop.

Photos courtesy of Dick Ashby

The Bee Gees: disco kings.

"The drums from 'Night Fever' basically consisted of two bars at 30ips," he says. "The tape was over 20 feet long and it was running all around the control room — I gaffered some empty tape-box hubs to the tops of mic stands and ran the tape between the four-track machine and an MCI 24-track deck, using the tape guides from a two-track deck for the tension. Because it was 4/4 time — just hi-hats and straight snare — it sounded steady as a rock, and this was pre-drum machine. For the tempo I used the varispeed on the MCI four-track, so the drums that ended up on the 24-track were at least third-generation, and because the tape heads were so badly worn I brightened the tracks that were already Dolby A-encoded with high-end EQ from the API console."

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Drum *Lupé*

The drum loop would go on to have quite a career in its own right, serving as the backbone to not only the 'Stayin' Alive' and 'More Than A Woman', but also Barbra Streisand's 'Woman In Love'. "That steady, steady track gave us the groove we wanted, and we then overdubbed everbody to it," Richardson continues. "The guys did their vocals, Alan played the guitar riff, Blue played electric piano and an ARP string synth, and when Dennis returned he overdubbed the toms, crash and hi-hat. He loved it. A case of a lot less work. And the fun thing was, when we listed the credit on the record, the drummer on 'Stayin' Alive' was listed as Bernard Lupé; a sort of French version of the famous session drummer Bernard Purdie. Well, we received an unbelievable amount of calls looking for this steady drummer named Bernard Lupé. You know, 'This guy's a rock! I've never heard anyone so steady in my life!'"

Barry and Maurice's lead vocal was laid down after they and Robin stood around a Neumann U67 to record their harmonies, compressed with a Urei 1167. "They just went out into the studio and nailed it," says Richardson. "It didn't take long at all. In other cases, if they didn't get the execution or the balance, it was easier to do it again. It would take longer to argue about it than to redo it, as they were all natural vocalists. The key was that Barry Gibb doesn't really have vibrato, he has tremolo, so his intensity changes but not necessarily his pitch, whereas Robin has very fast vibrato and there are lots of pitch changes. Mo was somewhere in between. Depending on where he was in his range, he either had a little bit of vibrato or just straight tone. So, the distinctiveness was all three voices combining to make this unusual blend that you'd never get anywhere else. Nobody was tracking each other's vibrato, I can tell you that."

Following the recording of vocals at the Chateau, the timbales on 'Stayin' Alive' were overdubbed by Joe Lala at Criteria, where the Miami String Section also embellished Blue Weaver's ARP synth parts, and more strings were added at Capitol Studios in LA.

"We knew we had a smash track," restates Karl Richardson. Nevertheless, the path to true happiness is never straightforward, and the bug in this case was a request from the movie's director, John G Avildsen, to write a bridge section for 'Stayin' Alive' that would momentarily slow it down for a scene where the two main protagonists fall in love on the dance floor. Then the song, and the characters, could return to the up-tempo disco beat. Do you get the picture? Barry and Robin tried to...

"They did write a bridge," Richardson says, "and there is a version of 'Stayin' Alive' where the song actually changes key and turns into a slow ballad for 16 or 32 bars, before there's a big drum break and everything reverts to normal. Well, after that had been recorded and I'd spliced it into the track, Albhy and I stared at each other and said 'Boy, we just ruined a hit record.' We then turned to Barry and said 'We can't use this. We've just screwed up a number one record,' and Barry said 'Yeah, we have.' So, we put the tape back together without the new bridge and called Stigwood to say 'This is bogus. We're not doing it.' And that's when Stigwood fired the director."


Karl Richardson today.

John G Alvidsen departed the project shortly before the commencement of principal photography and was hastily replaced by John Badham, who was happy to use 'More Than A Woman' for the pesky dancefloor love scene. So it was that any 'creative differences' were resolved, and 'Stayin' Alive' was ready for mixing once Barry augmented the fade-out with a vocal chant on which Karl Richardson boosted the high frequencies.

In the meantime, there were the other original numbers, including 'How Deep Is Your Love', a soulful, heart-rending ballad that was closer to the Bee Gees' folk-pop days of the late '60s and early '70s in terms of its compositional structure. "Again, I knew from the start that song was going to be a hit," says Richardson. "The songwriting was exquisite, and so at that point all we could do was screw it up. It was Barry who came up with the initial idea, and he and Blue Weaver then developed that. Blue was on electric piano, Barry was on acoustic guitar, and in an afternoon they wrote 'How Deep Is Your Love'. The music was written, I guess, in a couple of hours, and then Barry, Robin and Maurice kind of huddled together and came up with most of the lyrics later that day. I think the second verse still needed to be written — there was a lot of 'hummeny, hummeny, hummeny' — but three or four days later we were doing the vocals."

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An Awful Lot Of Fun

In all, the Chateau d'Herouville sessions lasted just under two months, which was all the more remarkable since Robert Stigwood also kept Karl Richardson busy there mixing the *Bee Gees... Live* double album, culled from an appearance at the Los Angeles Forum that had been recorded by the Wally Heider Record Plant mobile unit. Stigwood knew all about cashing in on a good thing, and RSO scheduled the live album for release before the *Saturday Night Fever* soundtrack.

After overdubbing in LA and Miami, the mix sessions for *Saturday Night Fever* took place at Criteria. And whereas a rough mix sufficed for 'Night Fever', that for 'Stayin' Alive' was a good deal more complex and, consequently, the first one on which Richardson ever used automation. "It was only 24-track, and in those days the same pair of tracks would jump from strings to percussion to guitar to strings to percussion," he explains. "There was a lot of moving around in the mix, so automation benefited that."

Released as a single in late 1977, 'Stayin' Alive' spent four weeks at number one in the US, remained on the charts for over six months, and has come to represent the best or worst of musical times depending on people's personal preferences. In either case, its significance cannot be ignored, and neither can its impact on many of the late'70s generation.

"In those days we were just having an awful lot of fun," says Karl Richardson, who would go on to do sound design for Broadway musicals from the late '80s through to the late '90s, before returning to Miami and basing himself at the Audio Vision Recording facility of Steve Alaimo and old Criteria colleagues Ron and Howard Albert. "There wasn't too much concern about making a hit record. It was all about cracking jokes, having fun in the studio and making music. There was no pressure. That would come later. Instead, we just thought it was normal to go to the studio, mess around for awhile, have some laughs, get on an airplane, do this, do that and produce a number one... Little did we know." SOS

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Comparing Audio Restoration Plug-ins In Pro Tools

Workshop

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Technique : Pro Tools Notes

For this month's *Pro Tools* feature, we line up three of the most popular audio restoration packages available. Which gives the most natural results, which is easiest to use, and how do you get the best from them?

Mike Thornton

High-quality audio restoration has become available to the project studio at sensible prices not from one but several different manufacturers, enabling the user to 'resurrect' otherwise unusable recordings. Typical defects, including noise, hum, clicks, crackle — and combinations of any or all of these — can now be reduced or eliminated using a variety of affordable tools.

In this month's *Pro Tools* feature I am going to take a look at three mid-range products — Waves' *Restoration Bundle*, Sony *Oxford Restoration Tools*, and BIAS *Sound Soap Pro* — and then run some comparative tests using the same source material, trying to achieve the best results possible with each. Subscribers and those who bought the magazine at UK newsagents will find the some of the source files and the treated versions on this month's covermount DVD-ROM.

Audio restoration can be both an art and a science. By and large, removal of clicks and tones such as hum is more of a science, in that you can clearly hear and measure how much of the problem signal has been removed, and the 'disturbance' can be easily distinguished from the signal in either the frequency or time domain. On the other hand, noise and crackle are more difficult to remove without starting to damage the original sound; side-effects, often called 'artifacts' appear as noise and crackle are removed, since these tend to contaminate all samples and all frequencies. An artistic judgement is required to determine how much noise can be removed before the artifacts become unacceptable.

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Know Your Noises

The unwanted noises found on vinyl and tape recordings are usually divided into several types, and dealing with them requires separate tools:

- Large clicks or pops

These are defined as an impulse comparatively long in duration — usually more than 2ms — and very high in amplitude compared to the surrounding programme audio. In the example shown the click is around 50 samples in length at 44.1kHz and has a very high peak. Typical large clicks would be scratches on a vinyl recording or digital errors caused by clocking problems. Note that physical damage to the media such as vinyl scratching usually corrupts the audio for more samples than digital errors, and so is more difficult to repair.

- Small clicks or crackles

These are much smaller than large clicks or pops. In this example the click is only a 'minor blip' on the audio waveform, and is only a few samples long. Unlike large clicks or pops, these do not obliterate the underlying audio. Small clicks or crackles usually come from surface noise on a vinyl recording or low-level digital clicks.

- Hum and buzz

When a recording gets 'infected' with hum or buzz, it usually isn't a single frequency such as 50Hz or 60Hz induced from the mains supply, but will often have harmonics at 100Hz, 150Hz and so on. A a good hum-removal tool must, therefore, be able to remove the harmonics as well, preferably automatically. Indeed, buzz generated from thyristor lighting dimmers will have very complex harmonics that continue well up the frequency spectrum. The other type of buzz that can be encountered is 24Hz film camera noise, and this, too, will include a lot of harmonics.

- Broadband noise

The term broadband noise refers to tape noise, hiss and so on, which usually 'infects' a recording if it was recorded at too low a level. Most tools for dealing with this have a 'learn' feature, where they can detect the noise profile and use that to remove the noise from the audio signal. Broadband noise reduction is the hardest to implement, because the unwanted signal is usually much lower in amplitude than the desired sound and so incorrect use or overuse of this tool can result in unpleasant audio artifacts like 'birdies' or 'tinkly' sounds.

In general, you should undertake click removal first as clicks can disturb the other processes, then crackle removal if necessary. Next comes hum filtering, and finally, broadband noise reduction. Most restoration packages have a suite of tools, each of which deals with a specific problem.

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Restoration Tips

■ When playing original source material, do the best you can to ensure a high-quality transfer. Do anything you can with the source material to to help, such as cleaning the tape heads, baking tapes where appropriate, and gently washing vinyl records with distilled water.

■ Use these restoration tools as sparingly as possible. Resist the temptation to push the tool as far as it can go; instead, choose a setting that just provides the desired result. Treating your audio using several passes with a gentle setting can be more effective and produce fewer side-effects than one pass with a more severe setting.

■ The effectiveness of audio restoration is helped significantly by the masking effect. This is a 'feature' of human hearing, where when two or more sounds occur simultaneously, we tend to notice the loudest sound and ignore the quieter sounds. This principle is also used in audio data compression techniques such as MP3 and the ATRAC format used on Minidiscs. The benefit for restoration work is that we don't necessarily need to achieve a 6dB reduction in the unwanted signal to make it 'sound' like a 6dB reduction.

■ When working on restoration projects, remember that your ears can become accustomed to a new sound and also become tired. Constantly refer back to the source audio: is it really better, or is the treated audio worse? Sometimes the treated audio can sound more objectionable than the original sound with its defects. Come back to the process with fresh ears, to check whether the treated version really is better or not.

■ All restoration packages use both 'look ahead' and 'look behind' to help in processing. You should use 'handles' to allow the plug-in to 'settle in' before you get to any audio you will need in the finished product, and likewise, leave extra audio on the end of a section you want to process. You will hear the 'settling sounds' in some of the processed examples. Also, if you use these plug-ins 'live' then you will need to use the delay compensation feature, if you have it, or slip the track to compensate if you don't.

■ Instead of treating an entire track with the same settings, consider treating small sections that have bad clicks with a severe setting and the rest of the track with a lighter setting. This can also apply to other restoration processes too.

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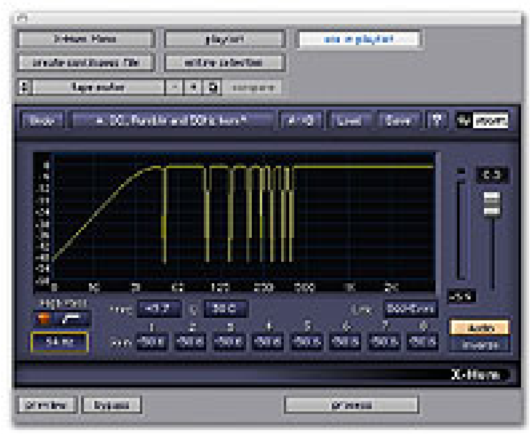
The Restoration Tools On Test

Waves' *Restoration Bundle* has been round the longest of the three packages on test, and is available either as a native bundle supporting all the major formats, or as a TDM bundle supporting HD cards and HTDM host-based processing. Both bundles include Audiosuite off-line versions.



The *Restoration Bundle* is made up of four separate plug-in modules. Waves *X-Click* handles the large clicks or pops and has two controls to adjust for optimum click removal: Threshold sets the amplitude of the targeted clicks, with higher values removing more clicks and a zero setting leaving all clicks untouched. A good starting point is between 30 and 50. Shape corresponds to the width or duration of the clicks to be removed. Lower values remove smaller clicks like digital errors, while scratched vinyl record clicks need a higher Shape value. I find a setting of 70 is a good starting point for vinyl record restoration.

For additional help in finding the optimum settings you can use the Difference feature, which enables you to hear the difference between the original and the processed signal — in other words, what is being removed by *X-Click*. If you start to hear too much of the programme audio in the Difference mode, try lowering the Threshold and/or Shape to minimise the removal of audio transients.

Waves *X-Crackle* handles low-level clicks and crackles that are interspersed in the audio, and again, has two main controls to adjust. Threshold sets the amplitude for targeted crackles; a setting of between 50 and 70 is a good starting point, and you can go as far as 80 or more on noisy signals without introducing too many artifacts. Reduction sets the attenuation applied to the detected crackles, and like *X-Click*, *X-Crackle* has a Difference feature.

Waves *X-Hum* handles hum problems as well as low-frequency rumble and DC offset, and consists of two sections. The high-pass filter section removes rumble and DC offset, with a Slope switch and a frequency control to determine the cutoff frequency. To eliminate only DC offset, set the frequency control at around 10Hz. The harmonic notch filter section handles the filtering of the hum and its associated harmonics, with a Frequency control setting the fundamental frequency of the first filter; the filters for the harmonics automatically track this one. There are also global Q and gain controls, which set the width and depth of the notch filters. An Inverse option switches the filters to boost rather than cut — it can sometimes be useful to boost a problem area to help tune into it and then flip back to the normal cut mode.

Waves *X-Noise* handles broadband noise such as tape hiss and air conditioning. There are the usual Threshold and Reduction parameters, as well as some additional controls including a Learn feature that enables you to 'teach' *X-Noise* the noise you wish to remove. You do this by selecting a section of audio which contains just noise and no programme material. Clicking on the Learn button causes it to flash. You then play the selected section of noise and click the Learn button again, whereupon *X-Noise* will create a Noise Profile, which you can save. If you cannot find a 'noise only' section, there are some standard profiles in the factory presets to choose from.

Now use the Threshold and Reduction controls to achieve the desired amount of noise reduction. Start with a Threshold of 10dB and then use the Reduction control to set the degree of noise reduction. Increasing the Reduction setting increases the amount of noise reduction and increases the chance of artifacts appearing. If these appear before you get the desired degree of noise reduction, try increasing the Threshold to 30dB and readjusting the Reduction control to suit. You can also adjust the Attack, Release, Resolution and High Shelf controls to improve the way *X-Noise* responds to the audio and noise.

As in *X-Click* and *X-Crackle* there is a Difference option in *X-Noise*, which enables you to listen to what *X-Noise* is removing and adjust it so noise is removed without removing programme audio as well.

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BIAS *Sound Soap Pro*

Sound Soap Pro, which was reviewed in February 2005's *SOS* (www.soundonsound.com/sos/feb05/articles/soundsoap.htm), can work with a wide range of host applications that support VST, RTAS, Audio Units or Direct X plug-in formats. Rather than being a series of separate plug-ins, it has a single main window with a series of tabs to access the different sections. The main window includes a number of global options such as loading and saving presets, four temporary settings which can be stored for instant comparisons, and a noise-only option like the Difference feature in the Waves bundle for listening to what is being removed. Across the bottom are the tabs relating to the four sections of the plug-in.

The Hum and Rumble section is very similar in operation to Waves *X-Noise*, in that there is a Frequency control to set the point of operation of the first filter, plus Q and Depth controls. The rumble filter control has a fixed slope of 12dB per octave and a variable Frequency control. The Harmonics control determines how many harmonics of the hum frequency will be filtered, with 1 being just the fundamental and 9 being notches for every harmonic up to the ninth. The Tilt control allows you to filter the harmonics progressively less severely than the fundamental, to compensate for the fact that the harmonics of hum and buzz tend to be lower in amplitude than the fundamental.

The Click and Crackle section is like Waves' *X-Click* and *X-Crackle* combined, and offers only two controls. The Click Threshold control has a range from 1dB to 25dB: the higher the threshold (taking the slider to the left), the more severe the process. The Crackle Threshold control works similarly and has a range of 12dB to 25dB. Unlike *X-Click* and *X-Crackle*, there are no separate Reduction controls, but there is a Click Meter which flashes red when a click is being repaired.

The Broadband noise reduction section has 12 control bands, each with Threshold and Reduction controls, and an audio level meter. The actual frequency range of each band is not fixed but is dependent on the learned noise profile. Broadband has two learning modes. The first is Snapshot, where a snapshot is taken of the noise profile when the Camera icon is clicked. The second is a timed learn mode; whilst playing a section of 'noise only' audio it averages out a noise profile from the point the stopwatch icon is clicked to when it is clicked again to stop the learning process.

The horizontal Threshold-Reduction slider changes the way the controls are displayed. At its leftmost position you see only the Threshold controls, whilst only the Reduction controls are visible when it is over to the right-hand side, and both are on show in the middle. The Threshold and Reduction controls work as expected, and the controls for the separate bands can be linked together. The is achieved with the master lock and unlock buttons, together with a button at the bottom of each section showing whether that section is locked into the group or not.

The Attack and Release controls operate as you would expect, with their respective Tilt controls allowing for different attack and release times across the different bands. With the Release Tilt set at 1.00, the release times will be the same across all the bands, but with the Release Tilt control set to 2.00, the release time for the lowest band will be as set by the Release control and the release time for the highest band will be half that.

The final section is a noise gate, which is not featured in either of the other suites under test here. It is a conventional arrangement, with a Threshold control ranging from 0dB to -60dB and a Reduction control setting the amount of attenuation of the signal below the threshold, with a range of 1.00 to 5.00. There is also an Attack control with a range of 10ms to 500ms and a Release control with a range of 50ms to 1000ms, as well as a visual display to show how the gate is functioning.

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Sony *Oxford* Restoration Tools

This set of tools is made up of three separate plug-ins, and is available as an RTAS plug-in with Audiosuite support.

Declick is similar to the Waves *X-Click* and *X-Crackle* plug-ins, but is presented as one plug-in with two sections. The first section deals with what are normally referred to as 'clicks', though Sony call them Pops. It has a Threshold control, which you start off at the minimum setting and increase until the clicks go away. If you start to get distortion, the Threshold setting is too high. The next control is Soft, which determines the length of the repair of a pop; the higher the setting of the Soft control, the longer the pop that can be repaired.

The final control in the Pops section is Sensitivity. This determines whether a piece of audio is detected as a pop in the first place. If you are unable to remove pops for their full duration just by using the Threshold and Soft controls, try increasing the Sensitivity and then readjusting Threshold and Soft to minimise the risk of distortion. Alternatively, if the Pops section produces distortion even at minimum Threshold and Soft settings, try reducing the Sensitivity and readjusting Threshold and Soft again.

The second section, which Sony call Clicks, is designed to deal with what is normally called crackle — Sony's terminology is somewhat confusing and I prefer the conventional expressions, which describe the problems more accurately. This section has two controls, Sensitivity and Repair, and to use it, you start with the Sensitivity at the bottom and slowly increase it until the crackles are gone. The Repair control helps to put some of the harmonics back into the sound. Set at maximum it should provide audio clean of crackles, but may sound somewhat dull. Slowly turn the Repair control anti-clockwise and the high frequencies should reappear, but if you turn it too far, you may get artifacts.

Debuzz is Sony's hum removal plug-in, and is very different from Waves *X-Hum* and *Sound Soap Pro*'s Hum and Rumble tab. It doesn't have a high-pass filter or Harmonics controls, but does have the ability to track the fundamental. This means that if the frequency of the hum drifts for any reason — for instance, if the tape recorder's speed varied — *Debuzz* will track it and adjust the filters to follow the drifts. The Frequency control sets the fundamental frequency for the filter section, while the Adapt button enables the tracking feature and Freeze disables it. Use the Strong option if the fundamental is much louder than the surrounding audio and the Weak option if the buzz is lower in level and so closer to the programme audio.

You might also find that if you are unable to remove all the buzz in either Strong or Weak modes, two passes will be more successful. Do the first in Strong mode to remove the main fundamental and the second in Weak mode to remove all the low-level buzz. Whichever mode you choose, set the Frequency control as close to the fundamental as you can get it with Adapt mode on. You will see *Debuzz* 'tune in' and 'lock on' to it. Then adjust the Gate control; starting from the maximum position, bring it down until just before the buzz becomes audible. Note if you change between Weak and Strong modes you will need to readjust the Gate control.

The Bandwidth control determines the maximum frequency up to which the buzz filtering will be applied, while the Sharp control is the Q control and Depth sets the depth of the filters. It can usually be set to 144dB, but if you find this is removing too much programme audio, you can reduce the Depth to suit.

Denoise is Sony's broadband noise-reduction plug-in, and has three modes of operation. Automatic Mode is the best option when the noise profile changes with time or where no clean noise is available to learn from. To use this mode, turn on both Auto and Adapt; the noise profile is then continuously updated by tracking the noise in the audio. The Level control functions as a threshold control and adjusts the height of the noise contour in the graphical section, while the Smooth control determines how closely the noise contour follows the noise profile. The lower the setting, the closer the contour will be to the profile. In Automatic mode, the Smooth control should be set close to the maximum.

The Correlation control only works in Automatic mode, and provides a means for fine-tuning the automatic noise profile. Low values of Correlation will produce fewer artifacts, especially on low-level sounds like decaying notes or intakes of breath, but with an increased risk of noise pumping, where you hear the noise coming and going with the loud and quiet audio. If you experience noise pumping try increasing the Correlation setting.

Fingerprint Capture Mode is Sony's 'learn' mode and is achieved by pressing the Auto and Adapt buttons, then playing a section of 'noise only' audio. Pressing the Freeze button then takes a snapshot or 'fingerprint' of the noise profile. Start with the Smooth control at maximum, and only decrease it if there are distinct notes or tones in the noise, as lower values of Smooth will allow the noise profile to follow the bumps created by the notes or tones.

The final mode is Manual and can be used for simple noise reduction, usually when the noise is 'whiteish' and static, like tape hiss. Smooth and Correlation do not apply in Manual mode. The Colour control adjusts the contour in the low-frequency range of the noise profile from white noise (fully counterclockwise) to pink noise (halfway) through to red noise (fully clockwise).

The Attenuation control is the primary means of adjusting the amount of noise reduction; the greater amount of noise reduction, the greater the risk of artifacts, the most common of which are tinkly random tones.

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The Tests

Let's now take a look at how these three suites actually work in practice. To do this, I treated the same audio examples with each suite, aiming to get the best results possible with each for comparison. If you bought your copy of *SOS* in the UK, or you are a subscriber, you will find these original audio examples on this month's covermount DVD-ROM, as well as the treated versions from each suite. For each example, I have included the original untreated file and then three treated versions, with the relevant postscript to the file name to indicate which package it was treated with. I have also saved the plug-in settings that were used in each case.

All treated files were done in one pass using one plug-in, with settings taken to the point where artifacts are just becoming audible. I have scored them out of 10 in two categories. A high 'ease of use' score means that it was quick and easy to get to a good result, and quality of processing indicates how good that result was.

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No Earth

The first clip is called 'no earth'. It was part a recording that came to me that had been recorded at someone's funeral; unfortunately there hadn't been time to check that the recording setup was OK, and so we have a recording where the hum is nearly as loud as the programme audio, as it turned out that a lead with a faulty screen connection had been used!

All three packages were able to filter out the hum very successfully, though Waves *X-Hum* and *Sound Soap Pro* did leave some high-frequency buzz. This could be treated with a broadband noise-reduction plug-in — which is what I did for the client at the time — but the Sony suite does stand out over the other two as giving the cleanest result. I also appreciated its ability to fine-tune to the exact frequency of the hum automatically; with the other two, I had to set the fundamental frequency myself, and in both cases the harmonics filters didn't go high enough to remove all the high-frequency buzz.

No Earth	Ease of use	Quality of processing
Sony *Oxford Restoration*	*9*	*8*
Waves *Restoration Bundle*	*7*	*7*
BIAS Soap Pro*Sound*	*6*	*7*

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Tape Motor Noise

This was a different challenge for the hum-filtering packages. This was part of a set of recordings brought to me by a client that she made when she was eight or nine on a Grundig open-reel tape recorder, recorded on quarter-inch tape at 3.75ips. I actually replayed them using her original machine! All three packages where able to remove the hum and reduce the motor noise, and as you will be able to hear, they all gave very similar results. Again, as with the 'no earth' test, a subsequent pass with a broadband noise reduction plug-in would be able reduce the motor noise even further.

Tape Motor Noise	Ease of use	Quality of processing
Sony *Oxford Restoration*	*9*	*9*
Waves *Restoration Bundle*	*7*	*7*
BIAS *Sound Soap Pro*	*6*	*6.5*

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Vinyl Record

This is a typical example of a scratched vinyl record, with the test being how well each package is able to remove the clicks and fill in the holes in the sound. This was the only test where two plug-ins were used together, as in the Waves *Restoration Bundle*, click and crackle removal are two separate plug-ins, which is not the case in Sony *Oxford Tools* or *Sound Soap Pro*. Interestingly, the Crackle section of the Sony plug-in was able to remove nearly all the 'clicks' on its own, but I chose a setting where both sections were working equally. I marked the Waves bundle down on its 'ease of use' because it is two separate plug-ins, so you are swapping between two windows all the time. As you will hear, all did remarkably well and were able to remove nearly all the clicks; the Sony *Oxford Tools* stands out as producing the best results, with a close call between the other two.

Vinyl Record	Ease of use	Quality of processing
Sony *Oxford Restoration*	*8*	*9*
Waves *Restoration Bundle*	*7*	*7*

BIAS *Sound Soap Pro*	*6*	*8*

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Hiss

This is an example of a track that had been poorly recorded to Minidisc using the wrong lead. There are a number of single-point stereo mics about that have unbalanced outputs on a three-pin XLR cable. The lead for these mics looks the same as one you might use for a mono mic with a balanced output — both have a 3.5mm plug on one end and a three-pin female XLR on the other — but the two leads are wired differently. If you use a stereo lead with a mono mic it will work, but with two problems. One is that the two tracks will be out of phase, and the other is that the recording will be much noisier. The phase problem can be resolved by only using one channel, but you are still left with a noisy recording, which is what this example reproduces.

Broadband noise reduction is always the trickiest of all the restoration processes as it is always a trade-off between good noise reduction and artifacts. Patience and constant use of the Bypass button are called for, and perseverance will be rewarded.

Using the 'learn' option to create a noise profile will always give the best results. The results were very similar for each of the plug-in and could have been improved by using a more gentle setting and multiple passes. I marked the Sony *Tools* down on ease of use, as the controls are not as intuitive as I would have liked, whereas the Waves controls are very clear.

Hiss	Ease of use	Quality of processing
Sony *Oxford Restoration*	*7*	*8*
Waves *Restoration* Bundle	*8*	*8*
BIAS *Sound Soap Pro*	*7*	*8*

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The Final Analysis

- BIAS *Sound Soap Pro*

This has some very nice features. I thought at first that the 'all in one' design would be annoying, but I came to really appreciate it, as most restoration jobs do involve using a combination of processes. In general I tend to use an RTAS or TDM plug-in on a track to set up the parameters, copy the settings to the corresponding Audiosuite plug-in for processing, and then take out the RTAS or TDM plug-in, so not tying up valuable DSP resources and also making the job more 'portable'. This is made easier with *Sound Soap Pro* as all the processes can be run together as one, whereas with the others you have to run the Audiosuite plug-ins one at a time.

On the down side, I didn't find the graphical display in *Sound Soap Pro* any help in adjusting the plug-in — it felt like it was consuming DSP power for no obvious benefit — and in my opinion, the Gate section actually devalued the package, as in the field of restoration using a gate is like using a mallet to crack a nut! However, I liked the ability to 'unlock' and adjust the Threshold and Reduction controls separately for each band in the broadband noise-reduction section.

- Waves *Restoration Bundle*

I own the Waves bundle and have always been impressed with their products. I did find it the most intuitive to use, even taking into consideration the fact that I have used it several times before.

However, compared to the other two packages, both of which are relative newcomers in this field, the Waves *Restoration Bundle* is getting a little long in the tooth. I hope Waves are working on a 'v2' *Restoration Bundle* with improved algorithms as Sony have shown how good restoration processing has become.

- Sony *Oxford Restoration Tools*

This is an excellent bundle and is very competitively priced. Although in general I found it easy to use, I did find the naming of some of the controls unintuitive and I had to refer to the manual quite a lot to check what they did. Having said that, the reason the Sony bundle got high 'ease of use' scores was that I often found the default settings were a very good starting point, while the 'auto tune' feature on the *Debuzz* plug-in was truly awesome. In addition, they have come up with an automatic mode for the broadband noise reduction plug-in, recognising that it isn't always possible to isolate a section of 'clean noise' to learn, although I did find it was able to produce better results when it had a noise profile to go at.

I missed having a 'noise only' option on the Sony *Restoration Tools*, as I found it a real help with the other packages to check what the process was taking away. However, spending too much time listening to the 'noise only' output did also lead me down some blind alleys, and with experience, I got the best results by listening mainly to the complete output, with occasional checks to the 'noise only' to make sure the process wasn't taking too much out. SOS

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Technique : Digital Performer Notes

As always, we've news and ideas from the *DP* scene — as well as some good old-fashioned rumour-mongering...

Robin Bigwood

They say that no news is good news, and this month, as far as *Digital Performer* itself is concerned, that might well be correct! *DP* 4.52 is proving to be a very stable incarnation of the program, particularly so in OS 10.4 'Tiger'. I've read on several internet forums and email lists that die-hard *DP 3* devotees have finally been tempted into the 21st Century by OS 10.4 and *DP 4.5* and have found the new platform very much to their liking. Tiger certainly appears to have delivered some performance improvements and runs superbly on older Macs, as well as shiny new ones.

New *DP* features are very definitely on the horizon, though. MOTU have, of course, already trailed *Autotune*-like track-based pitch correction and a few other gadgets, but recently, when I was reviewing the notes I'd made at the Sounds Expo show in London earlier this year, I was reminded of something very intriguing I'd spotted at the time. On the development version of *DP* that MOTU were showing, audio plug-in and virtual instrument windows all had an additional title-bar button, labelled with a mysterious single letter: 'V'. When I enquired about this, the MOTU representative very diplomatically told me not to ask awkward questions! I've since pondered what 'V' could stand for — my instinct tells me that MOTU might have some sort of distributed processing system up their sleeve, but how 'V' relates to this I can't imagine. If anyone has a brainwave on this I'd love to hear it!

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Plug-in Corner

- PSP: Like many other *DP* users, I've long been a fan of the Lexicon-inspired delays and other plug-ins produced by PSP Audioware, but one or two of them, particularly *Nitro*, have had stability problems when running as Audio Units in *DP 4.5*. PSP recently announced updates of the whole range and it looks very much as though all the problems have been fixed. For information, the latest revision numbers as I write are as follows: *Lexicon PSP42*, 1.4.4; *PSP 84*, 1.4.4; *Nitro*, 1.0.5; *Vintagewarmer*, 1.6.6; *Mixpack*, 1.8.5.

- Wave Arts: Building on the technologies used in the *Masterverb* and *Wavesurround* plug-ins, and apparently extending them significantly, the new *Panorama* plug-in from Wave Arts is an acoustics processor that includes highly configurable reverb and early-reflection algorithms with doppler-effect processing and pyschoacoustic filtering. It shares many similarities with Waves' *Doppler* and Audioease's *Orbit*, but is not quite the same as either. The selectable materials (Audience, Concrete Block, Curtain, etc) for the walls and floors seem to take their cue from *Realverb Pro* too. Plug-ins like *Panorama* aren't for everyone, but they can be useful for dialogue mixing and post production, as well as for unusual music-production effects where sounds seem to be located well outside of the speakers, or even behind the listener. If it's the sort of thing that appeals to you, rest assured that *Panorama* has the quality and maintains the ease of use that characterise all of Wave Arts' plug-ins, and it's available in a true MAS-format version. It costs $249 but is available as a 30-day demo. More information from www.wavearts.com.

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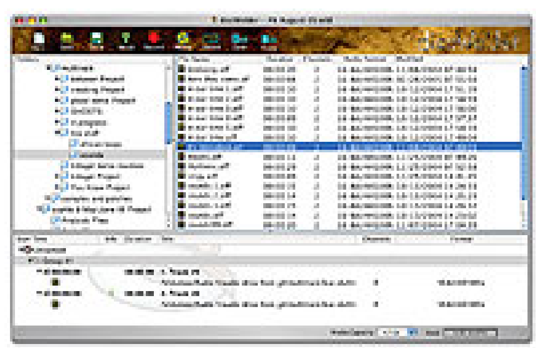
DVD Authoring

DP's surround-mixing support is amongst its strongest features, but actually doing something with a surround mix can be a problem. *DP* can play it, of course, but for wider distribution you're faced with getting it into a format that consumers can play back. This often means using specialist hardware (or the likes of Universal Audio's *Smartcode Pro* plug-in for Pro Tools) to encode a Dolby Digital or DTS file ready for DVD production, or again relying on big-money hardware or specialist software (ie. Sonic Solutions or Pyramix) to convert your audio into the DSD format that is used for SACD discs. Complicated stuff, and none of it cheap.



New copies of *DP* are now coming bundled with a trial version of *Discwelder*, a basic but very useful DVD-Audio authoring application.

In June, though, MOTU announced a collaboration with Minnetonka Audio Software, which means that new copies of *DP* now come bundled with a trial version of the *Discwelder Bronze* DVD-Audio authoring package. What's more, current *DP* users are not excluded from the deal, as they can download the trial version of *Discwelder Bronze* from www.motu.com. As a surround format, DVD-Audio can deliver 5.1 mixes at 48kHz/24-bit, but can also support high-resolution stereo recordings — up to 192kHz, 24-bit, in fact. You can even have surround and stereo tracks on the same disc, so listeners can choose what's best for their playback systems.

DVD-A is arguably the least widely compatible current audio distribution format, as it won't play on conventional CD players (unlike SACD), and isn't supported by most basic DVD (video) players or computer drives. Still, it has huge industry support, and as consumer hi-fi and home-theatre devices offer ever-broadening format compatibility, a cost-effective DVD-A authoring solution is welcome.

The *Discwelder Bronze* trial allows five DVD-A discs to be burned before becoming inoperative, but upgrading to a full license costs only $49. More details from www.motu.com and www.minnetonkaaudio.com. Find out more about DVD-Audio at www.digitalaudioguide.com. SOS

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SOUND ON SOUND
The World's Best Music Recording Magazine

In this article:

M for Mobile...

PC Notes

Published in SOS August 2005

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Technique : PC Notes

The Pentium M processor has various desirable qualities, not least its very low power requirement, which leads to a reduced need for cooling and a quieter life for musicians. However, it has always been confined to use in laptops, with corresponding limitations — until recently.

Martin Walker

When I bought my Centrino 1.4GHz laptop from Millennium Music Software back in October 2003, it was after considerable thought, but its Socket 479 Pentium M processor was the key to my choice. Intel generally quote a Pentium M performance factor of 1.5x compared to their P4 range, so the 1.4GHz Pentium M in early tests was claimed to provide roughly equivalent performance to a 2.1GHz Pentium 4. In fact, my own tests showed that with reverb plug-ins my 1.4GHz model performed as well as a Pentium 4 2.2GHz, while its performance with many EQ and compressor plug-ins measured closer to that of a 2.8GHz P4 (more details in PC Notes December 2003). However, time marches on, and the latest 2GHz Pentium M 'Dothan' models are now turning in performances roughly equivalent to Pentium 4 Prescott processors in the mid-3GHz area.

Actually, it wasn't processing performance that first drew me to the Centrino laptops, but low acoustic noise. Pentium M power consumption is considerably lower than that of Intel's other processor ranges, especially when compared with the Prescott models. For example, typical processors from both AMD and Intel offering performance roughly equivalent to a 3GHz Pentium 4 have a maximum power dissipation of between 70 and 90 watts (some of the faster Prescott models rise to 115 watts), whereas an Intel Pentium M Dothan processor of 2.0GHz is rated at just over 20 watts!

This low power consumption means that Centrino laptops have significantly longer battery life than models running different processors, but it also means that keeping Pentium M temperatures within safe limits is a lot easier, which, in turn, results in low acoustic noise. On my Centrino 1.4GHz model, for instance, the cooling fans rarely spin up at all, making this the perfect computer when you want to plug in a microphone and make an audio recording in the same room.

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PC Snippets

- If you're having problems with Firewire devices providing reduced performance after installing Windows XP Service Pack 2, you'll no doubt be pleased to hear that Microsoft have released a 509KB update (KB885222) that should cure the problem once and for all. It solves the issue of 1394a (Firewire 400) or 1394b (Firewire 800) devices plugged into 1394b (Firewire 800) ports being reduced to 100 Megabits/second instead of remaining at the default 400 Megabits/second speed. You can download it from http://support. microsoft. com/kb/885222.

- Anyone about to undertake a radical re-partitioning of their hard drives following my May 2005 *Sound On Sound* feature on the subject will no doubt be interested to hear about a free alternative to the famous *Partition Magic* utility. *Knoppix* is available as part of a Linux image file that you burn onto a CD-ROM. You can then boot from it to perform various drive-salvaging and troubleshooting tasks, including the partition management functions of *QParted*. *Knoppix* can be downloaded from www.knoppix.com.

- Since my discussion of *Cubase SX* MIDI timing problems in PC Notes April 2005, I've discovered that Steinberg have also confirmed that some of their users may suffer from audio timing problems with *Cubase SL 3* and *SX 3*, and they have posted a temporary solution. To see if you're affected, check the VST Audiobay page of Device Setup. If your audio interface displays different latency values for inputs as it does for outputs, there will be an offset when you're undertaking audio recording. To measure the offset and correct for it manually, you can download an Audio Loopback Test song from http://forum.cubase. net/phpbb2/ viewtopic.php?p=23991#23991. You can open the *Cubase* notepad to read embedded instructions on how to proceed.

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The Mobile Desktop

This Micro ATX-format motherboard from AOpen is ideal for a small 'shuttle' PC, but is unusual in supporting the Pentium M processor normally only seen in Centrino laptops.

With higher Pentium M clock speeds now resulting in laptop performance similar to that of many desktop PCs, but with much lower acoustic noise levels, by now some of you may have guessed where I'm heading with this discussion. Many musicians have dreamed of creating a Frankenstein-style desktop PC containing a Pentium M laptop processor, and various motherboard manufacturers have done their best to make that dream a reality.

Two examples are AOPen's i855GMEm LFS and DFI's 855GME MGF motherboards, both launched near the end of 2004. Both of these have a Micro

ATX form factor (whose maximum dimensions are 9.6 inches square compared with the 9.6 inches by 12 inches of the more familiar standard ATX format) and are generally used in mini-PCs (such as the one I reviewed from Red Submarine in *SOS* August 2003). As their names imply, these two motherboard models feature Intel's 855GME chipset, as used in Centrino laptops, which further contributes to their low-power design, as only the necessary portions of the chip are run at any given time.

Both boards are capable of providing good results, and although the 855 chipset doesn't directly support SATA hard drives, AOpen have bypassed this limitation by fitting an additional Promise SATA controller, while DFI fit a non-standard Intel 6300ESB Southbridge chip to provide the same function. This means that both boards offer two SATA ports and RAID support, in addition to two Ultra ATA/100 ports. AOpen's board also provides three PCI expansion slots, six USB 2.0 ports, two Firewire 400 ports and two Gigabit Ethernet ports, while DFI's board has two PCI slots and one 64-bit PCI-X slot, plus four USB 2.0 ports, two Firewire 400 ports and one Gigabit Ethernet port.

One disadvantage of using a Pentium M processor in a desktop is that it's built for notebook PCs with custom heatpipe cooling, so it doesn't incorporate an integrated heat spreader (IHS). This is the metal cap that you see on most desktop processor chips that sits between the inner core and whatever heatsink you attach. In contrast, the Dothan processor exposes its innards to the world, needs to be treated rather more delicately and requires a special heatsink. Fortunately, both of the motherboards I'm talking about are bundled with a suitable small heatsink/fan combination that will do the job, and although most DIY builders normally discard such bundled items in favour of more exotic and quieter options, in this case the cooling requirements are so modest that an alternative is not really necessary.

Intel's Pentium M 'Dothan' processor, pictured here, generally relies on exotic heatpipe cooling solutions. For this reason, it needs a special heatsink if it's to be installed in a desktop PC.

Sadly, it's the 855 chipset that lets down overall performance, since it only supports single-channel RAM up to DDR333 speeds, resulting in a peak memory bandwidth of just 2.7GB/sec, which is much lower than that offered by most other modern chipsets. This specialised type of motherboard is also significantly more expensive than most desktop alternatives capable of providing similar performance (the AOpen model I mentioned earlier retails at about £160). Nevertheless, they have been welcomed by those who wanted to build desktop machines that are truly silent from a distance of more than a couple of feet, yet still support the user's choice of AGP graphics card, PSU, RAM sticks, and hard drives. If you're interested in building such a machine, there's an excellent guide to be found on the Maximum PC web site (www.maximumpc.com/how_to/reprint_2005-01-28.html).

As I write this, the first reviews of AOPen's second-generation Pentium M desktop motherboard have just started to appear. The i915GMm-HFS features the much more advanced Intel 915 chipset used in Prescott-based PCs (see my September 2004 PC Notes column for more details) and supports dual-channel DDR-II or single-channel DDR RAM for a much higher peak-memory bandwidth, as well as much faster PCI Express x16 graphics cards, and provides integral SATA/RAID support, bringing its potential performance much closer to that of other modern Pentium 4/Athlon64 desktop designs. This motherboard has a truly innovative design but is even more expensive, at around £200.

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Tiny Tips

Further to my PC Musician feature on drive defragmentation in *SOS* June 2005, I've discovered that it is possible to encourage Microsoft's built-in defragmenter utility to 'defrag' a selection of drives or partitions in sequence. What you need to do is create a 'BAT' (batch) file. Old-time DOS users will already be familiar with these batch files, which are essentially text files containing a series of commands, one on each line. Just open Notepad, type in a line for each partition you want to include, such as 'defrag c:', 'defrag d:', and so on, and then save this text as a file with a suitable name, such as Defrags.bat. You can run it at any time by double-clicking on it. Many thanks to *SOS* reader Bill Blackledge for this hint.

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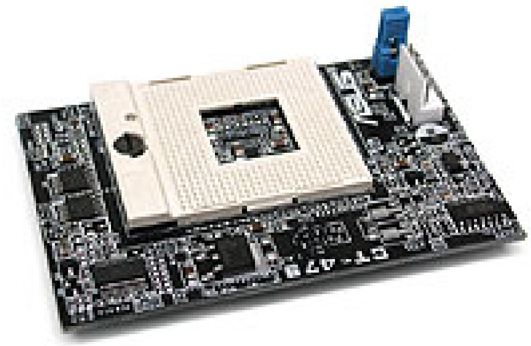
The Pentium M Desktop Adaptor

What many PC users have been hoping for is a way to plug a Pentium M processor into existing motherboards that already have more advanced facilities. After all, the Pentium M is electrically compatible with the Pentium 4, and there are even the same number of pins on its package. The problem is that the Socket 479 pin-out of the Pentium M Banias and Dothan ranges is completely different from that of the Socket 478 used by the Pentium 4 Northwood range (and early Prescott models).

Nevertheless, owners of some Asus motherboards may find that their dreams are about to come true. The Asus CT479 CPU Upgrade Kit can be fitted to various Asus motherboards featuring i865 and i875 chipsets and consists of a small circuit board with Socket 478 pins on the bottom. You plug this into your motherboard and then you can plug a Pentium M processor into the Socket 479 mounted on the top of the circuit board. There's also a three-pin jumper on the board to let you select between 400 and 533MHz FSB (Front Side Buss) speeds. A heatsink and low-noise fan complete the package.

Compatible Socket 478 motherboard models so far include the P4P800 SE, P4P800 VM, P4C800E Deluxe, P4GD1 and P4GPLX, but Asus have promised to add more to this list in due course. All require a BIOS update to accommodate the adaptor. Compatible processors include the Dothan range from 1.5GHz to 2.26GHz+, the Banias range from 1.3GHz to 1.7GHz+, and Celeron 'M's from 1.2GHz to 1.7GHz+. However, the best news is that the CT479 costs just £33, so could be partnered with a Dothan 1.86GHz processor for a total cost of about £240 (the current sweet spot), or £330 with a 2.0GHz model (just 7.5 percent faster for an extra £90 — it's your choice).



Selected Asus Socket 478 motherboard owners can now retrofit a Pentium M processor, courtesy of this CT479 CPU Upgrade Kit, for just £33.

With the fastest 2.13GHz Dothan model installed into an Asus P4C800 motherboard, test results have shown performance that just exceeded Athlon 64 3800+ and 3.6GHz P4 Prescott-based PCs in some areas, while the power consumption of the Dothan was roughly half that of the Prescott. Another benefit of this upgrade is that most of the motherboard BIOS overclocking options are still available. Already it has proved possible to overclock a 2.13GHz Pentium M Dothan up to a massive 2.56GHz fairly easily, by increasing the frequency of the FSB. The overclocked processor beat some Athlon 64 FX and Pentium 4 Extreme Edition machines in 3D game tests!

I can envisage many musicians who still have Asus Socket 478-based PCs wondering about buying a CT479 Upgrade Kit and a Dothan processor to boost their processing performance. It has to be admitted that the Pentium M isn't an especially long-term solution, since current models don't provide 64-bit support (though there will eventually be dual-core models when the next-generation Pentium M 'Yonah' core is released in 2006). Nevertheless, for those Asus motherboard users who have been tentative about abandoning their current PCs in the present climate of change, this could be the perfect quick and easy way to upgrade performance, with the added, and very welcome, bonus of reduced acoustic noise.

If Asus add support for my P4P800 Deluxe motherboard, I'm seriously considering abandoning my Pentium 4 2.8GHz Northwood and 'going Dothan'. In the longer term, I can also see why so many industry experts think that Intel should scrap the Pentium 4 range to concentrate on faster Pentium M models. SOS

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Macs With Intel Processors?

Apple Notes

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Technique : Apple Notes

Apple users are used to transitions, having moved from 68k-based Macs to Power PC processors, and the classic Mac OS 9 to Mac OS X. Now it's time for the third and most shocking transition of all: the move to Macs with Intel processors.

Mark Wherry

There's one word on the lips of most Mac users at the moment, and that word is Intel. After more rumours than usual over the weekend preceding Steve Jobs' keynote at this year's Apple Worldwide Developers Conference (WWDC), this time concerning the company moving away from the Power PC processor architecture to Intel's x86, the Apple CEO confirmed that the rumours were, indeed, true. What this means is that, starting from 2006, Apple will ship Macintosh computers powered by x86 Intel processors, the same processors used by computers running Windows today. Intel will not be manufacturing any Power PC variant for Apple.

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Let's Talk About Transitions...

Jobs introduced the topic of Macs with Intel processors by saying "let's talk about transitions". He went on to describe the two main transitions of the Mac's 21-year history. Firstly, the move from 68k processors to Power PC-based designs during 1994-96, which the Apple CEO described as having "set Apple up for the next decade", calling it "a good move" and, secondly, the more recent "brain transplant" from Mac OS 9 to X during 2001-03. The change from Power PC to Intel marks the third major transition for the Mac, and while Jobs commented that Apple have "great products right now" and "some great Power PC products in the pipeline", he conceded that the company didn't know how to make the products they were envisaging with the current Power PC 'road map'.

Acknowledging that two years ago he stood on the same stage when introducing the Power Mac G5 and promised a 3GHz G5 within a year, Jobs was admirably candid about the fact that Apple hadn't been able to deliver either a 3GHz Power Mac G5 or a Power Book G5. According to Jobs, Intel will be able to help Apple in both of these departments: great performance is assured by utilising Intel's Pentium D dual-core desktop processors or a couple of dual-core Xeon processors for future desktop and server machines, but where Intel have really succeeded in recent years is in the mobile market. The Pentium M has been a huge success for Intel, as part of the Centrino brand (see www.soundonsound.com/sos/feb04/articles/centrinos.htm for more information), and in his keynote Jobs mentioned that Intel's projected performance per watt for mid-2006 was over four times higher than that of the Power PC.

Towards the end of the keynote, Jobs invited Intel President and CEO Paul Otellini onto the stage. The latter said of the Apple/Intel arrangement: "The world's most innovative computer company and the world's most innovative chip company have finally teamed up."

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Mac OS X For Intel

Building a computer with Intel's technology shouldn't prove too difficult for Apple's engineers, but one of the most important factors in the transition to Intel-based Macs will be, as Jobs himself put it, "making Mac OS X sing on Intel processors". And here's where more rumours that have been floating around for a while turn out to be true, as the Apple CEO confirmed that "Mac OS X has been leading a secret double life for the past five years", and that "every release of Mac OS X has been compiled for both Power PC and Intel". This should really be no surprise, since OS X's heritage is Nextstep, the operating system for which Apple effectively acquired Next, which ran on Intel processors.

Jobs mentioned an Apple internal guideline stating that "designs must be processor independent and projects must be built for both Power PC and Intel processors", before revealing that the machine he'd previously used in the keynote to demonstrate Dashboard widgets had, in fact, been running Mac OS 10.4 on an Intel processor. It seemed to be working pretty well — although, in many ways, there's no reason it shouldn't. Most modern operating systems, including UNIX, Linux and Windows NT, were either designed or have evolved to run on multiple architectures through modular designs and hardware abstraction. So getting OS X to "sing" on Intel processors turns out not to be such a big deal, since Apple always had the 'just in case' scenario in mind. What *is* a big deal is the way in which third-party developers will deal with the Power PC-to-Intel transition, especially since they've only just got through the move to OS X.

Intel processors... coming fairly soon to a Mac near you.

This third transition finds Mac users and developers in pretty much the same situation they were in 10 years ago, during the move from 68k to Power PC, which many reading this column will remember. The biggest problem, in my opinion, isn't just getting the developers to port their code to the new platform: it's leaving them in a situation where they have to support two different architectures for the same operating system. In the first transition, Apple created what was termed a 'fat binary' that bound together 68k and Power PC binaries into a single package, so that developers could deliver one application to any Mac user. This time Apple have the same idea, except that the package containing both Power PC and Intel versions will be known as a Universal Binary. Developers may remember that porting 68k code to Power PC wasn't always straightforward, but (coming back to the present) Apple have released a new 2.1 update to the company's own *Xcode* developer tools, to make it simple to both port and maintain Mac OS X applications under two architectures.

The Universal Binary concept will really be important to you if you've just purchased a new Mac and want to be confident that it will be supported by Apple and third-party developers. Analysts and news reporters initially questioned whether the Mac platform could deal with another major shift; to counter this doubt, Apple are doing their best to convince everyone that it won't be too difficult for applications to be ported.

During the keynote, Jobs invited Wolfram Research co-founder Theo Gray on to the stage to describe how it had taken one of Wolfram's engineers only around two hours, a couple of days before the keynote, to make the Power PC-based Mac OS X version of *Mathematica* into a Universal Binary that could run under Mac OS X on the Intel platform. And since the keynote many other developers have commented on the speed with which they've got their applications running on Apple's Intel-based development systems. One such developer is Luxology, who managed to port their flagship surface-modelling application, *Modo*, in just 20 minutes. Apple are, of course, to be praised for making it easy for developers, but it's also worth remembering that, with the majority of applications being cross-platform, the source code should already be highly portable.

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Rosetta: Translating The Code

No matter how easy Apple makes the process of creating Universal Binaries (see main text), it's unlikely that every application you run will be available with Intel-native code by the time Intel-based Macs are shipping, especially if one app you rely on is no longer supported or developed, for example. When Apple moved from 68k processors to the Power PC, the Power PC-based Macs included an emulator that could enable 68k applications to run if a 'fat binary' (again, see main text) wasn't available. This worked well for general-purpose software.

For the Power PC-to-Intel transition, Jobs introduced a technology called 'Rosetta', to bridge the gap between Power PC and Intel-based Macs. It allows Power PC binaries to be translated at 'runtime' and be executed on Intel-based Macs. An application running via Rosetta will never be quite as fast as if it were running natively, since the translation process itself entails some processing overhead. This means that those requiring high performance from music and audio software aren't going to find Rosetta too useful, but Jobs showed Adobe's *Photoshop* and Microsoft's *Word* running pretty successfully with Rosetta during his demonstration.

According to Apple's Universal Binaries guidelines, available publicly at www.apple.com/developer, Rosetta is capable of translating applications that can run on a G3 Mac with OS X, and the major restrictions are that it will not run OS 8 or 9 applications, or any code with Altivec or any other G4 or G5-specific instructions.

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Mac, Music & Intel

So what does all of this mean for those running audio and music software on the Mac? Actually, it's probably mostly good news. It's no secret that, in terms of performance and battery life, Apple's current line of Power Books lags behind their Intel-based counterparts, so finally we should get Power Books that can once again live up to their name. And while the current high-end Power Mac offers good performance, as Intel and AMD-based machines move to faster and multiple cores it will be necessary for Apple to keep up with performance, since the hardware will now largely be the same.

In terms of music and audio software companies releasing Universal Binaries, this shouldn't be quite as bad as the process of 'carbonisation' required to port OS 9 applications to OS X. The general application code, such as the user interface and so on, isn't likely to pose a problem, but performance and optimisation are likely to be bigger tasks in some cases, as optimisations for the Power PC — and specifically the Altivec instructions — will require rewriting for the Intel and SSE (Altivec equivalent) instruction sets. Fortunately this isn't so difficult, as Apple provide information regarding SSE equivalents for Altivec instructions in the freely available Universal Binary guidelines.

Many of the major music and audio applications are already cross-platform, so it's likely that optimisations and other processor-specific instructions can simply be adjusted from code that already exists. This should definitely help in companies like Steinberg, Ableton, Propellerhead and Digidesign. And *Logic*'s developers at Apple have plenty of experience in developing Intel-based code! The bottom line is that, with most software being developed on portable cross-platform frameworks these days, Apple are perhaps right in claiming that this transition will be a relatively painless one.

Intel's CEO and President Paul Otellini: "We are thrilled to have the world's most innovative personal computer company as a customer".

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The Mactel Future

In the short term, Apple's move to Intel processors will not have a major effect on Mac users. Analysts have speculated that it might slow Mac sales until the newer Intel models appear, but Steve Jobs made it clear that "this is not going to be a transition that happens overnight". And that's probably a good thing. If you buy a Mac now, you're probably going to have a few good years of use from it before needing to upgrade to an Intel-based Mac. A year from now, Jobs said that Intel-based Macs would be shipping, and they're likely to be Macs that can benefit from the Pentium M chip, such as the Mac Mini, Power Book and iMac. But by the end of 2007 Apple expect the transition to be complete, and the thought of a new Power Mac based on multiple cores using x86 processors is pretty intriguing... SOS

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In this article:

Making The Most Of EQ In Sonar v4

Workshop

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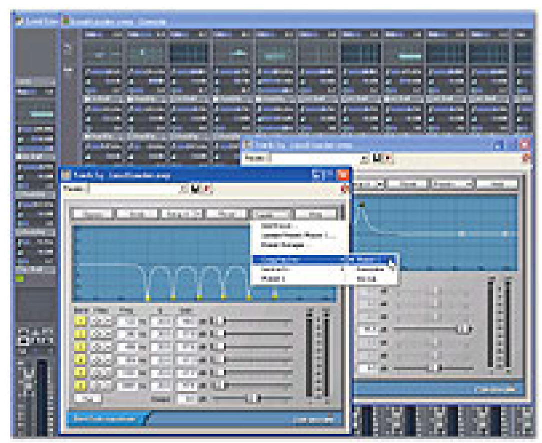
Technique : Sonar Notes

When is an EQ not just an EQ? When it's the *Sonar 4 Producer Edition* Sonitus EQ and you use it for some of the amazing processing tricks we're about to explain.

Craig Anderton



Although the Sonitus EQ bundled with *Sonar 4 Producer Edition* might not seem all that spectacular, it actually has a lot of hidden talents. For example, did you know it makes a great automatable phase shifter? Or models wah-wah pedals? Or that you're not limited to placing EQ pre- or post-effects, but can patch it both pre- *and* post-effects?

Sonar 4 Producer Edition includes per-track EQ that you can access via the Console View or Track View Inspector. With previous versions of *Sonar*, or *Sonar Studio Edition*, you need to insert EQ (or any other effects) manually in a track's FX bin by right-clicking on the bin, going to Audio Effects, then navigating to the desired plug-in. Even if you're using *Sonar Producer*, though, there can be advantages to inserting the EQ plug-in into the FX bin instead of using the per-track EQ... as we'll see in some of the tips that follow.

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Four Stages Vs Six Stages

Even though you see only four stages of track EQ in the Inspector or Console View, the EQ is based on the Sonitus EQ and you can actually use all six stages by 'opening up' the per-track EQ to its full interface View. This also gives you the other advantages of the full interface: you can call up presets, use Undo, have separate settings for Setup A and Setup B, and so on.

It's possible to open up the full EQ View only if the EQ is actually enabled. To enable it you can either right-click on the plot and select 'Enable EQ', or click on the EQ module's power button toward the bottom of the EQ module. Now you can open up the View. In the Inspector or Console View, double-click on the EQ plot and you'll see the full equaliser. The other option is to right-click on the EQ plot and select 'Show EQ Properties'. Note that if you open up several EQs it appears that they replace each other in a single window, but they actually don't do that: each window stacks on top of the other, so you can separate these windows and view the individual EQs if you want to.

One possible problem is that if you make adjustments in band five and/or six of the EQ and then close the interface window, you won't see them in the per-track EQ display and you could forget about the adjustments you've made. However, with respect to track EQ, you can make sure that what you see is what you get by right-clicking on the EQ plot and choosing 'Reset Hidden EQ Parameters'. This turns off stages five and six. If this option is greyed out, it means that stages five and six are not enabled.

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EQ: Before Or After Effects?

Insert EQ into the FX bin, click on it, and drag it into whatever position you want in relation to the other inserted effects.

There are two main ways of dealing with this issue. If you're a fan of the Console or Inspector View, right-click on the EQ plot and choose 'EQ Pre FX' or 'EQ Post FX' — simple. But you needn't be limited to just 'pre' or 'post' effects: you can place the EQ anywhere in a chain of effects, including the middle, so the EQ is both pre- and post-effects. However, you can't do this with the per-track EQ. Instead, insert your EQ in the track's FX bin, along with whatever other effects you want to use. Click on the EQ's label in the bin (dragging on the enable/disable button doesn't do anything) and drag it into the desired position in the chain (see the screen above). Note that the signal flow in the bin goes from top to bottom (or left to right, if you use horizontal rather than vertical FX bins).

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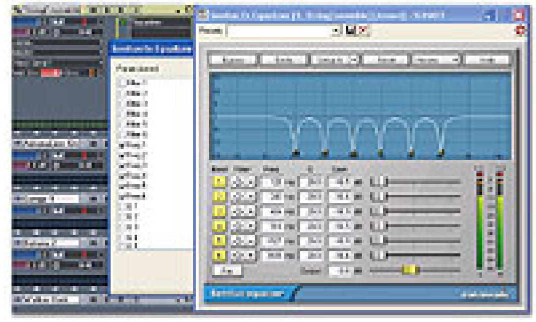
Phase-shifter Effects

The Sonitus EQ is a natural for automated phase-shifting effects. After all, phase-shifting places several deep notches in the frequency spectrum and moves their frequencies up and down in unison. We can do this by linking all of the EQ's frequencies, then automating the response so that they all track together. However, you can't do this trick with the pre-track EQ; you'll need to insert an EQ into the track's FX bin and work with that.

- First, make sure that all six filter responses are set to Peak/Dip. Set Q to 20, but experiment with various values once you get things working. 20 is fairly sharp; the highest Q the Sonitus filter can attain is 24.



The equaliser has been set up for the type of frequency response produced by a phase shifter; the six filter markers have been linked together, then armed for automation in a separate window.

- Now place each frequency about an octave from the adjacent one, starting at 120Hz. Then pull down the gain fader at each frequency so that each notch has the greatest possible depth.

- To create phase-shift effects, we want all the frequencies to move together. No problem: drag a rectangle around all the filter markers (the circles labelled 1-6) to select them; they'll change from yellow to black (selected). When you move one filter marker, they all move together. (Incidentally, you can also Ctrl-click points to select markers, which do not need to be contiguous.)

- Now let's automate these moves. Right-click on the 'Equaliser' label in the effects bin, then select 'Arm Parameter'. This brings up a list of all automatable EQ parameters. Tick the boxes for 'Freq 1' to 'Freq 6', then click on OK.

- Click on the toolbar's Record Automation button, and as you move the linked markers, their motion will be recorded in the track as six separate envelopes. On playback, you'll see the EQ's curve move.

Assuming you have *Producer Edition*, what about using the four effects sliders in the Console view to automate phasing? You can do this, but with the caveat that you'll only be able to shift four of the stages, not all six. You'll also need to specify that the filter markers retain their same frequency relationship (an octave apart) as they're moved. To set this up:

By adding lots of gain instead of taking it away, the phaser configuration can work as

- Start with the expanded EQ interface and edit the band 1-4 frequencies as desired (for example, 250Hz, 500Hz, 1kHz and 2kHz).

a resonator instead.

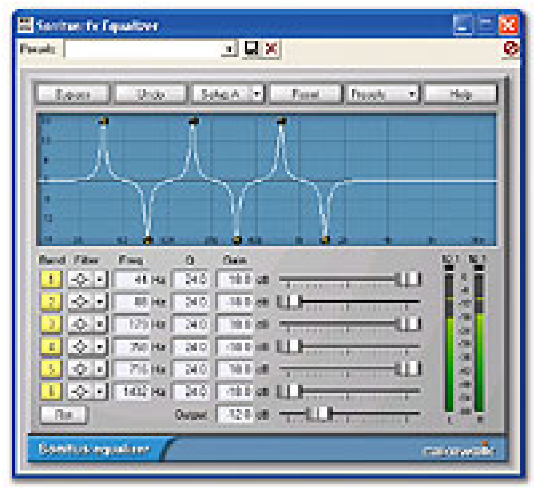
▪ Right-click on each of the four effects sliders and assign it to the desired parameters (Filter Freq 1-4).

▪ Now group all the sliders together: Right-click on each slider, select 'Group', then select the Group letter/color.

▪ Finally, right-click on any slider in the group, select 'Group Properties' and tick 'Relative' rather than 'Absolute' or 'Custom'. This ensures that all frequencies will move relative to each other and remain spaced one octave apart as they move.

A variation on the above technique creates a resonator. This produces a series of harmonically-related peaks that impart a sense of pitch to the material being processed.

▪ As with the phase shifter, the filter type should be Peak/Dip. We want an exact octave spacing between the peaks, so it's worth typing the optimum frequencies into each Freq field (remember to hit return after typing in each one): 100, 200, 400, 800, 1600, 3200. Turn the Q all the way to 24, and gain to maximum as well. Be careful; with all those peaks you'll have to pull the output control way down to avoid distortion. (See screen at top left.)



Yet another variation on the phase-shifter response produces yet another timbre-warping effect.

▪ You use the resonator in the the same way as the phase shifter: select all the filter markers so that they move in unison, then vary their frequencies. The resonator adds a sort of 'tuned' effect, as it emphasises those notes that fall within the response peaks.

As you spent all that time typing in frequencies, why not save this as a preset? Go to the Presets drop-down menu, select 'Add Preset', specify the bank where you want it to live, and you can use the same curve again in other projects.

Before leaving the world of phasers and resonators, I should mention one other response that gives some very interesting sounds: a combination of the phase shifter and resonator settings ('phasenator', perhaps), where there are three peaks and three valleys. The tone resembles some of the vintage, stomp box-type phasers that used lots of regeneration. Because of the peaks, you'll need to pull back on the output level, albeit not as much as with the resonator. (See screen above right.)

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Changing The Display Resolution

The upper graph shows an EQ curve with a ±20dB range. The lower graph shows the ±5dB range selected. Note how it's much easier to see the subtleties of the curve setting with the higher-resolution display.

Here's a basic but highly useful feature: you can change the resolution of the EQ's graph in the interface. It defaults to an amplitude-axis range of ±20dB, but it's possible to right-click anywhere in the graph and choose ±40, ±10, or ±5dB as the range. The finer resolutions are very helpful with subtle settings; I usually use ±5dB, with occasional forays into ±10dB. By the way, if you change the graph's resolution in the per-track EQ, you won't see any resolution change in the channel-strip plot (frequency-response graph). Resolution changes are visible only in the full interface.

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Vintage Wah-wah Emulation

We covered this trick in a more basic way in an early *Sonar* Notes, but that was before the Sonitus effects and the option to tie parameters easily to MIDI control, so it's time to revisit the subject.

The idea of sweeping an equaliser's band-pass filter to create wah-wah effects is not new, but most people find the end result unsatisfying compared to using a vintage wah-wah pedal. This is because in an EQ, frequencies above and below the peak remain at their normal levels, whereas with a wah-wah pedal, response progressively drops off either side of the peak. Fortunately, there's an easy way to simulate that effect.

Wash-wah emulation: Track two has been cloned and flipped out of phase; the equaliser has been set up for a wah-wah type of response, and inserted in the FX bin for Track one.

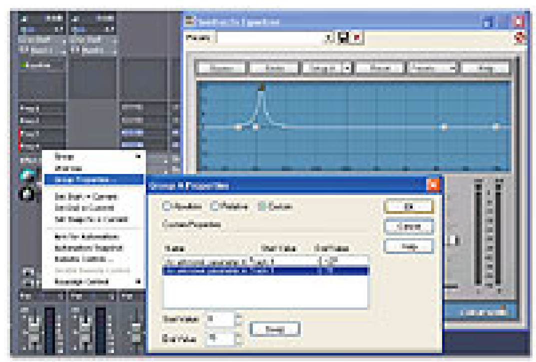
- Start by clicking on the track number (in the Track view) for the track you want to process, then select 'Clone Track'. This creates a copy in a separate track.
- Now reverse the cloned track's phase by clicking on the track's phase button (Ø). If you play both tracks together, they should cancel and you'll hear no sound
- Next, insert the EQ in the FX bin for the track you want to process, then open up the EQ's interface. A good starting point is to disable all the stages except for stage four.

- Set response to Peak/Dip, Q to 7.9, and gain to 15dB or so. Sweep the frequency back and forth, and you'll hear a very cool wah-wah effect. And, of course, it's extremely editable — you can change bandwidth, sweep range, whatever you like.

But what's a wah-wah without a footpedal? We'll assume you have a synth in your studio somewhere that allows for footpedal control, so that when you move the footpedal, MIDI Continuous Controller #4 messages emanate from the synth's MIDI out. We can tie the two together with the real-time effects sliders that appear in the effects section (*Sonar Producer* only). However, there's a complication: you probably don't want the pedal to sweep the filter cutoff-frequency's full range, but only a portion of it. There's a way to do this, although it's a bit of a workaround.

- In the Console view, make sure the real-time control sliders are visible underneath the FX bin containing the EQ plug-in. If they aren't, click on the mixer toolbar's FX button until they appear. They should default to showing (from top to bottom) Freq 1, Freq 2, Freq 3 and Freq 4. If you followed what was suggested above, the filter is being controlled by Freq 4 and all other frequencies are turned off. Move the Freq 4 slider to verify that the peak frequency is indeed controlled by the Freq 4 slider.

- Right-click on the Freq 3 slider, and assign it to a Group. Then right-click on the Freq 4 slider and assign it to the same Group. Moving one should move the other one as well.



The Freq 3 and Freq 4 sliders have been grouped together. The Properties window allows Freq 3's slider to cover its full range, but here you can limit the Freq 4 slider response to cover a narrower range when the two are grouped and moved together.

- Now right-click on the Freq 4 slider, select Group Properties and click on the Custom button. Click on the upper line to select it, then type '0' into the Start Value box, and '127' into the End Value box. This ensures that slider three covers its full range. Next, click on the lower line to select it. To limit the range of Freq 4's slider, type '1' into the Start Value box and '15' into the End Value box, then click on OK. Vary the Freq 3 slider (not Freq 4 — remember, they're grouped together). The EQ's Freq 4 filter marker should move over a restricted range from 175 to about 2500Hz.

- If you want to tweak the lower limit, move the Freq 3 slider all the way to the left (minimum value). Grab the Freq 4 filter marker on the EQ interface and move it where desired, or click on the Freq 4 numerical and drag it to the desired value. Then right-click on the Freq 4 slider, and select 'Set Start = Current'. Now, when you move the Freq 3 slider the Freq 4 filter marker should move over the newly-defined range. However, don't try this technique with the upper limit; for some reason, it doesn't seem to work.

- Now we have Freq 4 moving over a limited range when the Freq 3 slider moves over its full range. The last step is to tie Freq 3 to a MIDI controller. (We don't want to assign the controller to Freq 4, because the MIDI controller is presumably generating the full range of MIDI controller values, and we want Freq 4 to vary over the limited range we specified, which it will do only as part of a Group.) Right-click on the Freq 3 slider and select 'Remote Control'. Move the pedal (or other hardware controller), then click on 'Learn'. Click on 'OK', and now the wah frequency should move over its intended range in response to the hardware controller.

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Still Think It's Just An EQ?

If you really want to push the envelope, bear in mind that some of these techniques just beg to be used in stereo — in other words, clone a track, as with the wah-wah example, but don't invert its phase, and insert an EQ into each track. Set these both to do phase shifting, but have different rates and sweeps going on in the different channels...very psychedelic! SOS

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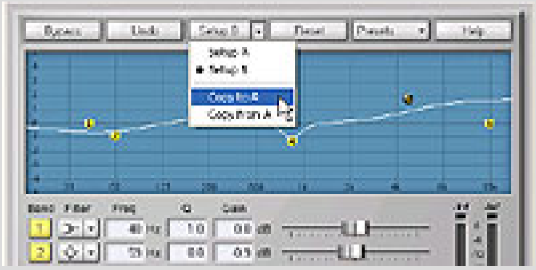
Compare & Contrast

It has always struck me as odd that so few plug-ins have a compare mode, which was common in hardware processors. However, the Sonitus effects have a very useful compare mode, accessed via the effect's toolbar. There are two independent parameter setups for each effect (three if you also count Bypass): Setup A and Setup B. You can toggle between them by clicking on the Setup A/B button.



Setup B is currently selected but, via the drop-down menu, is about to be copied to Setup A as well.

A drop-down menu provides further functionality. Suppose you're comparing two curves, you have curve B selected, and you decide that you like B better than A — but you think you can still do better. On the drop-down menu, select Copy to A and the curve in Setup B will be duplicated in Setup A. Now you can keep tweaking B, safe in the knowledge that the original is safely stored in Setup B if you want to revert to it. Obviously, you can also copy the curve from Setup A into Setup B.

If you change your mind, Undo will work with this function, but (as always) it affects only the last operation. Also note that the Reset function, which flattens the gain at all frequency points, works independently for the two setups, so that you can reset one without disturbing the other.

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In this article:

Practical Groove Quantisation in Digital Performer

Workshop

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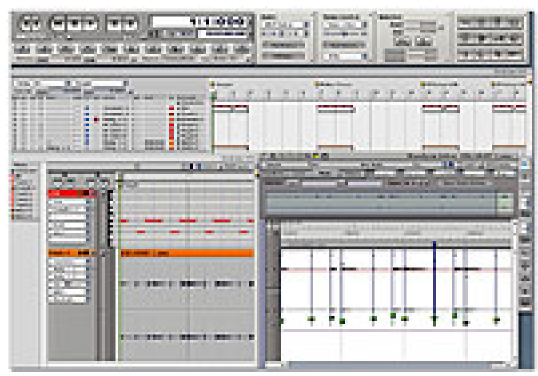
Technique : Digital Performer Notes

This month's *Digital Performer* workshop aims to help you get into the groove more effectively, through the use of *DP*'s sophisticated quantisation facilities.

Andrea Pejrolo



The Track List window, Sequencer window and Waveform editor in *Digital Performer*, an extremely powerful combination of tools for analysing, creating and altering MIDI and audio grooves.

The term 'groove' has historically been related to live performance — that is, a musical situation in which the 'human factor' plays a crucial role. In more recent years, though, the term has migrated from grimy jazz clubs to the more antiseptic environment of the project studio, a transition that's been facilitated by the availability of the 'groove quantisation' feature in some of the leading MIDI + Audio sequencers on the market. While *Digital Performer* initially led the way, competing applications such as *Logic*, *Pro Tools* and *Cubase* have recently caught up by revamping their groove quantisation features and by pushing the limits once again in terms of flexibility and ease of use. The four main digital audio applications on the market all address, in one way or another, how to groove-quantise a MIDI part or an audio part (in some cases, both), but this article is going to address some aspects of how *Digital Performer*, in particular, goes about the job. (For more about the principles of groove applied to quantisation and an explanation of how Groove Quantisation came about, see 'The Birth Of Groove Quantisation' box.)

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Applying A Groove

The latest version of *DP* (4.52 at the time of writing) has once again established a new standard in terms of flexibility, precision, and integration between MIDI and audio groove quantisation. Before going into more detail, though, let's run through how you apply a groove to an existing MIDI region in *DP*. The Groove Quantisation (GQ) function can be chosen by selecting Region / Groove Quantize or by pressing Command G. The GQ window has a simplified version and a more advanced form (see screens right). You can toggle between the two versions by selecting the More/Fewer Choices button.

As with any quantisation action you take in *DP*, you must first choose the region that you want to quantise. You can do so from any editing window, or directly from the main Track List. After selecting the region you're going to quantise, select the GQ option from the Region menu. The preset grooves in *DP* are organised into Folders, Main Categories and Sub-categories. The Folder level is used mainly to sort your grooves in a more efficient way, and the Folders are stored in the Grooves Folder inside your MOTU *DP* 4.5 Folder. You can create your own Folder directly from the finder and organise your grooves according to your personal workflow.

As default choices, *DP* offers DNA Groove and Other Grooves. The former definitely has the more interesting and musical grooves, while in the latter you can find grooves that recreate the feel of legendary drum machines such as the Linndrum and the MPC series.

If you choose the DNA Groove folder, you're presented with a choice of four main categories: Shuffles, Straight, Pushing and Laid Back. Double-clicking on one of the main categories opens up the Sub-categories selection menu. Here you can choose from a list of options based on the same overall groove, each selection providing small variations on the same groove, giving you the highest level of flexibility. Clicking on the More Choices button allows you to select how each parameter of a groove template (timing, velocity and duration) will affect your selected region. Altering a percentage with one of the sliders on the more complex Groove Quantise page changes the balance between the original feel with which you recorded the region and the feel of the groove template you're applying (higher values translate into parts that sound closer to the groove template chosen, while lower values allow you to retain some of the original feel that you played).

Although it's difficult to generalise, and the way you set up these parameters can vary depending on the style and the part you're quantising, I've found that values of around 70 percent for timing and 40 percent for velocity are a good starting point, especially for rhythmic parts such as drums and percussion. You can use the Preview button to temporarily apply the selected groove settings to the current region. Once you're satisfied, click the Apply button.

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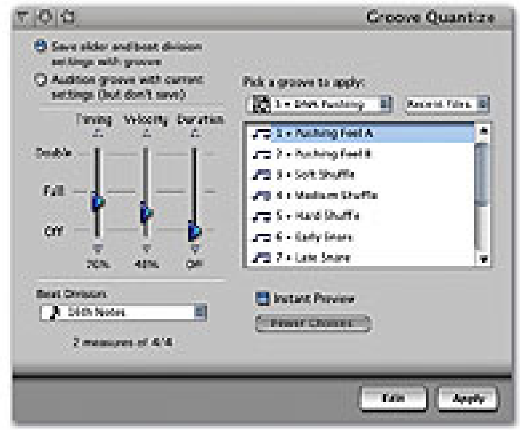
Create Your Own!

Groove quantisation is particularly appealing because it enables the composer to create or extrapolate, his/her own grooves from MIDI and audio parts. While the ability to create a groove template from a MIDI part has been available for a long time in *DP*, the ability to do the same from an audio part came as a new addition with version 4.5.



The simplified (left) and full-strength versions of the Groove Quantise window. The simple window allows you to choose and apply a groove and variation feel, while the more complex one (accessed via the More Choices button) also offers sliders for varying how much of the groove's note Timing, Velocity and Durations will be applied to the region you're quantising.

Creating a groove template from a MIDI part is fairly simple:

- Select the MIDI region from which you want to extrapolate the groove (a two-bar or four-bar selection usually works very well).
- Now select Region / Create Groove.
- In the Create Groove window select New File, in order to create a new main groove category (for example, Swing Grooves).
- Open the category just created, by double-clicking on its name, and use the 'New Groove Name' field to assign a name to the new groove (see the screen overleaf). Click OK.
- From now on, the groove just created will be available from the GQ window.

While this technique allows you to expand your groove library, in reality it represents only a small improvement over the bundled presets. Custom groove quantisation gets really interesting when audio material is used as the source for grooves. The technique involves, first, the analysis of a soundbite containing the groove, then the extrapolation of the groove itself, and is particularly useful not only because it allows you to expand your groove library, but also because it can be used as an analytical tool for studying the details of a particular groove or live performance. Let's first take a look at how we can create our own grooves from audio material.

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The Birth of Groove Quantisation

The original concept of groove quantisation and how it might be applied to a MIDI performance was developed by acoustic researcher Ernest Cholakis in the mid- and late 1990s. In his studies, Cholakis analysed the recordings of professional studio musicians (mostly drummers and percussionists) in order to be able to describe, in a rational and objective way, the essence of their grooves. The map on which these groove descriptions are based uses three main parameters: timing, velocity and duration. By altering slightly the relationships between each of these parameters over time, a musician is capable of building a groovy performance that is peculiar to his or her own playing style. Cholakis was able to 'can' the grooves of several musicians and extract their styles using mathematical algorithms. The grooves created are called DNA grooves and they can be bought from the Canadian company Numerical Sound (www.numericalsound.com). Basic DNA grooves have been bundled, with *Performer* first and with *Digital Performer* later, for many years. These basic bundled grooves include such general categories as Pushing, Laid Back, Straight and Shuffle, but keep in mind that, for each of these categories, you have the option of choosing from several additional variations, such as the one shown in the screens above.

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Engineering The Groove

The Create Groove window is where you can save the feel of a selected MIDI part as a new groove template.

DP 4.52 has a very sophisticated engine under the hood that enables you to analyse a soundbite in terms of tempo and hit-points — or 'beats', as *DP* calls them (a beat usually corresponds to a specific transient of the waveform that is being analysed). This terminology might sound familiar if you've used applications such as *Recycle* from Propellerheads. Once DP is 'aware' of the beats (transients) present in a soundbite, you can, for example, quantise the audio without having to slice the soundbite into separate parts, or easily adjust the tempo of the soundbite to the tempo of the sequence (and vice-versa), and, of course, create a new series of grooves based on the positions of the detected beats. I find the last capability the most groundbreaking and fascinating of them all. Can you imagine being able to extrapolate Peter Erskine's swing groove, or the famous hi-hat swing pattern played by Jimmy Johnson of the Duke Ellington Band? That's exactly what we'll do in the following examples. In addition, wouldn't it be great if we could compare the grooves from these two famous drummers and analyse how they interpreted the concept of swing? We'll do that, too, so read on!

The first step we need to take in identifying the beats inside a soundbite is to find the soundbite's tempo. This is a procedure that's probably familiar to many *DP* users, as it was introduced in earlier versions of the software. Although the operation is not strictly necessary in order to identify the beats inside a soundbite, it will definitely increase the accuracy of your final identification, so I recommend including it in your to-do list. To find the tempo of a soundbite:

- Select an exact number of bars (trim the soundbite so that it is exactly two, four or eight bars in length).

- Select the soundbite, then choose Audio / Soundbite Tempo / Set Soundbite Tempo.

- In the Set Soundbite Tempo window (see screen top right), input the Length according to how many beats and ticks you selected (for example, if you selected a two-bar loop in 4/4, input 8/000).

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About The Author

Andrea Pejrolo is a musician and music technology professor, composer, arranger and bass player. He teaches at both Boston's New England Institute of Arts and Berklee College of Music and has extensive experience of film and television scoring. His book, *Creative Sequencing Techniques For Music Production*, is published by Focal Press.

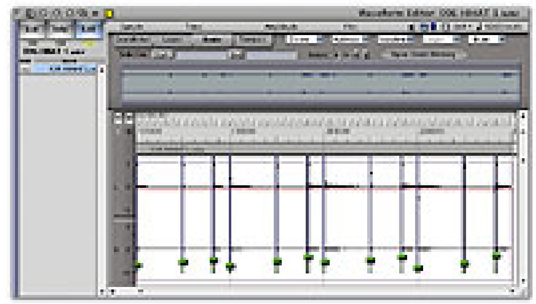
- *DP* will automatically show the tempo of the soundbite in the lower-right corner of the Set Soundbite Tempo window, and it will stamp it into the header of the audio file so that this particular soundbite will carry the tempo information across multiple *DP* projects.

- Once *DP* is aware of the soundbite's tempo, you can move on and start identifying the beats. First, select the soundbite.

- Open the Waveform Editor by selecting Audio / Edit in Waveform Editor. If you're a seasoned *DP* user you probably know that you can do several things in the Waveform Editor, most of them based on destructive editing. In order to analyse the transients of the soundbite and identify its beats, select the Beats tab at the top of the Waveform Editor window (see screen below).

You should now see a series of vertical blue bars running across the selected soundbite. These lines identify the transients/beats that *DP* automatically detected. The green handle linked to each beat represents the relative velocity/volume of each beat. To change that, simply click and drag up or down with the mouse on the corresponding handle. (Keep in mind that this won't alter the actual volume of the transient in the waveform.) You can quickly adjust the threshold used by *DP* to identify the beats by selecting

Setting Soundbite tempo is the first step recommended in the process of placing beat markers inside a soundbite. In this window you can tell *DP* the length of your selected soundbite, in beats and ticks. The example shows a soundbite of two bars in 4/4, or eight beats. *DP* calculates the soundbite's tempo using this information.



In the Waveform editor, open the same soundbite whose tempo you've just set and choose the Beats tab at the top of the window. The vertical blue lines you now see are the 'beats' (transients or places of rhythmic stress in the music) that *DP* has identified automatically. They can be moved manually if necessary, by dragging on the green 'handle'.

Adjust Beat Sensitivity from the Beats menu in the top area of the window. The Adjust Beat Sensitivity slider (see top screen, opposite) can be used to fine-tune *DP*'s beat-detection engine for the selected range of the soundbite.

Lower values will instruct *DP* to mark as beats only transients with higher amplitudes, while higher values will allow *DP* to select even transients with lower amplitudes. While the slider gives you control over the general transient/beats ratio in most situations, you will need to manually remove or insert beats in order to achieve a higher level of accuracy, but *DP* makes it extremely easy to manually change the beats detected. For example, you can suspend or reactivate a beat by simply option-clicking on its handle at the top of the window. To completely delete a beat or a series of beats inside a selected range, use, respectively, the Delete Beat and Delete Beats in Selection functions, found in the Beats menu. To move a beat and adjust its position relative to a transient, simply click and drag it left or right.

Once you're happy with your beats assignment, you can close the Waveform Editor window. At this point, *DP* 'knows' everything about your soundbite, its tempo and its beats, and you can use this information to accomplish a series of tasks that take full advantage of the new audio engine. For example, you can extrapolate the groove from the selected soundbite, save it as a groove quantisation template and subsequently apply it to a MIDI part, or apply an existing groove to the soundbite without having to slice the soundbite into separate regions.

To create a new groove template from a soundbite that has been analysed, simply select the soundbite and choose Region / Create Groove. At this point, the procedure is similar to that used for groove creation from a MIDI part. You can create a new groove folder, Category and Sub-category by navigating though the Create Groove window. The new template created will be accessible through the usual Region / Groove Quantize menu.

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Quantising & Groove Quantising Audio Parts

As mentioned above, a really exciting new feature of *DP* 4.5 is the ability to quantise (and groove quantise) audio soundbites without having to slice them first. This is accomplished by the use of sophisticated algorithms, based on time compression/expansion, that can be applied in a discrete way to each individual

The Adjust Beat Sensitivity slider, from the Beats menu in the Waveform editor, allows you to adjust how many beats *DP* detects automatically, by changing its amplitude threshold.

beat inside a soundbite. To quantise an audio soundbite, select the soundbite then choose either Region / Quantize or Region / Groove Quantize, depending on the type of quantisation you want to apply. If you choose the former option, make sure to select 'Beats within Soundbites' under the 'What to quantize' option in the Quantization window. When you're quantising the beats of a soundbite, *DP* will adjust the position of the transients of the audio file according to the quantisation option, and it will time-compress or expand the soundbite in order to make up or reduce the space between transients.

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How to Quantise Soundbites

In earlier versions of *DP* (as well as in version 4.52) you have the option to quantise the beginning of each soundbite. This function is very similar to the one found in *Recycle*, where a sound file can be separated into different 'slices', according to its transients. Each transient then becomes a separate soundbite that can be moved and quantised independently from the others. In *DP* you can easily 'slice' an audio file into separate soundbites by using the Scissors tool from the Tools window (bottom tool of the tool bar shown on the right).

- Open the Graphic editor (double-click on the audio track you want to edit from the Track List window), open the Tools window (Shift-O) and select the scissor tool.
- You can either create slices using Unit mode (the slices will be inserted only according to the Unit value) or without Unit mode (you can slice your audio file freely). If you need to automatically slice the audio file at regular intervals (let's say every 16th notes), it's best to set 'Unit' to the desired value (in this case, 16th notes) and press the option key while clicking and dragging across the file. A new slice will be inserted every 16th note.
- Quantise the soundbites/slices you've just created by choosing the normal quantisation function, making sure to set the 'What to Quantize' parameter to 'Soundbites'.

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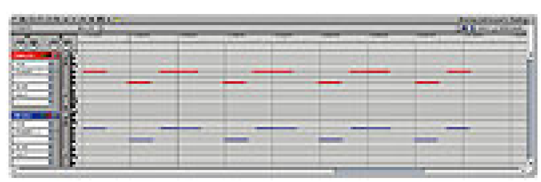
Compare & Analyse Grooves



Comparing the 'feel' of a Peter Erskine drum groove (A) with that of a Jimmy Johnson groove (B), using *DP*'s groove facilities.

An intriguing use for this type of technology is as an analytical tool for comparing grooves, or for showing (to a class, for example) how musicians interpret grooves in different ways. To demonstrate this alternative use of *DP* 4.5 groove extrapolation/quantisation, I analysed two grooves: one extracted from a Peter Erskine drum library, in which he plays a hi-hat swing groove (pattern A in the screen above), the other from a hi-hat pattern played by Jimmy Johnson in the 1959 *Anatomy of a Murder* recording with the Duke Ellington

Band (pattern B). In order to analyse the two patterns (played originally at different tempos), I first extrapolated the two grooves, using the beat-detection technique explained above. I created two new quantisation templates based on the beats of each groove, and then inserted two straight 8th-note two-bar patterns on two separate MIDI tracks and applied each of the two groove templates to a different pattern, basically transforming the real groove into a MIDI part. This step was necessary in order to level out the differences in tempo between the two original performances. The results are two two-bar MIDI patterns that can be adapted to any tempo, and that can be compared easily and accurately to each other in terms of where the events happen in reference to bars and beats. In the screen above you can see how Jimmy Johnson's feel is slightly 'ahead of the beat' compared to the feel of the groove played by Erskine. This technique can be applied to other situations where the analysis of a groove is needed.

As you can see, a sequencer and its features can provide a number of new techniques and tools that can greatly improve your sequences and your creative experience. If you explore and let your curiosity guide you, you'll be amazed at the innovative approaches you can evolve. SOS

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SOUND ON SOUND
The World's Best Music Recording Magazine

In this article:

Pro Tools Notes

News & Updates

Published in SOS August 2005

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Technique : Pro Tools Notes

This month's column brings you news of a new hardware DSP system from Waves, a mini-review of a keyboard sticker set for *Pro Tools* from Editor's Keys, the latest information on compatibility with Mac OS 10.4, and plenty of plug-in news.

Mike Thornton

Audio Ease's *Make A Test Tone* is a Mac OS X utility that can generate test tones for many situations, including sine sweeps and signals to help you check the phase of your speakers.

Perhaps the biggest news for many *Pro Tools* users this month will be the announcement of Waves' Netshell APA processing units. These are hardware devices that let you easily run multiples of the most CPU-intensive plug-ins from the Waves stable. For example, the rackmounting APA 32 lets you run six *IR-1* convolution reverbs, or nine *Linear Phase Equalizers*, or 12 *C4 Multiband Parametric Processors* at 44.1kHz. The APA 44-M is smaller and more portable than the APA 32, yet gives up to 30 percent more plug-in power, and the good news is that neither of the APA units requires a PCI card or a Firewire or USB connection. You simply connect them to a standard Ethernet port on your computer and install the new Netshell software, whereupon you will find that you will have the choice to run plug-ins on your normal system or on the APA. If you need even more power, you can add more units up to a maximum of eight APAs on one computer using standard Ethernet switchers and cables. You can mix and match the APA 32 and APA 44-M units within one system if you like.

You can easily take an APA 32 or APA 44-M to another system. Just plug in your

Waves-authorised iLok, connect the APA unit to the computer's Ethernet port, install the Netshell software, and you will be able to use any of your authorised Waves plug-ins on that system.

The APA units are designed to take the load off your computer when running the most CPU-intensive Waves plug-ins — Waves say that for basic plug-ins like compressors and equalisers, the system overhead involved in routing audio to and from the APA unit would outweigh the tiny gain in CPU resources on the host system. The following plug-ins can be run on the APA 32 and 44-M: *L3 Multimaximizer* * *L3 Ultramaximizer* * *IR360 Surround Reverb* (Mac only) * *IR-1 v2 Convolution Reverb* * *IR-L Light Convolution Reverb* * *Linear Phase Equalizer* * *Linear Phase Multiband* * *C4 Multiband Parametric Processor* * *Renaissance Reverb* * *Renaissance Channel* (no external side-chain) * *Sound Shifter* * *Morphoder* * *Trans X Multi* * *Q-Clone*. For an introductory period, the *IR-L* convolution reverb and *Q-Clone* EQ emulator will be included free with each APA. The APA 32 will cost £1199 including VAT with the APA 44 retailing at £1799, and both should be available by the time you read this.

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IK Upgrade Offer

For a while now, IK Multimedia's *Sampletank SE* sample-playback plug-in has been included as part of the Music Production Enhancement Suite software bundle that comes with all new Digidesign Pro Tools HD and Pro Tools LE systems. It's now been upgraded to version 2, and existing registered owners of *Sampletank SE* can download a free update at www.sampletank.com/digist2free.html. *Sampletank 2 SE* has some new options and features, including three different synthesis and sample-playback engines for playable instruments and loops, and 32 high-quality built-in DSP effects. IK Multimedia continue to offer a discounted upgrade route to the full version of *Sampletank 2 XL* from the free version bundled with every Pro Tools system. Registered users of *Sampletank 2* can also purchase a crossgrade to the company's *Sonik Synth 2* for only $249, a saving of $150.

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Testing, Testing

Make A Test Tone from Audio Ease (above) is a Mac OS X test signal generator that you can use to test your signal chain for distortion, aliasing, speaker defects, digital clock errors, crosstalk, cross-wiring and so on. It will also help you produce the classic test tones to precede programme material. Sine-wave tone generation ranges from simple test tones to precede a mix to flexible custom level, and frequency sweeps up to 192kHz. In the files, Audio Ease have an option to include a virtual announcer, 'Jennifer', who can speak introductions containing levels, frequencies, speakers and formats. In addition, *Make A Test Tone* can add markers in the sound files to indicate the exact frequency and amplitude at that point.

Make A Test Tone also can produce phase-check sequences from stereo up to 10.2 formats. Jennifer introduces each speaker, after which its phase correlation is tested against the previous speaker. All your connections can be checked in under a minute! All files are native to *Pro Tools*, so they can be imported without conversion.

Audio Ease offered the first 500 downloads free, and nearly 100 went within the first minute! However, the full paid version only costs $29.95, and can be found at www.audioease.com.

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Burning Bright(ish)

The latest upgrade to *Pro Tools* itself, version 6.9.2 adds support for Mac OS 10.4 'Tiger' and is available as a free download for qualified users. Updater CDs will be available soon through Digi Store for £5.00 plus shipping and handling for those who are unable to download large files. At the time of writing, the version 6.9.2 software is available for Pro Tools HD and LE hardware systems, with a Tiger-compatible version of *Pro Tools M-Powered* due to follow shortly. For up-to-date information, visit the Digidesign web site. Digidesign Engineering and Test Services have also completed testing and qualification of *Pro Tools* TDM 6.9.2 with Mac OS 10.4.1, so that OS version is now listed as compatible along with Mac OS 10.4.

There have, however, been some issues with *Pro Tools* and Tiger. First, two of the key commands that *Pro Tools* uses to initiate recording are assigned by default to Mac OS 10.4's Spotlight and Dashboard features. In order to use these key commands for *Pro Tools*, the key commands for Spotlight and Dashboard must be remapped. To remap Command + Space, open System Preferences, select Spotlight and untick the box for the Spotlight menu keyboard shortcut, or click in the box and choose a different shortcut that doesn't conflict with *Pro Tools*. To remap F12, open System Preferences, select Dashboard & Expose, choose F12 from the menu next to the Dashboard item and pick another function key or select any combination of modifier keys and a function key.

Another Tiger issue is that unmounting a removable drive from the workspace will prevent any subsequent drives from being mounted in the workspace, even though these subsequent drives will show up in the Finder. The workaround is to restart *Pro Tools*. After *Pro Tools* has launched, the drives will appear in the workspace.

Some users have also discovered that, in certain situations, the *EQ III* plug-in will not recall its settings at 192kHz sample rates: specifically, with a Q of ±2.0 and Gain of ±12db and High and Low bands set to notch (peak). Under these circumstances, the parameters will revert to 2.0 and 12.0 respectively after saving and reopening the Session. There is a workaround to help in the short term which is to use automation to set High, Low, Notch and Shelf settings to jump to their desired Qs and Gains at a point in the timeline before the song starts.

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At Your Fingertips

This month, I've been using a new keyboard sticker set for *Pro Tools* from Editor's Keys, which converts a standard QWERTY keyboard into a *Pro Tools* audio editing keyboard. Each key sticker is applied to your standard keyboard using their applicator, and contains the program shortcut, icon, key letter and also a text reference telling you what each shortcut does. The set is made up of 70 key stickers. I found them very easy to apply and I much prefer the Editor's Keys set to the Digidesign set, for several reasons. Firstly the normal letters are in black and are much easier to see then the light grey letters on the Digidesign set, which is especially useful for those of us who can't touch-type! Secondly, some of the icons are in colour, and all are clearer and easier to understand. Finally, the size and shape of the Editor's Keys stickers mean they fit most keyboards, whereas the Digidesign ones depend on an extra text section to be fitted on the side of each key, and not all keyboards have sufficient space for this. The only shortcoming of the Editor's Keys key set is that they have missed out some of the key functions on the numeric keypad, notably '3' for record as well as the '+' and '-' Nudge keys and the '=' , '/', '*', '.' and 'Enter' keys.

At the time of writing, the Editor's Keys *Pro Tools* set is on special offer for £8.99 and can be found at www.editorskeys.com.

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Drums On A Budget

A-Kit is a set of seven Mac OS X plug-ins designed to form a virtual drum kit. The plug-ins are RTAS-compatible and are triggered from MIDI tracks to play back the included sampled sounds. A 44.1kHz sampler is incorporated in each module of *A-Kit*. The total sound bank contains 45MB of samples, with up to 40 samples of each instrument. Each module of *A-Kit* can be inserted on its own track, allowing you to add filters, compressors and effects to each element of the 'kit'. The best bit is that *A-Kit* only costs 15 Euros.

W www.un-art.org/Archie/a-kit/a-kitEN.html

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TC News

TC Electronic have announced an update to version 2.0 of their *Restoration Suite*, which includes a new *Dethump* plug-in. This plug-in adds another feature to the *Restoration Suite* by resolving errors up to 12,000 samples in length without audible artifacts. *Dethump* is designed to eliminate low-frequency pulses that typically occur with strong impulse disturbances in LP recordings. With the new 2.0 update, *Restoration Suite* includes five real-time plug-ins: *Descratch*, *Denoise*, *Declick*, *Decrackle* and now *Dethump*, which can be used in VST, AU anf RTAS-compatible systems. The upgrade will be free of charge for existing users via the TC web site.

TC Electronic have also announced that System 6000 stereo plug-ins will now be available for Pro Tools HD, meaning that some of the most sophisticated algorithms around for mixing and mastering are finally available to the Pro Tools platform. In addition, they have unveiled a new reverb, the *DVR2 Digital Vintage Reverb*, which is an emulation of a well-preserved EMT 250. SOS

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In this article:

Reason Notes

News & Tips

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Technique : Reason Notes

Tiger tussles with *Reload*, Propellerhead's Akai sample-management software, the excellent free *Electromechanical* Refill gets an update to show off the powers of the *Combinator*, and we've patch browsing and *Stereo Imager* tips to offer.

Derek Johnson

Two *Combinators* loaded with patches from *Electromechanical* 2.0. Note neat graphics stuck to the right of the device's Controller Panel.

Whenever Apple upgrade their operating system, many of us hold our breath as we wait to see if our favourite applications are compatible or in need of their own update. This is especially the case when the company go for a major increment such as the new Mac OS 10.4, 'Tiger'. And because the new OS ushers in a number of changes and enhancements to Core Audio and Core MIDI, we need to be doubly reassured before we hand over our cash to Steve Jobs. The last issue of *SOS* covered Tiger in depth, and no doubt this and future issues will continue to do so as appropriate. Here I'll just pass on the news from Propellerhead that *Reason* v3 is fully compatible with MacOS 10.4. There are no known compatibility issues: during the development of the latest version of *Reason*, it was continuously tested with pre-release versions of Tiger.

The news is good elsewhere in the Propellerhead software family, with *Recycle* v2.1 also fully Tiger-ready. And then there's *Reload*, Propellerhead's neat and free (to registered *Reason* and *Recycle* users) utility for converting Akai-format sample CDs into something *Reason* can load. Apparently, *Reload* v1 does work with Tiger, but a minor anomaly, related to inserting Akai-formatted CDs into a

Mac, has surfaced. Propellerhead are on the case, though, and a solution is on its way: v1.0.1 of *Reload* entered beta-testing at the end of May.

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Quick Tips

Reason 3's new patch browser is so flexible that there's even an option to browse all synth and sample-based patches and add a device to the rack as its patch is loaded. An extension of this option is the fact that *all* patches can be viewed in *any* device's pop-up. Say you're using the pop-up to check out patch names in a given Refill from an *NNXT*'s patch selector. Up pops a patch for the *Combinator* that you'd like to play. Just click on it, and the *NNXT* is replaced by a *Combinator* holding the new patch. How convenient. Under Windows XP, you may find it helpful to see file extenders in the pop-up list, but on both PC and Mac platforms each patch has the device's icon graphic at the front of the file name. Once you're familiar with these icons (or the file extenders) you'll know exactly which file types you're looking at.

I hope you're not neglecting your new *MClass* 'mastering' processors — especially the *Stereo Imager*, a tool that can add a very nice subtle sparkle to your mixes. Just remember that there's not a lot of meaningful stereo information in the very low-frequency range. In fact, you might find a mix becoming tighter and more focused if you turn the Lo-Band width control fully to mono, just slightly widening the Hi-Band. The basic 787Hz crossover frequency already set is fine for most purposes, but don't hesitate to mess about with it, especially if you're widening different parts of your *Reason* mix separately (as you might do with drums or extreme stereo delay effects, for example). Set a crossover much lower than that and you may find the mud creeping back in. But if that mud does something you like, let it stick!

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Electromechanical Revisited

Reason 3's new features — mainly the *Combinator* — have been the catalyst for third-party developers to produce new or updated Refills, and now we hear of an updated version of the *Electromechanical* Refill that was released early last year, before most of us were aware that v3 was on the horizon. *Electromechanical* 2.0 remains a free download to registered *Reason* users (from www.propellerheads.se), but for those without a really fast Internet connection, or those who would just like to have the Refill on CD, it can be purchased from Propellerheads for just £7 plus £2.81 airmail postage to the UK.

And what a download it is: 106MB of excellent samples and patches from a range of classic instruments: electric pianos — Wurlitzer valve EP100 and solid-state EP200, Fender Rhodes MKI tube and MkII solid-state — plus Hohner's Pianet T and Clavinet D6. The collection even includes a Model A, the grandaddy of Hammond organs. The NNXT's excellent multisampling and velocity layering are shown off rather well, and performance tweaks abound — just move that mod wheel! The collection is largely the same as the original release, but where the original showcased the capabilities of the *NNXT* sample player, *Electromechanical* 2.0 highlights the *Combinator*. If you already have experience of the original *Electromechanical* release, you'll recall Song setups that added stylistically suitable effects to various *NNXT* patches. In the updated Refill, setups of this type make up the contents of individual *Combinator* patches, offering *NNXT*s processed by complementary effects chains. The custom graphics (an arty image of the sampled instrument is 'stuck' onto an otherwise plain *Combinator* background) are fun too. No *Reason* user should be without this Refill! SOS

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Reason UK Tour

Reason 3 is on tour in August. Confirmed dates are: 8th, Sound Control, Glasgow; 9th, Sound Control, Edinburgh; 10th, Boomerang Sounds and Sound Control, Salford, Manchester; 11th, PMT, Birmingham; 12th, Absolute Music Solutions, Poole; 13th, Sound Control, Virgin Megastore, London; 15th, Temple Bar Music Centre, Dublin; 16th, Sonic Arts Centre, Belfast; 18th, Sound Control, Southampton; 19th, Digital Village, Acton, London. Check www.propellerheads.se or www.maudio.co.uk for more.

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SOUND ON SOUND
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In this article:

Recording Electric Guitars In Logic

Workshop

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Technique : Logic Notes

Find out how to record a great electric-guitar sound into *Logic*, and which of the plug-ins have the most impact at mixdown.

Paul White

Photo: Richard Ecclestone

This article effectively covers two areas — the basic methods of recording the electric guitar and some techniques for recording and processing that are specific to *Logic*. So, if you're not a *Logic* user, the majority of this article will still be useful to you, and most of the *Logic* part can be applied to other software platforms. At one time the only accepted way to record the electric guitar was to mic the amplifier, but now there are guitar recording preamps (both analogue and digital), active and passive speaker simulators, and software modelled guitar preamp plug-ins to choose from. There's also the option of buying a speaker cabinet in a soundproof enclosure, which you can connect to your own amplifier and then mic up conventionally (the mic goes inside the box with the speaker, just in case you were wondering!) without allowing more than a whisper of sound to leak out into the room. Which option you choose depends both on your recording situation and on the sound you wish to achieve. Most players agree that a nice amplifier, properly miked, still gives the best results for traditional electric guitar sounds, but some of the digital emulations now get very close indeed. Interestingly, we tried all these options recently during a seminar that we put on for the London Guitar Show and the results were most enlightening.

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Recording Tips

If you are going to mike up an amplifier, it obviously needs to be a nice-sounding amp, and for most discerning players that means a well-maintained tube amplifier of some kind. Unless you have a very large recording space, a smaller amplifier is easier to mike up than a large one and the 6W Cornford amp that Dave Lockwood played through at the guitar show proved to be so loud that we still needed to use a power soak with it to cut down the level reaching the speakers. If you have a larger amplifier than this, and most people have, then a power soak between the amp and speaker is a good way to keep the level down while still allowing your amplifier to work hard.

While it's easy to suggest mic types for vocals or specific acoustic instruments, you can try just about anything you have on a guitar amplifier and get an interesting tonality. The old standby is a Shure SM57 or Sennheiser MD421 dynamic cardioid, but you can try literally any dynamic or capacitor mic you have in your locker, from the cheapest to the most esoteric. Often where you place the mic makes more difference than what the mic is, and sometimes a mic that sounds dreadful on vocals can sound very musical on guitar.

You'll notice that the sound becomes more focused as you move the mic closer to the speakers, and it also gets more mellow as you move away from the centre of the speaker towards the edge of the cone. You can also turn cardioid mics slightly so the sound is hitting the mic off axis if you need less top end. In most studio situations, cardioid mics are used to reduce the amount of spill or room interaction, but as guitar amps are relatively loud and the mics generally set up pretty close, you can also try omnis or figure-of-eights if you have them and the spill shouldn't get much worse.

Placing your guitar's amp on a chair or stand to raise it off the floor will change the recording, because of the different way the sonic reflections combine with the direct sound.

Although the traditional rock approach is to put a dynamic mic right up against the grille, you can often get a better sound by backing off from the speaker slightly, just by a few centimetres. I've also had good results miking the back of an open-backed cabinet or, in a decent-sounding room, putting the mic up to a metre away from the front grille. In this latter case, placing a reflective board on the floor between the amp and mic can liven up the sound in a very useful way. You'll also find that the sound changes depending on whether the amp is on the floor or on a stand (or chair), as the floor reflections will interact in a different way. Open-backed cabinets tend to have a 'bigger' sound than closed ones, as the speaker doesn't have a cushion of air to damp it, so low-frequency sounds seem more pronounced. Where you have the facilities, you could also try combining the outputs from two mics, one close, and the other further away, or one in front of the amplifier and one behind.

Of course, using two mics on any source raises the issue of phase. If your mixer has a phase switch, listen to see what difference reversing the phase of the distant mic makes, as this will affect the way the sounds combine — you can also vary the mic distance to adjust the relative phase of the two mic signals. If you want to be purist about it, you can use *Logic*'s *Sample Delay* plug-in to delay the close mic so that it is in phase with the distant mic. Sound travels at roughly one metre every three milliseconds, so to add three milliseconds of delay at a 44.1kHz sampling frequency you'd need to add a delay of about 132 samples. You can fine-tune the result by ear to see what value gives the most solid sound.

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Speaker Simulators

Where the sound of a cranked guitar amplifier would cause problems with spill (or neighbours!), a good combined power soak and speaker simulator will allow you to DI your guitar amp without the speaker connected and still capture something very close to its natural tone. However the results vary drastically from model to model.

A typical speaker simulator comprises a reactive dummy load, allowing the amplifier to work normally, followed by circuitry that approximates the filtering effect of a guitar loudspeaker. Apart from the dummy load, which is, of necessity, passive, the filter circuitry that replicates the speaker's frequency response may either be passive or active. The output appears as either a mic- or a line-level signal, which can be plugged directly into a mixing console. Most power soaks can handle between 50W and 100W of input power, which means that the majority of guitar amps can be run flat out to get the best overdrive sound.

Isolation cabs can work exceptionally well and also allow you to experiment with different microphones, though the tonality of the speaker in the cabinet may not exactly match the one in your amplifier. Nevertheless, this is a very practical way to work where you wish to retain the essential character of your amplifier, even though a little EQ may be needed to get closer to the sound of your own speakers. Using power soaks or speakers inside isolation cabinets falls somewhere between true amp miking and the 'short cut' world of digital amp emulations.

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Digital Modelling

Although there are still some excellent analogue recording preamps made for guitar, digital models are currently the most popular and arguably the most versatile. Both are used in essentially the same way, by DI'ing their outputs at line level, though with an analogue model you'll probably have to add your own delay, chorus, and reverb effects afterwards if you need them.

Some of the digital emulations are incredibly good, and very versatile tonally. If anything, they miss out on capturing the real low-end thump of a close-miked tube amplifier, but they can get pretty close to the sound of a range of real amplifiers, and the better ones also respond well to playing dynamics, such as picking intensity or backing off the guitar's volume control. *Logic Pro* now includes a very simple but useful guitar preamp plug-in, and an even simpler version (lifted from *Garage Band*) is included with *Logic Express*.

The only technicality to consider when recording a guitar for processing via a software plug-in is that guitars need to be fed into a high-impedance instrument input, not a mic or line input, so if your audio interface doesn't have one of these you'll need to buy an active DI box with a high-impedance input, and then connect this to the mic input of your audio interface (or the mic preamp connected to your audio interface). Most active DI boxes can be run from a phantom-powered mic input to save on batteries. You'll also need to set your system latency (buffer size) to its lowest stable value if the player is going to monitor the sound with *Logic*'s effects and amp modelling added.

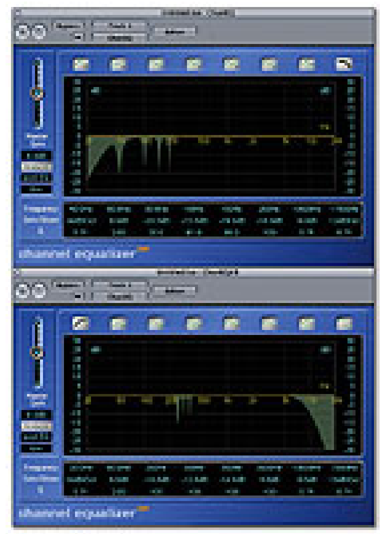
If you aren't able to record with a real guitar amp for whatever reason, there are at least two other options available: you can use one of the large range of modelling guitar preamps on the market (such as the Vox Tone Lab or Line 6 PodXT shown on the left), feeding the output into your audio interface's line inputs; or you can use a DI to feed the clean electric guitar signal into your audio interface's mic input (right), relying on the amp and speaker simulation built into *Logic*.

The only realistic way to evaluate a modelling guitar processor or plug-in is to listen to the sound over the studio monitors and see how it stacks up against the guitar sound you hear on records in a similar style. You can't expect to get the same listening experience sitting in front of studio monitors that you get standing in front of a 100W stack, because the volume level when you play back a record is very different to what the guitarist hears at a live gig. This may sound obvious, but it's surprising how many guitarists comment that the sound isn't as big or powerful as it is when they're actually playing.

Electric guitars don't have great signal-to-noise ratios, especially those fitted with single-coil pickups which are prone to receiving hum from surrounding wiring and equipment. In particular, CRT-type computer displays and TV monitors affect guitar pickups very badly, so a flat-screen computer display is strongly recommended. *Logic* doesn't have a specific de-humming plug-in, but by using two EQs in series you can set up a whole series of very narrow notch filters starting at the fundamental frequency of your mains supply. In the US this is 60Hz, but in Europe we like to do things a little more slowly, so we have a 50Hz supply. In Europe, the filters can be set for 50Hz, 100Hz, 150Hz, 200Hz, 250Hz, and so on. If the hum doesn't have too many buzzy harmonics, four or five filters should be enough. Because of the precision available with digital filtering, even quite severe buzzes can be removed with little subjective effect on the wanted part of the guitar sound. However, don't make the filter notches any deeper than you need to, as you may then hear too much tonal difference.

Broad-band noise can be removed with de-noising software, but sadly *Logic*'s *Denoiser* plug-in is next to useless (aside from emulating live news reports from war zones!), which is odd when you think how good the other plug-ins are. If you have really serious noise problems, then the TC Powercore or Waves sound-restoration plug-ins will do a good job on steady background hiss, as will Bias *Sound Soap* and *Sound Soap Pro*.

Logic's *Noise Gate* or *Expander* may be used to clean up the pauses between notes or phrases, but as electric guitars can sustain for a long time, there may be few periods of true silence where the gate can be effective. In any event, the gate release time needs to be set long enough to allow the guitar to decay naturally without being cut short. Any such noise-removal processing should be applied before any delay or reverb effects are added. This way the reverb or delay will still decay naturally and help cover up any audible artefacts caused by the gate or filter action. Where you're using a lot of overdrive, cleaning up the sound with a gate or expander is a good idea.



Here you can see how to set up a chained pair of Channel EQ plug-ins to reduce levels of hum and noise in a guitar recording.

Logic's *High Cut EQ* and *Low Cut EQ* can also be effective in cleaning up the sound of the electric guitar, because, as a rule, the guitar sound has quite a limited frequency response, while the noise may continue right to the extremes of the audio spectrum. By setting the upper cutoff frequency to between 2.5kHz and 4kHz, it is often possible to significantly improve the signal-to-noise ratio of a typical electric guitar sound without dulling it excessively — use a filter slope of between 12dB/octave and 24dB/octave. Rolling off all frequencies below 80Hz or so may also help clean up the low end.

Finally, some of the best clean electric guitar sounds are obtained simply by connecting the guitar to the computer's audio interface via a DI box — just add a little compression, gentle EQ, and perhaps a hint of reverb. This style of recording can suit funk music, though rumour has it that the guitar on Pink Floyd's 'Another Brick In The Wall' was also DI'd clean and then compressed.

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Processing Options In *Logic*

Recording engineers tend to like leaving as much processing as possible to the mixing stage so as to keep their options open for as long as possible, whereas guitar players like to hear something approaching the final sound as they record — what they hear affects the way they play. For that reason, stomp-box effects are often recorded rather than added later, though effects such as delay or reverb might be safest left until mixdown. If the player needs to hear these to feel comfortable, they can be mocked up in *Logic* for monitoring using any of the available reverb or delay plug-ins. Ultimately, the performance is what really counts, so whatever keeps the player most happy tends to be best in the long run!

Some engineers have been known to re-amp a guitar track by feeding it out through a guitar amplifier, which is then miked up and re-recorded onto a spare track. This can sound very effective, but can also be a bit tedious to set up. However, if you have the facilities to do it, then give it a try. It's also a great way to take the nasty edge off a guitar track that has been recorded with some inadvertent clipping!

Rotor Cabinet, designed for use with *Logic*'s internal tone-wheel organ instrument, is great for giving a little modulation to guitar sounds.

A lazy equivalent is to take a basic guitar track and then further process it through a modelling preamp plug-in such as *Logic*'s *Guitar Amp Pro*, and I've done this very successfully with both guitar and bass-guitar parts. If only the tonality needs changing, you can pick a clean, benign amp model (or even bypass the amp model altogether) and then try out different speaker types, whereas if you want more overdrive you can add that by picking an amp model designed to produce overdriven tones. The electric guitar sound is not, and never was, natural, so there are no real rules as to how it should sound.

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EQ: Suggested Settings

You can also use *Logic*'s *Channel EQ* to shape the sound. Some of the frequency ranges described below may be of help:

- Cut applied at between 120Hz and 280Hz can help reduce lower mid-range muddiness or boxiness. Boost in the same range can fatten a thin sound, often in combination with some upper mid-range cut.

- Cabinet thump can be accentuated by boosting at around 75-90Hz, though how successful this is depends on what is present in the original signal. If all else fails, you can use a little of *Logic*'s *SubBass* plug-in to bring in those deeper frequencies that may be missing from your recording. Just don't overdo it!

- Bite or definition can be added to the sound anywhere between 2kHz and 4kHz. There's little point in boosting higher than this, as guitar speakers tend to roll off sharply above 4kHz, so there may not be anything left there to boost other than hiss.

- A fizzy high end can be tamed by using a high-cut filter with a steep roll-off, adjusting the frequency as low as possible such that the essential bite of the sound isn't adversely affected. This will attenuate all frequencies above the filter cutoff point and produce a more focused sound as a result.

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Creative Processing Ideas

Aside from optimising the EQ, what else can you do to a guitar sound? The answer depends on whether you want to enhance the sound in some way or change it into something radically different. *Logic* offers tools for both as well as ones that fall somewhere between these two extremes. The obvious effects to explore for traditional guitar sounds are reverb, delay, chorus, phasing, flanging, vibrato, and so on, but I find simple compression very useful, on both clean and distorted electric guitar. You'll often find that using as much distortion on a recording as you do live results in a very messy sound that spreads right across the frequency spectrum, so a useful trick is to use less overdrive on the guitar and then add compression to get the sustain back.

A compressor setting such as the one shown allows you to back off your guitar overdrive without sacrificing too much in the way of sustain.

Using a faster release time in combination with a high degree of compression can cause audible level pumping, but this may be used creatively to enhance the sense of power and loudness. Medium to high compression ratios (between 4:1 and 10:1) work well for this, with the threshold set to give between 8dB and 15dB of gain reduction on the loudest peaks. In *Logic*'s *Compressor*, try the Peak and RMS side-chain settings and see which one suits the sound best. Of course you have to bear in mind that every decibel of compression you add reduces your signal-to-noise ratio by a decibel, so it is advisable to clean up the signal first using a gate or expander.

If you're after a conventional rock guitar sound, then the basic overdrive sound often only requires a little EQ and reverb to make it sound right. In this respect, a short reverb with a fairly bright character is ideal for rhythmic parts or staccato playing, as it adds a sense of space and impact without making the mix sound cluttered or messy. Longer reverbs, or combinations of delay and reverb, can be used for more languid guitar parts. Further movement can be added by feeding the effects send through a chorus or flange unit before it gets to the reverb unit. This is easy to do using *Logic* by inserting the modulation effect directly before the reverb plug-in in the Buss Audio object being used as a return.

Slightly less common treatments include feeding the guitar through the rotary-speaker simulation from the *Logic* tone-wheel organ plug-in or (one of my favourites) feeding it through the stereo vibrato plug-in adjusted to give square wave chopping at eight or 16 pulses to the bar, sync'ed to the Song tempo. I've used this trick to turn simple chordal guitar parts into integral parts of the rhythm track in all styles of music, from dance to mainstream pop. If you also filter out some low end, what you end up with is almost like a melodic hi-hat part, especially if the source is a bright, clean guitar sound.

Another trick I've experimented with is playing solo guitar lines that include string bending through *Logic*'s *Pitch Correction* plug-in. Set normally, this ensures all notes are bent to the nearest correct semitone, but you can also speed up the correction rate to produce a mild 'yodelling' effect that gives the sound a slightly Eastern flavour. This may be even more effective if you set up your own custom scale based on an Arabic note sequence. SOS

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SOUND ON SOUND
The World's Best Music Recording Magazine

In this article:

Recording Piano & Harpsichord

Gordon Giltrap

Published in SOS August 2005

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Technique : Recording/Mixing

Recording acoustic keyboard instruments in a home environment can be pretty demanding, but with a little care you can still get professional results.

Paul White

Following our Studio SOS visit to Gordon Giltrap back in *SOS* October 2003, Gordon asked me if I could help with a project he was working on. His plan was to record a piano album based on interpretations of his own guitar compositions, where his friend Richard Leigh Harris would play the piano, while Gordon acted as musical director and producer. The album would also include some tracks on harpsichord. As Richard has a rather nice Steinway piano and a double-manual harpsichord in the front room of his Oxford home, Gordon decided to record the album there. I suggested that we take a purist approach by recording the instruments using a single Soundfield microphone set up to behave like a pair of coincident cardioid microphones. We could use my Alesis ADAT HD24 as a two-track recorder at 44.1kHz, 24-bit resolution and then transfer the finished files to Apple *Logic* for editing and mastering.

Paul White plays Gordon Giltrap some initial test recordings to gauge the success of the mic positions.

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Setting Up For Harpsichord Recording

Richard's home has one long room on the ground floor extending from a bay window at the front to French windows overlooking his garden at the rear. The piano is located near the bay window, while the harpsichord is positioned against a side wall at the other end of the room, close to the French windows. The only practical place to set up my recording equipment was on a table opposite the harpsichord, and to facilitate monitoring I connected up my Alesis 3204 mixer, which was used as little more than a glorified headphone amplifier. The Soundfield mic's decoder box plugged directly into channels one and two of the HD24 and I also set up my pair of Mackie HR624s, driven from the mixer, so that we could all listen to the playbacks.

The harpsichord recordings were tackled first, and a well-balanced sound was achieved by positioning the mic about 18 inches from the edge of the instrument about half way up the open lid. By aiming the mic roughly at the hinge at the back of the lid and setting it up around a third of the way back along the instrument (from the keyboard end), the high and low strings were captured with a good natural balance and the playback tonality sounded essentially the same as we were hearing in the room. The only technical problem was that, due to the close proximity of the recording setup to the harpsichord, I was concerned that some of the drive noise from the HD24 would end up on the recording. Our simple solution to this was to build a tent of towels over the machine in such a way as not to block off all the airflow. This didn't silence the machine completely, but the noise levels were now negligible. In fact our main noise worry turned out to be the sound from outside, as Richard lives on a side road that carries some traffic, and his windows are not double-glazed.

Richard had engaged the services of a piano tuner to get the piano and harpsichord ready for our arrival, but once we started recording, we noticed that one of the lowest harpsichord notes had gone slightly flat, so Richard had to find the wrench and retune it. This meant re-recording a couple of pieces. Richard was working from a book of guitar transcriptions of Gordon's music, and some on-the-fly re-arrangements had to be made to get the material to work for harpsichord. We ended up typically recording three versions of each piece, with some additional introductions and short bits that we could use to patch up any sections where we thought there might be problems. This part of the session went without any further hitches, so when we broke for lunch, I backed up the recording drive on the HD24 to its second drive — just in case!

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Miking The Piano

When it came to recording the piano, the traditional mic position didn't work, as the room was only two or three feet wider than the piano on each side. Ideally I'd have liked to set up the mic around five feet from the side of the open lid with no obstruction behind, but this clearly wasn't going to be possible — at least without a wrecking ball, a couple of RSJs, and a serious carpet-cleaning bill! I tried getting the mic as far away as the wall would permit, but the proximity of the wall to the microphone really coloured the sound, so I decided to try miking from the tail end of the piano instead. For this, the lid was fully open and the mic around 18 inches from the end of the piano, about 12 inches above the piano body. This produced a well-balanced and tonally neutral sound, but it was obvious that some 'concert hall acoustics' would have to be added in post-production.

The final position of the soundfield mic for the harpsichord recordings.

Almost as soon as we started recording, we discovered that one of the piano pedals was squeaking quite loudly, and no amount of soap or candle grease would stop it. The problem was clearly inside the pedal housing where we couldn't get to it. After a little experimentation, we found that by pushing the right pedal to the left while operating it, we could reduce the squeak, so after several futile experiments with elastic bands, we resorted to the simple solution of binding both pedals tightly together with Gaffer tape. This still allowed sufficient vertical motion for the pedals to operate independently, but exerted enough sideways pressure to cure the squeak. We had to start one tune again after an Argos home-delivery truck rumbled past just outside the window, and we also had to take a couple of enforced breaks when a car alarm went off down the road, but on the whole the level of noise intrusion was surprisingly low.

As with the harpsichord pieces, we recorded around three takes of each piano piece plus as many variations and patch-up pieces as we thought we might need. Richard also tried alternative interpretations of some of the songs, so we ended up with quite a lot of material to choose from.

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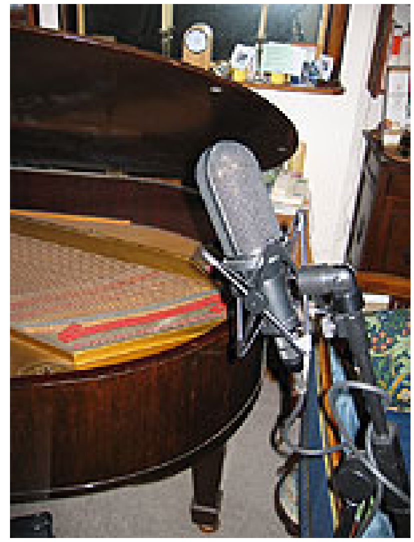
Editing & Compilation

All the recorded material was transferred to my Mac using the excellent Alesis Fireport, so I ended up with one long stereo track for each of the four half-day recording sessions. Gordon listened to all the material and made copious notes before we started editing. Most of the songs needed at least two or three edits to compile a best version or to avoid noisy pedals on a quiet intro, but some of the more challenging pieces needed significantly more. We even dropped in a single note at one point!

However, the procedure was the same in each case, using *Logic*'s Scissors tool to separate the clips we needed to insert, then cutting holes in the main track to drop them in. As ever, some fine-tuning of the edit points was necessary, and



Because of the dimensions of the room, no conventional mic position could be found which offered an uncoloured sound. Therefore, after some experimentation, the final mic position ended up being at the foot of the piano, as shown from two angles here.

the zoomed-in waveform view helped line up note transients. Occasionally even our best shot at an edit didn't work out, so we had to move the edit point a note or two back or forward to find the most seamless transition. During this process, I made extensive use of *Logic*'s Object Colours facility to identify the various clips and takes, and once a track was assembled, I used a very short crossfade over most edits to avoid the possibility of an audible glitch. Usually a fade of as little as 50ms will achieve this.

As each track was completed, it was bounced down to a single stereo file ready for adding effects and compiling. Because they were so many edits, and because for some songs completely new arrangements needed to be created from the material we had, the editing took around a day and a half to complete satisfactorily. One tune, 'Giltrap's Bar', was created out of numerous snippets taken from our various takes, with the best phrases occasionally repeated. Gordon was initially concerned that this particular tune wouldn't translate well to piano, but once we'd compiled all the best bits, we both felt we'd arrived at something exceptional and yet still natural sounding.

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Finishing Touches

Gordon was also a little worried that the sound of the piano damper action was too pronounced, which is often a problem when pianos are recorded in a small space where the sound doesn't have a chance to breath. However, when auditioning the recordings with a suitable concert-hall reverb applied courtesy of *Logic*'s *Space Designer* convolution reverb, the overall sound gelled really well and sounded much more natural. The straightforward concert-hall reverb was used for the more conventional piano pieces and to a lesser degree on the harpsichord pieces. A couple of the more ambient tracks were treated with a very low-level repeat delay (*Logic*'s *Tape Echo*) combined with a more generous reverb to give them a floating, shimmery quality. For these tunes we chose a 3.3-second cathedral reverb coupled with a delay (with feedback) of around half a second.

Gordon also felt that the overall piano sound should be a little more sparkly to give the album a contemporary feel, and it turned out that a very subtle application of *Logic*'s *Exciter* plug-in was the best way to achieve this. Its filter was set to around 3kHz and then the Harmonics control was turned up to 14 percent. The final plug-in processor was the Waves *L1* limiter, just to ensure that no peaks escaped our notice. *L1* also allows the gain to be adjusted relative to the limiter threshold, so it's a convenient way of getting the track levels maximised.

Stopping the pedals squeaking involved gaffer-taping the two pedals together.

Each track was set up with the required amount of reverb and then bounced to a new file before being 'topped and tailed', then assembled as a playlist in Roxio's *Jam* for burning the final master CD. I habitually use Bias *Peak* for the final trimming of tracks, and as it gives the files a new *Peak* icon, it provides an easy visual indication of which tracks are finished. *Jam* includes the means to turn levels up or down from within the playlist so that fine level adjustments can be made between tracks. On this project, the main challenge was getting a nice balance between the harpsichord and piano tracks, as the strident tone of the harpsichord makes it seem relatively louder. The track gap time can also be fine-tuned by ear within *Jam*, as the best value subjectively depends on how long the decay of the previous track ending is and also on the pacing of the material. This doesn't compromise the resolution of the audio, as the final dithering from 24 bits to 16 bits is the last thing *Jam* applies before burning the disc.

I was prepared to return to this project after Gordon had taken the original mixes home to listen to, but a few days later he said it was fine exactly as it was, so our first test CD burn ended up also being our final master. SOS

The finished CD, Gordon Giltrap's *Another Time, Another Place*, is to be released on Voiceprint records.

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In this article:

Replacing Drums In Cubase SX

Workshop

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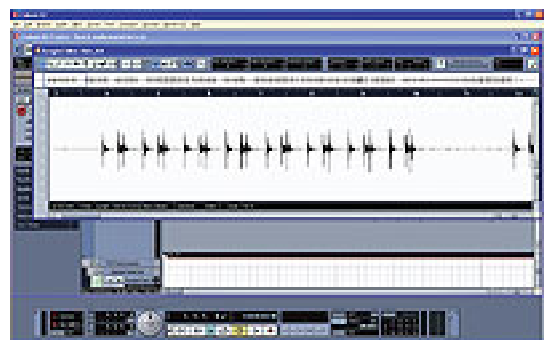
Technique : Cubase Notes

You've recorded a real drummer, but the drum sound just doesn't cut it. Is there anything you can do to rescue things?

Sam Inglis



A typical kick drum part in *Cubase SX's* Sample Editor.

Since the use of sequencers, click tracks and samplers became widespread in the '80s, rock and pop drums have changed out of all recognition. It's commonplace for rock bands to use a blend of live and sampled drums, or to augment real percussion with loops and triggered sound effects. Sometimes these techniques are used to create an obviously processed or electronic drum sound, or to construct patterns that no human drummer could ever play. In these cases, it's usually the sequencer that drives the song forward, with the flesh-and-blood percussionists desperately trying to keep up.Even where the intention is to present a natural-sounding band performance, however, samples have become part of the pop producer's armoury. Provided the drum kit is individually miked, it's possible to take a real drummer's performance and replace or augment every hit with a sample of your choice, even where that performance is not played to a click. There are numerous reasons why you might want to do this. Inexperienced drummers often struggle to tune drums properly, or bash the hi-hats and cymbals too hard, making it difficult to boost the kick and snare in the mix without suffering unwanted hi-hat spill. Sometimes a kick or snare that sounded great at the tracking stage just doesn't cut through in the mix, and requires more thickening or brightening than can be achieved with EQ or compression; layering it with a sympathetically chosen sample can do wonders.

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Replacing *Sound Replacer*

The industry-standard tool for replacing drum hits is Digidesign's *Sound Replacer*, but this is an off-line Audiosuite plug-in that only works in *Pro Tools*. So what sort of options exist for those of us running *Cubase SX*?

The Hitpoints Detection dialogue offers various presets plus an Advanced mode that allows you to set Sensitivity and Threshold values yourself.

Well, if this is the sort of thing you need to do often, you should definitely investigate Wavemachine Labs' *Drumagog*. This is a multi-format, cross-platform plug-in that can be inserted on a kick, snare or other drum track. When it detects a beat on that track, it triggers a drum sample of your choice. The triggering can be done on a velocity-sensitive basis, allowing you to use multisamples to replicate loud and quiet hits. *Drumagog* can load samples in most major formats, and will also output MIDI, allowing you to use it to trigger another soft sampler or synth of your choice. However, at the time of writing, the only version of *Drumagog* available was the full *Drumagog Pro*, which costs $269 — not much cheaper than *Sound Replacer*, and more than many people will be able to justify for the sake of beefing up the odd drum track. As a paid-up cheapskate, I decided to see whether it was possible to achieve the same results using *Cubase* itself.

I first got interested in the idea last year, when I was given some distinctly iffy drum tracks as part of a mixing project. At the time, I was still using *Cubase VST 5* on an old G3 Mac — and perversely enough, this particular job turns out to be easier in *VST 5* (as described in the box) than in the new and shiny *Cubase SX*. Nevertheless, it is possible.

The method I eventually came up with uses *SX's* Hitpoint functions. These offer a number of powerful features for matching the timing of existing audio or MIDI parts, dividing rhythmic audio into its constituent hits, and creating tempo maps or Groove Templates. Although they are very good at extracting information about the timing of hits within a drum part, however, they don't offer an obvious way of creating another part from scratch which incorporates the same timing information, making a slightly convoluted workaround necessary. The procedure I'm going to describe involves a slight abuse of *SX's* Markers feature, so if you want to try it on drum tracks in a Project that already incorporates Markers, it would be best to create a new Project without any Markers, and import or copy the drum tracks to that.

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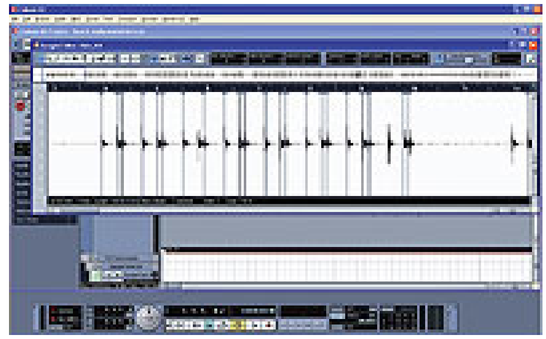
Hitting On Point

The first thing you need to do is to force *SX* to create Hitpoints in the right places. Double-clicking on the drum part you want to replace or augment will open it in the Sample Editor. Now go to the Audio menu and choose Hitpoints / Calculate Hitpoints. Here, you can set Sense and Threshold parameters which will aid *SX* in detecting all and only the drum hits you want. A number of presets are available, but I find it's invariably necessary to use the manual Advanced setting and try various combinations of Sense and Threshold in order to pick up, for example, all the genuine snare hits on a snare track without any of the hi-hat spill. Usually, a Sense somewhere above 80 and a Threshold somewhere below 10 seem to give the best results. You'll notice that there's a separate Hitpoint Sensitivity slider at the top right of the Sample Editor itself. This appears to control the same thing as the Sense parameter in the Hitpoints Detection window, and because you can adjust it later, it's best to leave yourself some room to do so — if you need a very high Sensitivity, ramp up the Sense parameter at this point, rather than the slider.



My first attempt has generated Hitpoints for most of the kick-drum hits, but not all, so I need to adjust the Hitpoint Sensitivity slider (top right). There are also a couple of false doubles which will need to be dealt with, either by disabling those Hitpoints or later MIDI editing.

Both the Hitpoints and the drum hits themselves should be clearly visible in the Sample Editor, as long as your horizontal zoom setting is appropriate, and if you're lucky, the right choice of Sense and Threshold parameters will generate Hitpoints on every drum hit, without any spurious ones. Snare tracks seem to be luckier than kick drums, in this respect!

However, it may well be that even the optimum Sense and Threshold settings either miss out one or two drum hits, or create a few spurious Hitpoints. In this case, the obvious thing to do is to manually add and delete Hitpoints. Sadly, though, this is one of those areas where *Cubase SX* doesn't let you do the obvious thing. Even though it was possible to select, cut, copy and paste Hitpoints in the old *Cubase VST*, *SX* appears to offer no way of achieving this. Instead, you have no option but to fine-tune the automatic detection algorithm and hope that it can catch all the drum hits on its own. The easiest way to do this is to increase the Hitpoint Sensitivity slider until all the necessary Hitpoints appear, then deal with any unwanted ones by selecting the Hitpoint Edit tool from the Sample Editor's toolbar and Alt (Windows) or Option (Mac) clicking on the triangular handles of any spurious Hitpoints to disable them. (Since we're going to be creating a MIDI drum track that matches these Hitpoints, you could also choose to leave any weeding out until later and do it in one of the MIDI editors.)

The Create Markers from Hitpoints function does exactly what you'd expect. My Marker track (above) now contains a Marker for every drum hit in my kick-drum Part (below).

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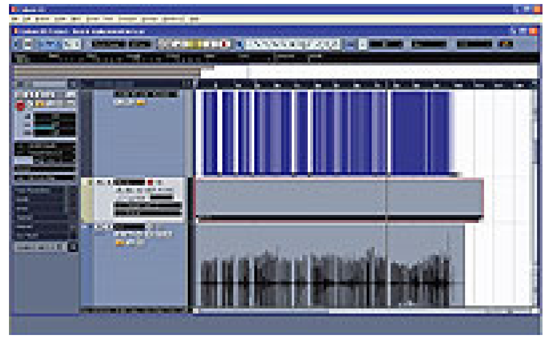
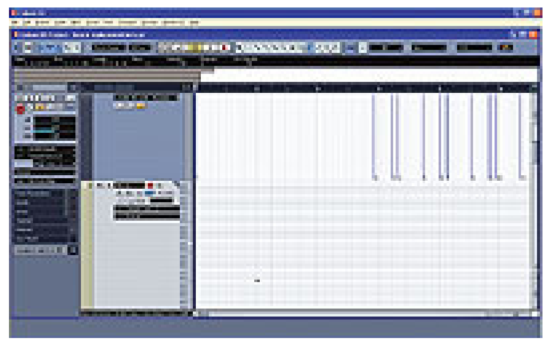
Turning Hitpoints Into Hits

By this stage, you should have created a Hitpoint to match each of the drum hits you want to replace in your snare or kick part. The problem now is how to use these Hitpoints to create the necessary replacement sounds. The most natural and flexible approach is to use them to generate MIDI notes, which can be routed to a hardware or software sampler of your choice. You could even use *SX's* own *LM7* drum machine, although your drum recordings would have to be pretty bad before this would represent an improvement.



The next step is to create an empty MIDI Part extending the full length of the kick-drum Part.

Unfortunately, although there are lots of clever things you can do with Hitpoints to turn them into tempo tracks or Groove Maps, there appears to be no way to generate MIDI note data directly from Hitpoints. What *SX* does offer, though, is a command to Create Markers from Hitpoints, again located in Audio / Hitpoints. Selecting this will create a Marker track and insert a Marker that corresponds with every Hitpoint (as well as one at the very start of the Project).



Selecting the MIDI Part for In-place editing will allow me to check that my MIDI notes line up with the Markers in the Marker Track.

Again, there's no command to instantly convert Markers to MIDI notes *en masse*, but with a little help from the Macro engine in *SX3*, it is possible to automate the procedure in a relatively painless way. Close the Sample Editor and create a MIDI track in your Project. Route the output of this track to the sampler you want your drum part to trigger, and create a MIDI Part that extends the full length of your song (ie. one that starts at zero and is at least as long as the drum part you're replacing). If you're using *SX3* rather than one of the 'lite' versions, the easiest way to do the next bit is probably to open this MIDI Part for In-place editing, either by clicking the button on that track that looks a bit like an office in-tray, or hitting Ctrl + Shift + I (Apple + Shift + I on the Mac). With the MIDI track adjacent to the audio track containing your original drum part, In-place editing will allow you to confirm visually that the MIDI drum notes you're creating are lining up correctly. In-place editing isn't included in other versions of the program, so *SL* users will have to open the Part in the Key Editor instead, but it should still work.

At this point, you need to decide which MIDI note to use to trigger your sample. Select the pencil tool and draw in a single note into the MIDI part. Now switch back to the pointer tool, select the note you've just drawn and cut it to the clipboard by hitting Ctrl + X (or Apple + X on a Mac). Click in the ruler at the top of the Project Window to place the Project cursor somewhere to the left of the first drum hit you want to replace. If you now hit Shift + N, you'll see that the cursor jumps forward to the first Marker, which should be exactly in the same place as your first drum hit. Hit Ctrl + V (Apple + V on a Mac) and your MIDI note will be pasted here. Hit Shift + N and you'll move to the next drum hit; Ctrl + V will paste another MIDI note here, and so on.

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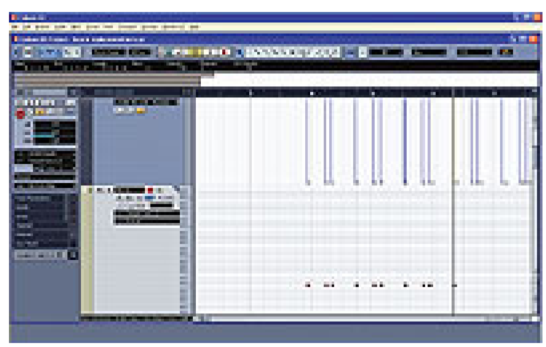
Macro Magic

Now, you could use these shortcuts to step manually through every Marker, and thus every drum hit, but it gets pretty tedious when you have several hundred drum hits to replace. The answer, if you're using *SX/SL 3*, is to use the Macro function built into the Key Commands dialogue. If you've not created a Macro before, the process is explained pretty well in the *Cubase* documentation, so I won't give a blow-by-blow account. What I did was to create a basic Macro consisting of a single Paste command (which is found in the Edit group) and a single Locate to Next Marker command (which lives in the Transport group). I then gave this the name 'Paste note at markers' and created another Macro called 'Paste note 10 times' which, in turn, consisted of 10 instances of this first Macro. Finally, I assigned an unused key combination to the 'Paste note 10 times' Macro.

Here, I have created two Macros. The first Locates to the next Marker and Pastes whatever is on the clipboard. The second simply repeats the first 10 times. I've given it the keystroke Ctrl + Alt + P.

Back in the MIDI editor, hitting your chosen key combination will enact this new Macro, allowing you to step through the Markers 10 at a time, creating a matching MIDI note for each one. Obviously you could create a Macro that pasted 20 or even 50 times if you wanted to, but even a 10-step version allows you to replicate an entire drum part in a matter of seconds. With the output of your MIDI track routed to the sampler of your choice, going back to the start of the song and pressing Play in *Cubase* should mean that you hear every drum beat in your original part precisely doubled with whatever sample you've chosen.

If you want to replace more than one drum track in a Project, be aware that deleting the Marker Track you've created doesn't actually delete the Markers on it. In order to get rid of them so that you can repeat the procedure for a different drum part, you'll need to visit the Markers window, either by mousing to the Project menu or hitting Ctrl + M (Apple + M on the Mac). You can't do a Ctrl + A here, but if you click on the first Marker, then scroll all the way to the bottom and Shift-click on the last, you can select all of them before clicking on Remove.



Having cut a MIDI note to the clipboard and placed the Song Position Pointer to the left of the first drum beat, I hit Ctrl + Alt + P — and now I have MIDI notes to match my first 10 drum beats. Repeating the command will do the next 10, and so on.

In most circumstances, it's useful to be able to create a MIDI part to match your original drum recording, as I've described, because there's lots you can do to edit this; you can always bounce or freeze the results as an audio track for further processing later. However, if you simply want to replace every drum hit in an audio part with a single, unvarying sample, it might be quicker to bypass the MIDI stage altogether. Make sure that the drum sample you want to insert is hard trimmed at the left, and copy it to the *Cubase* clipboard. If you now create a new audio track and insert the Project cursor somewhere before the first Marker, the same Macro should allow you to paste the sample as an audio Part at every Marker point.

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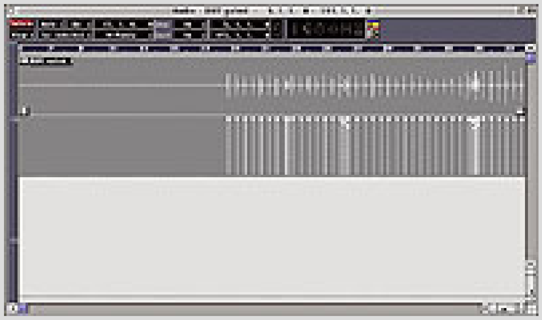
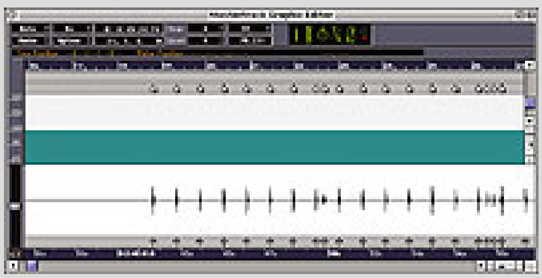
A Brush With History

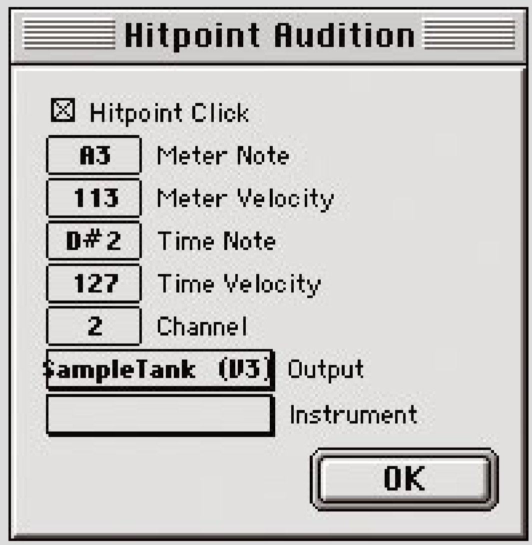


My drum part in the Audio Editor, with M-Points displayed below the waveform.

The Get Match Points dialogue is *Cubase VST's* equivalent of the Hitpoints Detection window in *SX*, and offers similar parameters.



In the Graphical Mastertrack Editor, my M-Points have been turned into Hitpoints.



The Hitpoint Audition dialogue allows me to trigger a MIDI note with every Hitpoint. Note that Time and Meter Hitpoints are set to trigger different MIDI notes — because I have one of each for every drum hit, I don't want them both triggering the same note, and in this case, I've deliberately chosen a note for the Time Hitpoints that won't trigger a sample at all.

As I mentioned in the main text, I was first confronted with the need to create replacement drum tracks when I was using *Cubase VST*. Although the older program is officially obsolete, I know there are plenty of people out there still using it, so for their sake, I thought it might be worth describing how it works there — especially as it's actually rather quicker and easier!

First, double-click on your chosen audio Part in the Arrange page to open it in the Audio Editor. Each lane in this editor window shows a waveform view in the top half, while the bottom half is used to display Dynamic Events: volume (the default) or pan. However, this part of the screen can also be used to display M-Points, which are the markers that *VST* generates when it analyses an audio file for rhythmic information. Select M-Points from the pop-up menu just to the left of the label that says 'Quant', and the bottom half of your lane should appearempty. If you now select Get M-Points from the 'Do' menu, you'll be presented with a dialogue that allows you to fine-tune *VST's* automatic rhythm detection algorithm. With the correct settings of Sensitivity, Attack and Max. Number of Events per Sec, clicking Process should generate an M-Point for every drum hit in your Part. As with *SX*, you'll probably find that you need to try several times with different settings before you get it right, but *VST* has the advantage that you can edit the M-Points manually using the arrow tool to move them and the eraser to delete them. In general, I find that *VST's* transient detection is actually better than *SX's*, but it's also more sensitive, and it can be difficult to stop it picking up hi-hat spill as well as genuine snare hits. If this seems unavoidable, I usually bounce the snare part through a gate to get rid of spill.

> Once you're happy that you've got an M-Point for every drum hit, return to the Do menu and choose Match Audio and Tempo. If you now close the Audio Part Editor and open the Graphical Mastertrack Editor, you'll see that your M-Points have been turned into Meter and Time Hitpoints, visible in the ruler above and below the Tempo Track. Now for the clever bit. Unlike *SX*, *Cubase VST* has the option to make Hitpoints audible by having them output a MIDI note to the device of your choice. From the Graphical Mastertrack Editor's Do menu, choose Edit Hitpoint Note. This will allow you to specify separate notes that will sound whenever a Meter or Time Hitpoint is encountered, and choose a MIDI output. Since we have both Time and Meter Hitpoints, it's a good idea to set the velocity or pitch of one of them to a value that won't generate any output, or else you'll get your drum hit doubled. Set the other to the note that's assigned to your chosen drum sound, set the MIDI output tothe hardware or software synth or sampler of your choice, and press Play. If all has gone according to plan, you should hear the sampler in question playing along with your original audio part. The new part can then be bounced or recorded to a new audio part.

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Variations

One limitation of the method I've described above is that the MIDI drum part it produces contains no variation in velocity, because it's created by pasting the same note over and over again. In practice, this is not always a problem: where a kick or snare sound needs this kind of treatment, it's often because the drummer wasn't consistent enough in the first place, so augmenting it with a uniform sample can help to even things out. If you do want to add some dynamic variation, there are a number of options. The first is simply to draw in the necessary velocity information, either in the Velocity view of the In-place Editor (if you're using *SX*) or by adding an automation lane to the MIDI track and using it to vary the level of the Velocity Shift Track FX. Again, the advantage of doing this kind of editing in the main Project window is that you can view both the MIDI and audio tracks at the same time, which makes it easy to draw in velocity variations that mirror the levels of the drum hits in your original audio part.

However, it is also possible to refine the method I've described in such a way as to automatically generate a MIDI part containing notes of two, three or more different MIDI velocities that correspond to the forcefulness of the original drummer's playing. The idea is to use the same procedure, but in multiple passes. When you first create your Hitpoints in the Sample Editor, bring down the Hitpoint Sensitivity slider until only the very loudest drum hits are 'caught'. Create the corresponding Markers and follow the steps described above, making sure that the MIDI note you paste has a high velocity — for instance, the maximum 127. Once you've created MIDI notes to match all of these Hitpoints, reopen your audio part in the Sample Editor and raise the Hitpoint Sensitivity a little, so that further Hitpoints are created for the 'medium' drum hits. Choose Create Markers from Hitpoints again and go back through your MIDI Part, this time pasting in MIDI notes at a rather lower velocity, say 100. You can repeat this procedure with higher and higher Hitpoint Sensitivity settings and lower MIDI note velocities until all your drum hits have been mirrored.

If you do a total of, say, three passes, you will now have a MIDI Part containing one MIDI note for every quiet drum hit, two for every medium-strength one, and three for the loudest hits. To get rid of the duplicates, select this Part and choose MIDI / Functions / Delete Doubles. Where two or more MIDI notes are stacked on top of one another, this seems to delete the newer ones, leaving you with a MIDI part containing loud notes where there are loud hits in the original drum part, and so on.

The technique outlined here has helped me to rescue mix projects that would otherwise have been unlistenable, but let me end on one note of caution. For some reason, whilst many musicians are perfectly happy for an engineer to EQ and compress their output to death, they're not so enthusiastic when it comes to being replaced by samplers — even if, as in this case, you're keeping every aspect of their performance intact bar the actual sound of the kick or snare. Tread carefully! SOS

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SOUND ON SOUND
The World's Best Music Recording Magazine

Sonar Notes

News & Updates

Published in SOS August 2005

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Technique : Sonar Notes

News this month includes more about 64-bit computing and details of a Mackie control-surface update for *Sonar* users — and we offer handy tips on taming mute tools and MIDI notes.

Craig Anderton

The recent *Sonar X64* 64-bit jam session at Seattle's Experience Music Project.

If you haven't seen the Experience Music Project (EMP) in Seattle, USA, it's basically an homage to music from Microsoft's Paul Allen — as well as being the most amazing musical museum I've ever had the pleasure of visiting. So it was a natural venue for a recent *Sonar X64* jam session, hosted by the 2005 WinHEC (Windows Hardware Engineering Conference). Along with Intel employees and other conference attendees, jazz organist Tony Monaco used the just-released *Project 5* as his host, relying primarily on the *Dimension* sampler for grand piano, clavinet, rhodes, organ and analogue lead patches (see photo below). What's more, the jam session was recorded live using the *Sonar x64 Technology Preview* on a 955X Express chipset-based PC with 8GB of RAM, running a Pentium Processor Extreme Edition (a dual-core processor with 3.2GHz core speed in each core, supporting Hyperthreading, for four threads in total). Looks like all this stuff really does work!

And while we're talking 64-bit, Cakewalk have added lots of 64-bit computer-related information to the Cakewalk DAW Labs PC Resource Guide at www.cakewalk.com/PCResource/default.asp. The info is helpful if you're considering buying or building a computer for audio, but almost essential if you're entering the brave new world of cutting-edge 64-bit computing.

Topics covered on the site include 64-bit operating systems in general, recommendations on compatible workstations, suitable audio/MIDI interfaces and drivers, and an FAQ that answers fundamental questions. And don't forget that you can download the *Sonar X64 Technology Preview* for free at www.cakewalk.com/x64.

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Where Are My Notes?

You drag a group of MIDI notes, and are baffled when you see them erase existing notes in the area to which you're dragging. Or you drag a bunch of notes and they push existing notes out of the way. What's going on?

Cakewalk's DAW Labs Guide: reliable info about computers and digital audio.

No prizes for answering 'annoying software bug'. The correct answer is 'Drag and Drop Options' (these affect audio and MIDI clips as well as MIDI data in the piano roll view). To call them up, right-click in the Clips pane and select 'Drag and Drop Options'. The section called 'What to do with existing material' has three options: Blend Old and New, Replace Old with New, and Slide over Old to Make Room. In most cases you'll want to select Blend Old and New, so that nothing that already exists is moved or erased. If you select Replace Old with New, anything 'underneath' the clip will be removed; if Delete Whole Measures is ticked and a moved clip covers only a few beats of a measure, any other data in that same measure will be deleted as well. If you slide one clip over another and Slide Over Old to Make Room is ticked, the new clip will split the existing data at the insertion point, and existing data to the right of the split will be moved further right to make room.

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In Brief

- Mackie Control: If you're using Mackie's Control Universal (MCU) control surface, here's some good news: An updated control surface plug-in for *Sonar* now accommodates the optional XT and C4 controllers, as well as the main MCU. For more information, visit www.cakewalk.com/Support/kb/ kb2005263.asp .

- MP3 Encoding: The MP3 encoder in *Sonar 4* is a trial version, which means that it works for a while then plays hard to get. You can purchase a Cakewalk MP3 activator for unlimited use, but what if you've already bought it for a different Cakewalk product? The good news is that you don't have to buy it again. Just go to www.cakewalk.com/owners/mp3, where you'll find a link to *MP3Upgrader04.exe*. This small (193K) application will work to unlock the *Kinetic*, *Project 5* V2 or *Sonar 4* MP3 encoder.

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The Mute Tool

I've never experienced any problems with the Mute tool, but some people do complain about "unpredictable" behaviour. I think the secret to ensuring that the Mute tool does what you want it to is probably the knowledge that where you drag within a clip — above or below the zero-crossing line — determines what happens. Dragging in the lower part of the Clip causes a mute, while dragging in the upper part unmutes. Of course, when a track is set to its minimum height it can be difficult to see exactly where in the track you're dragging. Just look at the cursor: if it has transformed itself into the international 'no' sign, you're in the mute (lower) part of the clip. If the cursor has a square shape, you're in the unmute (upper) part of the clip.

Note that Cakewalk does allow your aim to be a bit sloppy: once you click in the clip's upper or lower half, you've 'set' the cursor for as long as the mouse button is down. You can now drag anywhere on the clip and the cursor's role won't change.

There's one other Mute-tool subtlety. If you choose Mute Entire Clips from the Mute tool drop-down menu, you can simply click on a clip to mute it and click again to unmute (you don't have to drag through the clip). This is confirmed by a small 'no' symbol in the clip's upper-left corner. SOS

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SOUND ON SOUND
The World's Best Music Recording Magazine

In this article:

Spreading Your Music Across Networked Computers

PC Musician

Published in SOS August 2005

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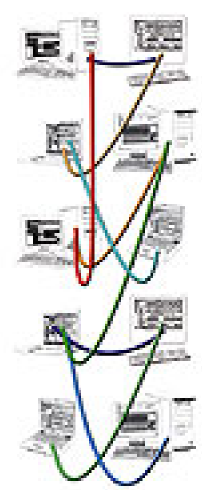
Technique : PC Musician

Networking computers is now more straightforward than it used to be, there's a good choice of connection protocols, and Macs can get in on the act alongside PCs. The benefits for musicians can be considerable, as we discover...

Martin Walker



When your PC begins to protest at the number of plug-ins, soft synths and soft-sampler notes you're attempting to run on it, there are various ways to proceed. Last month I discussed various upgrades that would make a single PC more powerful, but it's sometimes easier to spread the processing load between several computers, so this month I'm going to concentrate on networking. This may also be a cheaper option than upgrading, if you've already got a spare PC that can be pressed into service. After all, many of us have amassed several computers over the years.

Traditionally, home networks have been employed to let multiple family users share the same printer or Internet connection, access shared files, stream media content around the house or play multi-user games. However, the musician can also benefit from their distributed processing capability, as well as being able to back up audio projects to another computer for greater safety, and to share other devices, such as DVD burners, external hard drives and so on.

Even if you've already got a very powerful PC, setting up a network will create a resource that can expand with your requirements: if you need to run more tracks and notes in your scores, add another PC. And if you have a laptop for live work, but a more powerful desktop PC in the studio, networking them will let you combine their processing power during the mixdown phase when you generally need to run more plug-ins.

If, like many musicians, you're worried about having your music PC connected to the Internet, a network can also provide a little more security. One computer can be net-enabled and stuffed to the gills with firewall, anti-virus and anti-spyware utilities, so that you can deal with emails, visit the *SOS* forums and download application updates. Any other computers on the network need only be active when you're actually making music or want to transfer music updates across via the network. For even more security, you could use interchangeable hard-drive caddies on the music PC, so that the one containing your Windows music partition is never plugged in while the Internet machine is on-line and exposed to virus attacks.

Yet another advantage of a network is that you can make it cross-platform, so that, for instance, Mac users could add PC-only software to their sonic arsenal by having a PC on their network. The advantages of networking are clear — but how do you actually send music signals from one PC to another?

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PC As Stand-alone MIDI Synth

The simplest way to musically join two or more computers is via a MIDI cable. By connecting the MIDI interface outputs from the master PC running a MIDI + Audio sequencer to a MIDI interface input on the slave machines, you create a MIDI network and you are effectively turning the additional machines into stand-alone hardware synths. If you already have MIDI and audio interfaces on each machine there may also be no extra hardware to buy or set up (but see 'Audio Routing & Mixing' box for alternative approaches).

If you only want to run a few soft synths on your additional PCs, you may not need a host application for them, either. For example, most products from Native Instruments are supplied in stand-alone versions as well as in plug-in form, as is the AAS range (*Lounge Lizard*, *String Studio* and *Tassman*), along with many other manufacturers' ranges, while 'software studios' such as Arturia's *Storm*, Cakewalk's *Project 5* and Propellerhead's *Reason* all run as stand-alone applications.

There are, however, some situations in which running a sequencer on each machine and synchronising them via MIDI (running one as the Master and the others as Slaves) provides advantages. A MIDI connection would, for instance, allow you to link different sequencer programs together, either all running on PCs or on a mixture of Macs and PCs.

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The Simple Host

However, even if you do need a sequencer for your additional computers, it needn't necessarily be an expensive flagship version. Not only are there plenty of entry-level versions that could be perfectly adequate (see PC Musician April 2005 for more details), but since the 'remote sequencers' only need to be able to host plug-ins and soft synths, and require no recording or editing functions, a much simpler host utility could be pressed into service. There are now plenty to choose from. As they all differ slightly in their facilities but are all comparatively small applications, your best bet is to download a few demo versions to see which one suits you best.

One of the simplest is Xlutop's *Chainer* (www.xlutop.com), which allows you to stream up to 10 chains of VST instruments and plug-ins. The demo version is a tiny 245K download and the full version costs just $60. Dsound's *RT Player Pro* (www.dsound1.com) supports up to 24 VST plug-ins and Instruments and multiple MIDI inputs, and it includes 13 effects, for around 150 Euros, but doesn't have a demo version (although M Audio interface owners can download the simpler OEM *RT Player Express* version for free.

To run plug-ins and soft synths on remote PCs, you may only need a comparatively simple host application, such as Xlutop's *Chainer*, Brainspawn's *Forte* or Toby Bear's *Mini Host*, shown here.

Spin Audio's *Virtual Mixing Console* (www.spinaudio.com) is another host with rather more advanced effects included, allowing up to 16 VSTis to be treated with the integral four-band parametric EQs in each of its mixer channels, as well as with other VST effects. A 4.5MB demo is available and the full version costs just $59. Brainspawn's *Forte* (www.brainspawn.com) costs £75, supports up to 32 VST Instruments and associated effect plug-ins, and was discussed in detail by Marillion's Mark Kelly in our May 2005 *SOS Live* supplement. While it's optimised for stage use, it's ideal as a low-latency host driven from a separate PC, and the demo is only 3.3MB in size.

Steinberg's *V-Stack* (www.steinberg.de) is only available as a 10.4MB download direct from the Steinberg web site but is discussed in more detail later on, since it also offers internal synchronisation with other Steinberg applications. Finally, there are also a few free host applications out there if you look hard enough, such as the 84K *Simple Virtual Host* from FXpansion (www.fxpansion.com/skunk/svh14.zip), and (possibly the most popular of all) the donationware *Minihost* from Toby Bear Productions (www.tobybear.de/p_minihost.html).

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Definitions

Although some *SOS* readers might already be able to wire up a network in their sleep, having had vast experience of them in office life, others find the whole process totally baffling and full of arcane terminology. Here are some definitions that explain the basics.

■ Ethernet: The most widely installed Local Area Network technology, with a transmission rate of 10Mbps (Megabits per second). The newer Fast Ethernet standard increases this rate to 100 Megabits per second, and if the LAN port on your computer is specified as 10/100 Ethernet, it supports both of these standards. Finally, as its name implies, Gigabit Ethernet runs at 1000Mbps. It's the most desirable of the three if you're going to attempt to send audio signals across your network.

■ Intranet: A private network of various computers within an organisation such as a company or educational establishment, which looks to its users like a private version of the Internet.

■ IP (Internet Protocol) Address: A string of four numbers, separated by dots, that uniquely identifies each machine on a network. For instance, to link two computers on a home network you might give them the addresses 192.168.0.200 and 192.168.0.201.

■ LAN (Local Area Network): This is the most popular configuration, used by offices and in homes to interconnect lots of computers that are all situated in the same physical area, so that they can operate independently while communicating with each other.

■ mLAN (music Local Area Network): A Firewire-based proprietary protocol developed by Yamaha for high-speed transmission and control of audio and MIDI ports over a network.

■ NIC (Network Interface Card): The majority of PC motherboards are now equipped with built-in LAN network ports and associated RJ-45 connectors, but if yours is not you can buy an NIC very cheaply, in either PCMCIA format for a laptop or PCI for a desktop machine.

■ Peer To Peer: Also known as ad hoc or point-to-point mode, this is the most common example of two PCs connected together without requiring any other equipment, often simply to share files or a printer.

■ WAN (Wide Area Network): Generally a bunch of computers or LANs, but in different locations, connected via a phone line or satellite link. Once again, each machine operates independently while communicating with the others.

■ Wi-fi: Also known as 802.11 networking and wireless networking, this is a very convenient way to install a network, since the various computers can be in different parts of your home or studio and don't need cables running between them. For musicians, it has the added advantage that you can place noisy backup machines in the garage or loft. The IEEE (Institute of Electrical and Electronic Engineers) created three standards for wireless networking. The first, 802.11b, is the slowest and cheapest, transmits at 2.4GHz, has a bandwidth of 11 Megabits per second, and is part of Intel's Centrino specification; 802.11a operates at 5GHz and can handle up to 54 Megabits per second; and 802.11g offers the same maximum bandwidth but operates at 2.4GHz, to bring the price down. None of these technologies seems to generate any audio interference problems in a studio environment, although 802.11a is less prone to interference from wireless phones, microwave ovens and other communications devices. However, if you have the choice, wired networks nearly always offer much faster performance.

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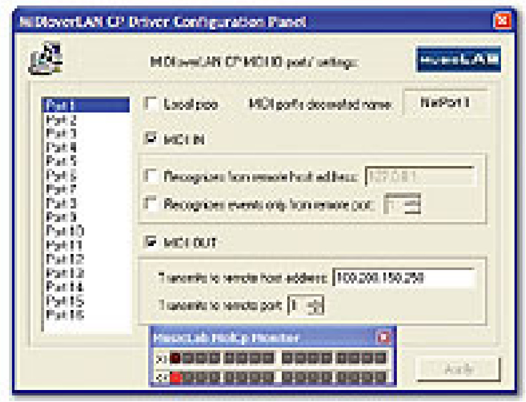
MIDI over LAN

If your PCs already have some type of network port, you may not even need to have a hardware MIDI interface on each one to run synchronised music applications. *MIDI over LAN* (www.musiclab.com) was designed as a simple, robust utility to connect MIDI applications spread amongst severally locally networked computers. However, it also allows you to interconnect several MIDI applications on the same computer, so that you can port MIDI data from one to the other at will.



With a Local Area Network connecting your computers, the *MIDI over LAN* utility provides a simple and rugged way to send MIDI signals between them, without needing hardware MIDI interfaces.

In 2004, a major update turned this utility into the rather more ambitious *MIDI over LAN CP* (Cross Platform), enabling you to create a MIDI network containing both Mac and PC machines. MIDI timing accuracy is also claimed to be even better than in previous versions. It requires Windows 2000 SP4 or XP SP1 and later, or Mac OSX 10.2.4 or later. Now at version 2.2, *MIDI over LAN CP* is available in a Standard edition that supports up to 16 'Netport' MIDI devices (each device acts like a standard MIDI port that provides up to 16 simultaneous channels, so the maximum total number of MIDI channels is 256), while the Platinum version supports up to 64 ports, for up to 1024 MIDI channels. Each *MIDI over LAN CP* device also has multi-client driver support for up to four clients, making it easier (for instance) to run a sequencer output and synth editor simultaneously to the same destination synth.

Prices start at $129 (about £70) for the most typical two-computer setup ($169 for the Platinum version), and jump to $199 ($269) for a bundle for four computers and $299 ($369) for eight. A 1.4MB demo version that runs for 14 days is also available to download. I had no problems installing and getting the Standard version running. So that it can be configured to suit your setup, the MIDI In and Out can be enabled independently for each of the available ports (labelled Netport1 to Netport16) and the MIDI In can be configured to receive data from any host PC, or only from a specific IP Address (see 'Technical Terminology' box). The MIDI Outs must always be set to point to the recipient PC's IP Address. There's a handy MIDI monitor that displays *MIDI over LAN* activity, and overall I found that this utility worked faultlessly on my PCs.

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Open Sound Control

OSC (Open Sound Control) is rather different from most of the other networking options discussed here, because support for it needs to be implemented in the application itself. It's defined as an 'open, network-independent protocol developed for communication among computers, sound synthesisers and other multimedia devices', and lets you send MIDI data between multiple computers over any Local Area Network (LAN) or the Internet. Various scenarios are suggested as possibilities, including Internet-based collaborations between musicians, co-ordinated synthesis between two or more computers to increase processing power, and sound installations with dozens of computers in a single room.

Some PC music applications that support OSC include Plogue's *Bidule* modular environment (www.plogue.com), and Cycling 74's *Max/MSP* real-time audio development environment (www.cycling74.com). Anyone interested in Jazz Mutant's amazing Lemur controller (www.jazzmutant.com) will also discover that it communicates using a LAN interface and the OSC protocol. However, possibly OSC's biggest commercial supporter is Native Instruments, whose *Intakt*, *Reaktor* and *Traktor* synths have all got OSC extensions. Support for OSC has been extended in the just-released *Reaktor 5*, which has a dedicated OSC Settings window where you activate it, set up synchronisation and scan for and display other OSC members to whom a connection has been established.

NI consider OSC to be a sadly overlooked technology that's far more powerful than MIDI, and other enthusiastic users also point to improvements over the MIDI spec, such as its much greater 32-bit resolution for altering parameters such as filter frequency in real time, and the fact that no extra interface is required. If you want to find out more, Open Sound Control's Home Page is at www.cnmat.berkeley.edu/ OpenSoundControl.

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Steinberg VST System Link

If your MIDI + Audio sequencer is Steinberg's *Cubase SX/SL*, *Cubase VST* 5.2 or *Nuendo*, you already have the proprietary VST System Link networking functions built in. To use them, you need an audio interface on each computer, running ASIO drivers, and with an ADAT, AES/EBU, S/PDIF or TDIF digital audio connection. The networking signal is then sent between the machines using one bit of the existing digital audio signal. It makes no audible difference to audio quality.

If you've already got one of these applications installed on all the computers in your proposed network there's nothing else to buy. However, If not, Steinberg's *V-Stack* is a cheap way to expand your VST System Link studio. It's a simple host application that lets you run up to 16 VST Instruments with up to five VST or DX insert effects and eight send effects per channel, as well as providing four-band EQ and up to four subgroups in its integral mixer. It can also be used as a simple stand-alone host to run VST Instruments in a live setup, and provides layering, transpose and split functions. Version 1.2 runs on both Mac and PC and you can buy and download it directly from the Steinberg web site for just 50 Euros. *V-Stack* certainly provides a familiar interface and I had no problems using it as a stand-alone host.

Steinberg's *V-Stack* is another simple host application, with the advantage that it can be linked to *Cubase* or *Nuendo* running on another computer, via a simple digital audio cable, for sample-accurate synchronisation.

With System Link, since the digital audio cables already carry a clock signal there's no need for separate sync, and linked machines should stay locked together with sample accuracy. One device should run from its internal clock and therefore be the Master, providing the clock signal for all the others that are set to external clock as Slaves. The only exception is when you're using a dedicated word-clock generator or digital mixing desk, in which case all the audio interfaces should be set to Slave mode.

Those with multiple copies of *Cubase* and/or *Nuendo* and *V-Stack* should find System Link an elegant solution, but unfortunately success is not 100 percent guaranteed. Steinberg have found that the drivers of some makes and models of audio interface simply don't comply sufficiently well with the ASIO 2.0 protocol for them to work reliably with System Link, so a few may suffer from clicks, pops and crackling. I personally experienced some lockup problems when attempting to link multiple PCs. It's worth reading the VST System Link Troubleshooting section of Steinberg's Knowledgebase (http://service. steinberg.net:80/ knowledge_pro.nsf) for fault-finding tips.

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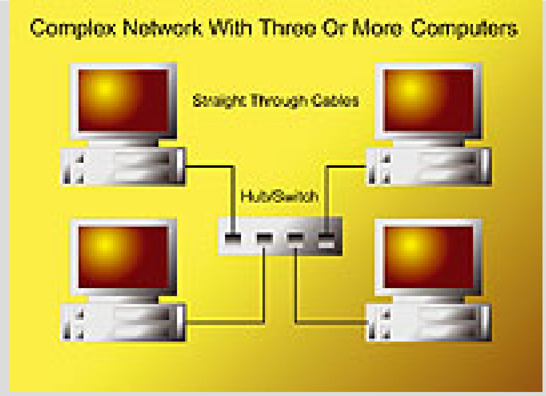
Setting Up A Simple Network

In essence, there are two approaches to creating a network, depending on how many computers you want to link together. The simplest setup is for two machines, each with RJ45 LAN ports, since they can be directly linked via a CAT 5 crossover cable that costs about £5. If you want more machines in your network, you will instead need a stand-alone hardware hub or switch and CAT 5 'straight through' cables connecting each of your computers to it. A hub is generally cheaper and easier to set up than a switch, but since it sends the data received at one of its ports to all the others, your network may become bogged down when lots of data is flying around. Switches are generally preferred on networks containing four or more computers, or when you intend to transfer lots of data, because they only send incoming data to its intended destination, resulting in less network congestion.



Another approach if you have two PCs with Firewire ports is to use these for the network connection, with a standard Firewire cable between the two. If, on the other hand, the PCs on your proposed network have wireless (Wi-fi) network support, you don't need any cables at all, although the much lower bandwidth won't be very suitable for transferring real-time audio streams.

People's experiences of setting up a network vary enormously, from those who plug in the appropriate cable to each computer and find everything works immediately without any further setup, to those who spend days trying to get things working properly, often due to some obscure hardware parameter setting. Fortunately, Windows XP is generally regarded as the easiest Microsoft operating system to date on which to set up a network, and there are copious details on creating a home or small office network in the integral Help and Support Center, and a Network Setup Wizard that you can run by double-clicking on its icon in Control Panel (you'll need to do this on each PC destined to be connected to the network).

If you still run into problems, you'll also find plenty of step-by-step guides on the Internet. For a beginner's guide try 'How Home Networking Works' on the How Stuff Works web site (http://computer. howstuffworks.com/ home-network.htm). Those of a more technical bent may prefer Tweak Town's Windows XP Home Networking Guide (click on the 'Guides' link from the www.tweaktown.com home page).

By the way, if you can't get your new network to work at all and have followed the advice of some of the web sites that suggest 'unwanted' Windows services, you may have disabled some that you now need. One of the best places to find out which services are required and which are not is the Elder Geek's Service Guide at www.theeldergeek.com/services_guide.htm.

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FX Teleport

System Link will obviously appeal to anyone who already uses Steinberg software and who has an S/PDIF-equipped interface in each computer. However, *FX Teleport* (www.fx-max.com) is another very clever network-based utility that takes a rather different approach: it appears to the audio application running on your main PC as an extra set of plug-ins and soft synths that are, in fact, being run on remote PCs via a 100Mbps or Gigabit TCP/IP LAN connection.

As with *MIDI over LAN*, this means that you don't need MIDI interfaces on each computer, but even better for those on a budget is the fact that you don't need to run a host application on the remote PCs either. The installed VST-format plug-ins and instruments on the remote PCs simply appear to your VST-compatible host application (*Cubase*, *Nuendo*, *Sonar* and so on) on the main PC as 'LAN' versions. Best of all, since the remote VST plug-ins rely on audio send/return streaming via the LAN connection, and the remote VST instruments return their audio outputs via the the same means, you don't even need an audio interface on the slave computers.

A single 'FXTeleportSetup.exe' file is supplied that you run on each PC in your network. During this process, you need to designate the machine running your DAW host/sequencer as the Master (Host) computer, and each of the others that will run extra plug-ins as Slave (Server). Then you start the *FX Teleport Server* utilities on each Slave PC and run the *Install Effects* utility on the Host computer. The latter will scan the VST plug-in folders on the various slave PCs to find what 'remote' plug-ins and instruments have been installed there and add these as options inside the VST plug-in folder of your Host PC with '(LAN)' appended to their names.

When you first launch one of these new options, a window pops up to inform you that 'Network Latency detection is in progress'. *FX Teleport* is calculating the extra delays involved in sending/receiving the audio signal via the network, so that it can compensate for them automatically, just like a DSP effects card such as TC's Powercore or Universal Audio's UAD1. On my PC this latency was 512 samples (11.6ms at 44.1kHz).

There's little to see on the host PC once *FX Teleport* has been installed, other than a new FX Teleport folder in your plug-in and Instrument lists, containing the (LAN) versions of everything installed in the remote PC VST plug-ins folder, and these small information windows displaying connection and preset details. Detailed editing is performed on the remote PC.

You can choose presets from either Slave or Host PCs, but more in-depth parameter editing is done on the Slave's screen. However, the tweaks you make there are remembered at the Host end and saved with your song, so that the next time you open it the relevant LAN plug-ins are reloaded on the Slave machines, with their settings intact. Any automation data relating to the *FX Teleport* devices also remains on your main PC, to be saved with your song. This is far easier to manage than having to save another set of data for each song on each Slave PC, to remember the plug-ins, instruments and settings associated with it. The only disadvantage I can think of is that DirectX plug-ins aren't supported.

Overall, *FX Teleport* (see screen above) provides such an elegant approach to distributed processing that I'm not surprised so many musicians are already using it, especially as its most popular '1 host + 1 server' configuration costs just $99, and the 'per server' price drops for more complex setups. FX-MAX also have *Giga Teleport* in their range, which performs similar magic for remote PCs running Tascam's *Gigastudio*. It fools *Gigastudio* into thinking that it's running with an audio interface with GSIF format, but in fact sends the audio data over a 100Mbps or Gigabit network connection, like *FX Teleport*. Offering up to 32 audio outputs at sample rates of up to 192kHz, and as many as 128 MIDI channels when used with *Gigastudio 3*, it runs on Windows 2000 or XP and can be used on multiple PCs to achieve higher *Gigastudio* polyphony (subject to network performance).

Of course, given the much greater bandwidth requirements of audio over MIDI data, *FX* and *Giga Teleport* will both benefit from a fast Gigabit network. However, even with the 100Mpbs network I was running I experienced very few *FX Teleport* problems. I chose to mostly run a few demanding reverb plug-ins and soft synths on the remote PC, rather than lots of less demanding plug-ins, so that only a few simultaneous audio streams were active.

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Practical Examples

While researching this feature, I talked to some very different musicians who have used various types of networks with music applications. Samplecraze (www.samplecraze.com) is owned and run by Eddie Bazil (AKA Zukan), a sound designer, sample developer, programmer and musician who has a network of three computers — two PCs and a Mac. "One PC is dedicated to the Internet and nothing else, and the other two are for audio only, for working on different platforms. The network enables all three to share files and for their ultimate upload to the net. By working remotely, I minimise the virus threats considerably. The computers are networked to each other using a switch as opposed to a standard hub. This way all three can act as both slaves and masters and can be routed to each other in any combination".

Visual artist John Dekron and multimedia artist Markus Schneider are already pushing the boundaries with OSC-networked computers in a live performance environment. This is a screenshot captured from their specially developed *ES-X* live performance software, which mixes and treats live satellite TV images while *Reaktor 5* modifies the incoming audio, with overall control via Jazz Mutant's Lemur controller.

Among their many and varied multimedia activities, visual artist John Dekron and multimedia artist Markus Schneider (forming RISC, together with Christian Riekoff) are currently involved in setting up an audio-visual live performance environment, with NI's *Reaktor* generating live audio, synchronised using Open Sound Control (see box on page 156) to Cycling 74's *Max/MSP* graphical environment, which is in turn running *ES-X*, a video live-performance program developed by John Dekron and Michael Brux using Cycling 74's range of Jitter graphic objects for the *Max* environment. Overall control of both audio and visual elements will be using the new Lemur controller by Jazz Mutant. Markus told me that "OSC is pretty new in our work setup and right now we're not sure how far we will go with it — it partly depends on the future developments within the protocol and companies supporting it. For a project on June 11 we are going to integrate the Lemur. We will mix satellite TV images live on stage in the Maria Club (www.clubmaria.de) here in Berlin. The TV images will be treated by *ES-X* and the TV audio will run through *Reaktor*. We will control image and sound simultaneously using the Lemur." If you'd like to find out more, visit www.risclab.org or www.thisserver.de/john_dekron.php.

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The Global Recording Studio

You don't have to restrict yourself to physically connecting multiple computers in the same room to network with other musicians: there are also various ways to collaborate via an Internet or similar connection using a WAN (Wide Area Network). One of the simplest is Microsoft's *Remote Desktop*, which you can use with Windows NT Server 4, 2000 Server, Server 2003 or XP Professional. Using this, you can access a Windows-based PC and work with its applications and files from another computer (even from a Mac if using the *Remote Desktop Client* for Mac OS X version 10.2.8 or later).

Alternative remote control software includes *VNC* (*Virtual Network Computing*), a free download from www.realvnc.com. Like *Remote Desktop*, it allows you to view and interact with a remote server computer using a simple 'viewer' program running on another computer anywhere on the Internet. However, if you want to be able to send your MIDI and audio tracks in both directions in a fairly transparent way, you'll need something a little more sophisticated.

Fancy collaborating on a track with other musicians around the globe? *Digital Musician Net* runs as a VST 2 plug-in inside many host applications, transfers audio and MIDI data in real time, and even lets you see your collaborator via a two-way video link.

In 1998, Rocket Network Inc created a way for musicians to share their files, which was taken up by Steinberg at the beginning of 1999. A new 'Rocket' button appeared in *Cubase VST* that let users post tracks to the Rocket Network Central Server so that they could be shared, within minutes, with other musicians collaborating on that project. *Logic Audio* and *Pro Tools* users later got in on the act as well, and the files could be sent in any format, including WAV, MP3 and MIDI. Rocket Network's Central Server held the most up-to-date arrangement of each project, which was downloaded to each musician as they entered a 'session'.

At the time it was hailed as revolutionary, but before broadband was widely available it could be very slow to update (sometimes several minutes elapsed before new sessions were received), so real-time collaboration between musicians wasn't possible. The *Cubase* and *Logic* versions wouldn't talk to each other either. Nevertheless, the Rocket Network was used in some high-powered projects, such as the 2001 soundtrack for the film *Bridget Jones' Diary*.

Avid/Digidesign acquired the assets of Rocket Network Inc in 2003 and subsequently released Digidelivery — a fast, reliable, and secure way to send any kind of digital file via the Internet that has been adopted by many professional music developers around the world. It's particularly useful to *Pro Tools* users, since they can drag and drop their sessions onto the Digidelivery interface to package and deliver all related files.

Fast-forward to 2005 and *Pro Tools* owners can now interact in real time with other *Pro Tools* users, using the clever technology of the *Source-Connect Pro Tools* plug-in (www.source-elements.com). Broadcast-quality audio is transmitted between remote locations via broadband Internet connections, so you can collaborate with other musicians around the world. However, only *Pro Tools* users can benefit from this product, and at $1495 not everyone will be able to afford it.

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Audio Routing & Mixing

One of the reasons why I was particularly impressed with *FX Teleport* is that it enables you to view and edit your entire mix on one computer, even when some of it is actually running on other machines. With all the other 'solutions' mentioned in this article you still have to decide how to combine the multiple audio contributions into the final mix, even once you've got your PCs successfully networked together. There seem to be almost as many alternatives as there are musicians running such systems, so here are a few options to consider.

One of the simplest approaches for anyone who's already using an analogue mixer is to patch all the audio 'PC synth' outputs into it, adding analogue EQ and effects to taste, and recording the overall mix, along with the contribution of any hardware MIDI synths, via the analogue inputs of the audio interface on your master PC.

The preferred approach for many musicians is to use spare soundcard inputs on the host machine to directly patch in the remote audio contributions. This means you don't need an external mixer, and with sufficient audio interface I/O you can stream all your remote soft synths to separate tracks on your host sequencer and treat them there individually with host plug-in effects. Even better, those with ADAT I/O can stream their audio digitally for a cleaner signal.

It's certainly a more elegant solution if you can set all your channel levels, pans, automation and so on, plus plug-in treatments for both local and remote soft synths, so that they can be saved in one host project or song file, rather than as several files across multiple computers. Nevertheless, some musicians prefer to run both soft synths and associated plug-in effects on the remote computers, keeping these settings together, as you do with the presets of most hardware MIDI synths, and then do the mix and automate settings on the master PC. It's up to you.

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Digital Musician Net

For the rest of us, *Digital Musician Net* (www.digitalmusician.net) might be exactly what the doctor ordered. Again, it uses a proprietary plug-in that needs to be run on each computer, but this time the format is the rather more universal VST 2.0. Audio or MIDI collaborations are not only in real time, via a broadband Internet link, but both parties can also see each other and discuss the performance (just as you can in the studio through the control-room window) via twin video-conferencing windows in the plug-in that run at up to 25 frames per second.

Digital Musician Net already runs on both Mac and PC platforms. If used with *Cubase SX 3* and *Nuendo 3* it offers synchronised audio, MIDI, MTC (MIDI Time Code) and remote control, and synchronised audio on *Live*, *Sonar, Logic 7 (*and even *Logic 5.5* on the PC). MIDI, remote control and other applications may well be supported on these and other applications by the time DMN officially launches in August 2005.

Bit rates of up to 256kbit MP3 can be achieved in real time and, unlike Rocket Networks, no central server is required. Audio or MIDI data is sent to and fro directly between collaborators. Time-stamping and intelligent buffering is used to compensate for latency or web congestion, and if your particular link means that you have to resort to a lower bandwidth while recording takes place, you can transfer the final take later on at CD quality, using the plug-in's built-in Send File function.

Digital Musician Net is aimed squarely at musicians and other audio professionals who want to network their services around the world, and there's already been lots of interest from individual artists, producers, studios and film composers. Amazingly, *Digital Musician Net*'s Standard Account is completely free and entitles you to join the *DMN* web site community, where you can meet like-minded souls around the world, advertise your talents and arrange collaborations, the sole limitation being that the audio bandwidth is limited to 128kbps.

A premium Studio Account, available from just 24 Euros a month, ups the bandwidth to 256kbps, provides email support, an extended studio home page to allow you to advertise your services around the world to other professionals, and the ability to set up sub-accounts (so that a studio can let multiple clients use their higher-bandwidth link at different times). *Digital Musician Net* is poised to launch in August 2005, but it was already extremely stable when I saw it demonstrated in early April at the Frankfurt Musikmesse, and I am quite sure that it will prove very successful.

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Network Bandwidth

If you've got a choice of network connections, here are some guide figures showing how their transfer rates compare, from slow to fast:

- Ethernet: 1.25 Megabytes/second (10 Megabits/second)
- Wi-fi 802.11b: 1.38 Megabytes/second (11 Megabits/second)
- Wi-fi 802.11a/g: 6.75 Megabytes/second (54 Megabits/second)
- Fast Ethernet: 12.5 Megabytes/second (100 Megabits/second)
- Firewire 400: 50 Megabytes/second (400 Megabits/second)
- USB 2.0: 60 Megabytes/second (480 Megabits/second)
- Firewire 800: 100 Megabytes/second (800 Megabits/second)
- Gigabit Ethernet: 125 Megabytes/second (1000 Megabits/second)

So, for instance, if all the computers on your network have both Fast Ethernet and Firewire 400 ports, you'll manage more plug-ins and soft synths using Firewire.

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Final Thoughts

As we've seen, music-making need no longer be confined to a single computer, or even a single location. Whether you need more processing power, to run several applications simultaneously, or want to collaborate with other musicians, the key may be creating a music network.

However, before you make a final decision on whether or not to install networking capability on your music PC, don't assume that it will *automatically* provide you with better performance. While there are plenty of musicians providing glowing reports of how liberating their multi-PC setups have proven to be, I've also come across plenty of others who either never got a network to work reliably with their particular machines, who experience dropout problems during audio recording and playback after installing a network interface card, or who have never managed to completely eradicate audio clicks and pops or timing problems in a more complex setup.

Fortunately, nearly all the applications and utilities discussed in this feature can be tried out using free demo versions, so you can test the water before committing yourself, and many musicians I know have tried out two or three programs before deciding on which one to go with. Once you've spread your networking wings you may never want to go back to a one-box solution! SOS

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SOUND ON SOUND
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In this article:

The Lost Art Of Sampling

Part 1

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Print article : Close window

Technique : Sampling

Most modern musicians use samples, even if only in S&S keyboards or virtual instruments. But sampling itself has become something of a lost art. In the first of a short series on rediscovering this skill, we look back at how the technique and the technology developed.

Steve Howell

Sampling technology has become so widespread that it is no longer considered remarkable. Indeed, in many ways, it has become 'invisible' — so widely accepted and taken for granted that no-one notices it any more. Everywhere you look, you'll see sampling in action: in the stored messages on modern ansaphones, in those muffled train and airport announcements you can never quite catch the gist of when you're in a hurry, and in those dreadful menu-driven customer service lines so beloved of companies claiming to offer "a better, more focused customer service experience". Wherever we turn, we have first-hand experience of 'sampling' in one form or another.

It's the same in the modern music-making industry — sampling technology is at work in the vast majority of synth, keyboard and virtual-instrument products on the market. Whereas synths were once all powered by analogue oscillators which generated a limited range of electronic waveforms, these circuits have largely been replaced in modern hardware synths by chips containing samples of a colossal range of instruments, including analogue-synth-style sawtooth, square and triangle waveforms to complete the illusion. Even some of the 'modelled' analogue synths use carefully engineered multisamples of analogue (and other) waveforms as the basis of their synthesis methods.

But even though samples lie at the heart of much of what we do, few of us actually sample audio ourselves any more. So-called software samplers also rarely sample, although they're happy to sort raw audio into multisamples and keymaps if you can find some other way of getting the audio into your computer in the first place. And nearly all sample libraries are supplied on CD-ROM these days, or as sample libraries with a virtual-instrument 'wrapper', which completely side-steps the former need to sample, trim and if necessary loop audio before we can use it. Not surprisingly, outside the select world of the sound designer, there are few people who can be bothered these days.

This is all very well if you remember the days of searching for seamless sample looping points with horror, and can't get enough of purchasing your samples in a ready-to-go, easy-to-use format, but what of all those people who've come to the world of music technology in the last five years, and never learnt any sampling skills in the first place? If this applies to you, or if you're sick of spending soul-destroying afternoons trawling through your vast sample libraries looking for 'the right sound' and would prefer to be able to create your own from scratch, or if you'd just like a refresher course in what sampling is, and what you need to know about it in order to handle it yourself, then this series is for you.

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In The Beginning...

Let's start at the very beginning — what *is* sampling? Strictly speaking, it means 'digital recording and playback' — the act of turning a sound source into a stream of 0s and 1s that a digital processor can deal with. The term was first coined in the late '70s by Kim Ryrie and Peter Vogel, inventors of the Fairlight CMI, to succinctly describe one of the features of their then-new digital synthesizer (for more on the Fairlight, check out our Retrozone feature in *SOS* April 1999, or head for www.soundonsound.com/sos/apr99/articles/fairlight.htm on the *SOS* web site).

The groundbreaking Fairlight CMI.

As with some of mankind's greatest inventions, sampling was originally a kind of 'bonus feature' that was included almost as an afterthought. The Fairlight was designed as a digital synthesizer on which waveforms could be constructed using additive synthesis, or 'drawn' using the innovative light pen and touchscreen. It was also possible to define start and end waveforms and 'morph' between the two to create dynamic tonal movement throughout the course of a note. Of course, the Fairlight also had a sequencer, and as such, was also the world's first 'workstation'. Yet it proved to be the sampling feature that captured the imagination of the musicians and producers who were wealthy enough to slap down the £20,000 required to buy a Fairlight

CMI in 1980. In retrospect, perhaps this is not so surprising — it was already possible to sequence electronic synths, and the idea of being able to do the same with the sound of *acoustic* instruments was a powerfully attractive concept. However, not everyone understood the great leap forward at first — another recurring theme in the story of many of mankind's greatest discoveries! One long-defunct music technology magazine of the day sneered at the new 'sampling' feature, wondering why anyone would want to use such a powerful digital synth in this way, and concluding that it was a gimmick. In fairness, even Ryrie and Vogel regarded their invention as a digital synth first and foremost, and whilst the sampling feature allowed complex sounds to be recorded and played back, it gave you very little control over the sound other than the pitch and basic envelope, whereas the digital synthesis components gave precise control over almost every aspect of a sound. But as time went on, it became clear that people weren't buying the Fairlight for these innovations — it was the sampling feature they wanted, and the technique was to change the world and the music it made... forever!

Of course, even with its headache-inducing price tag, the sampling facilities on the Fairlight were very limited, offering recording times of a few seconds at best (and usually less, as sample RAM was then unbelievably expensive), and all at a very poor quality compared with what we're accustomed to today. A 'looping' facility was added to allow a portion of the recording to play back over and over again for as long as a note was held, but that was all. At the time, however, it was a revolution. The general idea of using audio recordings within larger recordings was not new, and nor was the more specific idea of using short recordings of real instruments playing individual notes to create melodies — both of these ideas had been explored using analogue technology from the 1940s onwards (see the box on the last page of this article), but the digital nature of the Fairlight CMI, and its built-in sequencer, made the concept much more workable and immediate.

Inevitably, the success of the Fairlight meant that other companies began making their own samplers (see the box over the page), and over the years, stiff competition in the field ensured that the specifications of samplers improved out of all recognition, prices dropped drastically, and eventually, we arrived at the situation we have today, where broadly speaking, what comes out of a modern sampler is pretty much what went in. It wasn't always that way, but today's technology means that you don't have to think too much (if at all) about the mechanics of the process.

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Sampling Dissected

Those of you unfamiliar with what goes on beneath the user interfaces of modern samplers are no doubt wondering how exactly the process *does* work. Well, if you remember that sampling is basically digital recording, then in order to begin to understand sampling, you need to understand the principles of digital recording.

Analogue signals, like the sounds our ears pick up, are continuously variable, whereas digital signals are made up of strings of zeros and ones so that a computer can process them. To convert analogue audio into digital data, we need to feed it through an analogue-to-digital converter (or A-D converter, also known as an ADC). This slices the audio waveform up into little sections, each of which is assigned an amplitude (loudness) value which can in turn be expressed as a binary number, so that a digital audio processor can understand it, or a computer can store it. Each of the loudness values taken, properly speaking, is called a sample, so this process is known as sampling the waveform. To replay the sound, the binary amplitude values are fed through a digital-to-analogue converter (or D-A converter, or DAC) which reconstitutes the slices at different loudnesses, and puts them back together in the right order as a continuously variable signal that can feed your speakers, and which you can then hear as a sound.

If this explanation has left you totally lost, try thinking of cinema film — I often find that making this analogy helps people to grasp the concept. As you probably know, cinema film is a strip of still photographs which, when replayed fast enough, gives the illusion of continuous movement. Typically, when making a film, 24 frames (or still photos) are taken every second. The more frames there are in any given second, the smoother the movement on screen will be. 24 frames per second (or fps) is largely accepted as the optimum rate for commercial cinema film.

The same is pretty much true of digital audio. When you record a sound digitally, you are taking a series of 'stills' of the loudness of the continuously variable input signal, each of which is then stored as a binary value in Random Access memory (RAM) or, these days, on hard disk. Then, when you play back the sound, the stored loudness values are played back in the correct order, reconstituted by the DAC as a variable waveform, and what we hear is an accurate replica of the sound we recorded. That's the idea, anyway!

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From Fairlight To Fxpansion

The Fairlight was the hippest piece of music technology for years, but its high price tag meant that it was also very exclusive. Of course, this may have only added to the buzz it generated! For the price of a Fairlight CMI on its release, you could buy a country house in the UK, so only the wealthiest of musicians and producers could buy one. Inevitably, a race began to get the price of the technology down to a more affordable level. New England Digital expanded their Synclavier digital synth to include sampling following the success of the Fairlight, but this was equally expensive.

Emu Systems (who, until this time, were primarily manufacturers of modular analogue synths) were the first to truly take things further with their imaginatively named Emulator. This offered basic 'sampling' features similar to those of the Fairlight in a £5000 wedge-shaped keyboard. They went more upmarket with their £9000 Emulator II (below), which added filters, envelopes and other synth functions, as well as individual outputs, timecode synchronisation and other features into the equation. But both of these were still a bit pricey for most people.

Emu's Emulator II.

Ensoniq's £1200 Mirage sampler set an all-time low price for sampling keyboards in 1985.

In the meantime, while mere mortals waited for the price of sampling technology to drop, some were exploiting the new breed of digital delay lines (DDLs) to 'freeze' (ie. sample) audio and then trigger it. The downside of this technique was that sampled sounds could not be 'played' (although it was possible to tune the sample up and down). The DDLs didn't have any form of storage, either, and so the sound was lost as soon as the unit was switched off. Despite these limitations, and in the absence of anything else, many producers and engineers used DDLs to trigger drum sounds. They also used them to sample, say, backing vocals — they would mix down the BVs into the DDL and then 'spin them in' wherever they were required in the song.

As with anything new, the first DDLs were expensive... very expensive! Probably the first manufacturer to offer this facility was UK company AMS with their DMX DDLs. They were initially mono, although the company later made a stereo version, as well as one that could be played from a keyboard that generated control voltages, although it was several thousand pounds in the UK. But the DMXs had a unique feature for the time — expandable memory — and many bought them so that they could 'spin in' even longer sections of audio. Producer Steve Levine had his AMS fully expanded (at a cost of several thousand UK pounds) when he was producing the band Culture Club in the mid-'80s. Another UK company, BEL, also produced a similar DDL, but this was considerably less expensive, and consequently, the distinctive blue BD80 DDL graced the racks of many studios at the time.

Of course, it was only matter of time before the cost of the technology fell, and in 1983, aided by mass-production techniques, Boss brought out their DE200. This was a flexible DDL in its own right, with modulation for chorus and flanging effects, but it also had (low-quality) sampling, and all for a few hundred quid. This was followed in the mid-'80s by Ensoniq's Mirage, which, at £1200 in the UK, went head-to-head with the S612, the first sampling product from another then-unknown company, Akai. Sequential Circuits joined the fray shortly afterwards with their Prophet 3000 and 2000, but sadly went bust before their sampling products could reach maturity.

The Mirage was a landmark product at the price, but it sounded pretty rough and was a pig to use, with just a two-digit display. It also had a limited sampling memory of just 144K! Akai's S612, on the other hand, offered much higher-quality sampling and longer sampling times.

The mass market really woke up to the potential of sampling following the release of Akai's S900 in 1986. It offered good audio fidelity, up to a minute of recording (albeit at compromised quality), individual outs and advanced sample- and program-editing facilities for the time. It was also rackmount and fitted in perfectly with the emerging world of MIDI. The S900 faced stiff competition on many fronts, as everybody was beginning to make their own samplers, from Roland with their S-series (S50 keyboard and S550 rack) to Yamaha with their TX16W and even Casio, who offered the FZ1. Ensoniq also followed the Mirage with their popular EPS series, which incorporated workstation-like sequencing features. But none of these could really compete with the market supremacy of the S900. In fact, the only real competition came from Emu's Emax, which offered most of the features of the big Emulator II, but at a fraction of the cost.

The former 'gold standard' of hardware samplers, the Akai S1000.

Of course, the S900 paved the way for the CD-quality S1000 and coincidentally, Emu released their similarly endowed EIII at around the same time. From that moment on, it was pretty much a battle between the two giants of sampling, Akai and Emu. Roland hung on in there with the S330 and their new S700 series, but by the early '90s, everything had come down to Akai and Emu, who seemed to operate almost in parallel, each one offering comparable products but with slightly different features and functions. And so allegiances were born, and users of each brand swore an evangelistic commitment to their chosen sampler that rivalled the passions in today's Mac vs PC debates. The final result was probably a draw — the US just seemed to prefer Emu samplers, whilst Europe favoured Akai's.

And then, at the turn of the Millennium, software samplers arrived, and the whole Akai vs Emu debate rapidly became academic. As computing power had increased on both the Mac and PC, it had become increasingly viable to make a software sampler that could give the hardware equivalents a good run for their money. Akai and Emu continued to bring hardware samplers to the market, but unfortunately, the writing was on the wall. The software versions were not necessarily better, but they were cheaper and integrated more easily with users' preferred DAW and/or sequencer. Piracy also played an enormous part in the downfall of the hardware sampler — whereas before, aspiring musos would invest in a budget Akai S2000 or Emu ESI32 with every intention of upgrading to an S5/6000 or E6400 Turbo when budget allowed, those same people could now get a knocked-off copy of a software sampler from a friend or an Internet-based peer-to-peer network for nothing.

Many people (myself included) still prefer hardware samplers for a variety of reasons, including sound quality, reliability, portability, and the lack of inexplicable software conflicts, latency, and CPU strain, but from a standpoint halfway through the first decade of the 21st century, it is clear that hardware samplers have had their commercial day, and the future of sampling — for the moment, at least — lies with software.

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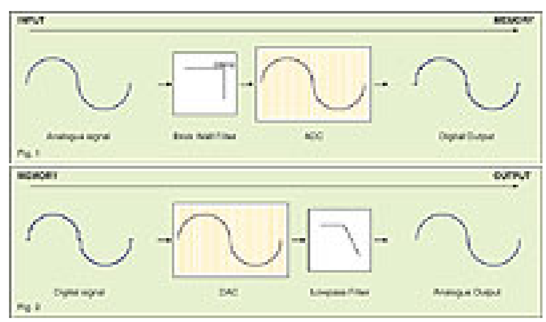
Sample Rates

Sticking with the film analogy for the moment, let's move on to consider the idea of sampling rate. As I've mentioned, it is accepted that 24 frames per second is the optimum number to give an accurate perception of smooth movement when played back through a film projector. With CD-quality audio, it is largely accepted that 44,100 audio slices (or samples) per second is the optimum number to give an accurate perception of the original sound. This figure, expressed as the number of samples taken per second, is known as the sampling rate. 44,100 samples per second can be written as 44,100Hz (Hertz being the standard unit of frequency, expressed per second), or, as it is more commonly written, 44.1kHz. But why such an odd figure?

Well... the highest frequency a human being can hear is agreed to be around 20kHz (20,000 cycles per second). As it happens, this is usually untrue — at best, it's more likely to be 17 or 18kHz, if that! Newborn babies might be able to hear frequencies as high as 20kHz, but as we get older — or attend too many deafening concerts — the highest frequency we can hear tends to drop. To accurately capture any frequency in a digital audio system, you have to sample it at twice that frequency. Therefore, to capture potentially audible frequencies of up to 20kHz, you need a sampling rate of at least 40kHz. This principle was discovered by one Harold Nyquist back in the 1950s when he was working at Bell Laboratories and researching the possibility of digital audio transmission for telephone systems. It is known as the Nyquist Theory. To explain why the sample rate must be twice that of the upper frequency limit, I'm afraid we'll have to return to the film analogy!



A simplified diagram of the sampling/ digitisation process.

No doubt you've seen old, silent moves — Charlie Chaplin, The Keystone Cops, that kind of thing — and the movement seems jerky. This is because in the early days of cinema, they used slower frame rates — typically 16 frames per second, sometimes less. It was thought that this would be sufficient to create the illusion of smooth movement on screen, but as it turned out, it wasn't. Increasing the frame rate made the illusion believable, which is how the film industry arrived at its rate of 24fps.

It's similar with audio. At lower sample rates, the sampled audio is not a true representation of the original analogue input signal, and is 'jerky' — which in audio terms, translates into 'fuzzy' or 'murky-sounding' audio. But there's another reason why high sample rates are required to play back audio accurately. Back to cinema film again!

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Aliasing

You must also have seen films where a moving vehicle's wheels appear to be going in the opposite direction to the vehicle itself. If the wheel is spinning faster than 12 revolutions per second (ie. faster than half the film's 'sampling rate' of 24fps), the wheel completes more than half a revolution per frame, and our brains are then unable to tell whether it got like that by turning more than half a revolution forwards, or by turning less than half a revolution *backwards*. So, for example, if the wheel is spinning at 18 revolutions per second, it will *appear* to be going backwards by six revolutions a second.

The reason for this is that the camera (and our eyes) cannot distinguish between positive and negative frequencies — to the camera, a revolution of 18 24ths is exactly the same as a negative revolution of six 24ths. This phenomenon is known as 'aliasing'.

The same is true of sampled audio. Any frequency that exceeds the Nyquist frequency (ie. half the sampling rate) will be 'reflected back' into the audio spectrum by the amount that exceeds the Nyquist frequency. For example, if the sampling rate is 20kHz (giving a maximum upper frequency response of 10kHz according to the Nyquist theory) and the sound being sampled (for example, a cymbal) has a high-frequency overtone at 12kHz, that overtone will be 'wrapped around' at 8kHz. 8kHz is the Nyquist frequency of 10kHz minus 2kHz, the amount by which the cymbal overtone of 12kHz exceeds the Nyquist frequency. This undesired 8kHz overtone (also known as an alias) is unlikely to be mathematically related to the frequencies in the original sound, which is a prerequisite for sounding pleasingly harmonious to our ears. So the chances are that it will sound highly enharmonic — in a word, nasty! Cymbal sounds in particular consist of many complex frequencies, and are likely to have other overtones at high frequencies which will also be reflected back into the audible spectrum in mathematically unrelated ways, creating further sonic mayhem. You can now begin to understand why early and/or cheap samplers, with their low sample rates, sounded pretty horrible. There was a whole lot of aliasing going on!

So surely a sample rate of 40kHz (and a consequent theoretical upper frequency response of 20kHz, the accepted practical upper limit of human hearing) will guarantee that this will not happen. Or will it? The fact is that many natural sounds *do* contain frequencies in excess of 20kHz. We might not be able to hear them, but that doesn't mean that they're not there. And what we would certainly hear are the aliases from audio frequencies beyond our hearing, which would be reflected back into our audible range around the Nyquist frequency in the manner I've described, even when the sampling rate is 44.1kHz.

The reason we don't hear these aliases in CD-quality digital audio is that modern ADCs filter the audio input to prevent any frequencies above 20kHz entering the system. These are known as 'brick-wall' filters because of the steep shape of their frequency responses, and they rather brutally remove anything above 20kHz to exclude the possibility of audible aliasing making it into digital recordings. You can see the complete process in the diagram on the left. The analogue signal passes through a brick-wall filter, where any frequency above 20kHz is removed. That signal is then 'scanned' and sliced at the sampling rate in the ADC and the resulting digital signal is stored in memory (in RAM or on disk). To play the sound back, the digital signal passes through a DAC where the slices are reconstituted but, because there might be some 'rough edges' in this process (which would be heard as distortion), the signal passes through a low-pass filter to smooth out the signal, and the reconstituted analogue signal is passed to the outputs.

Armed with this knowledge, why is a sample rate of 44.1kHz required, given that to reproduce upper frequencies of 20kHz, you only really need a sample rate of 40kHz? Well, some overhead was added to allow for the various filtering processes involved, and the Audio Engineering Society settled on 44.1kHz as the *de facto* standard for CD-quality recordings in 1985.

Of course, this description is highly simplified, but is pretty much all (if not more than) you really need to know without resorting to mathematics. If you do want to find out more on the intricacies of digital recording, refer to Hugh Robjohns' excellent series on the subject, which appeared in *SOS* between May and October 1998, and can be viewed for free at the *SOS* web site. The first part, which deals with sampling rates, Nyquist's theorem and aliasing, can be found at www.soundonsound.com/sos/may98/articles/digital.html, and links to the other instalments can be found at the end of the on-line article.

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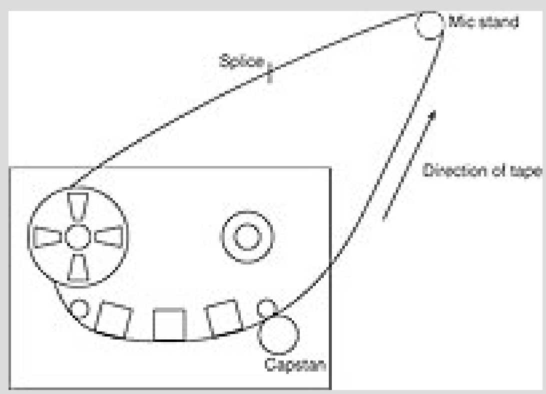
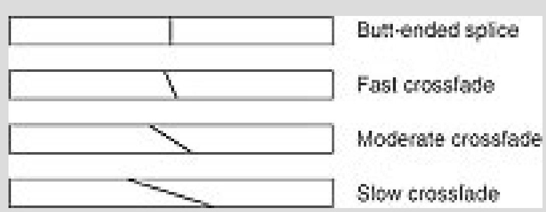
From *Musique Concrète* To The Fairlight

Although the digital approach to sampling introduced by the Fairlight made the process of using recorded sound in larger compositions much easier, the technique itself considerably pre-dated the late 1970s. As far back as the mid-1940s, Frenchman Pierre Schaeffer was pioneering a new musical form called *musique concrète* which involved recording acoustic sounds to short lengths of recording tape, splicing them together, speeding them up and slowing them down, reversing them and otherwise manipulating them to create collages of sounds. His first piece (in collaboration with fellow French composer, Pierre Henry) was *Symphonie Pour Un Homme Seul* and was made up entirely of sounds from the human body. Later works included sounds from locomotives



A diagrammatic representation of how lengthy tape loops used to be set up 'in the old days'.



'Old-school' crossfading: different types of tape splice.

to kitchen utensils.

The technique also used tape loops. Before the age of digitised audio, the only way to have a sound repeat endlessly was to splice a tape recording end-to-end. In *musique concrète* it was common practice to make the tape splices at various angles to create smoother edits, a technique which later made its way into mainstream tape-based recording techniques. With a right-angled splice, the transition from one tape section to another is of course instant, but by using varying splice angles, the recordist could smooth out the transition (see the diagram below). Of course, nowadays, such things are handled effortlessly in any decent audio-editing software with a single mouse drag, but before the age of digitised audio, tape splicing was an art form that required considerable dexterity and skill that could only be acquired through experience.

Once the section to be looped was spliced together, it needed to be run through a tape machine, and if it was a long section which wouldn't fit inside a reel-to-reel recorder, you had to resort to all sorts of Heath-Robinson solutions, including the famous one of wrapping the loop around a spare microphone stand (see the diagram above). This was a hit-and-miss affair, as the tension of the tape had to be carefully adjusted — if it was too tight, you ran the risk of scraping the oxide off the tape (as well as potentially introducing wow and flutter), and if it was too loose, the tape could slip and be chewed by the transport mechanism.

Assuming all this could be set up correctly, a tape section could be made to repeat endlessly. Schaeffer even invented a unique tape recorder called the Phonogene that allowed him to play loops at 12 different pitches from a keyboard, with 12 carefully constructed capstan rollers which (much like gears on a bicycle) could be selected from the keyboard. The Phonogene also had a two-speed motor for octave transposition.

Musique concrète wasn't very popular with the traditional classical music fraternity; the notion of using acoustic sounds in a non-real-time performance was not classed as 'music'. However, that didn't stop composers such as John Cage, Edgar Varèse, Karl-Heinz Stockhausen and Iannis Xenakis experimenting with the form. By the 1950s, Louis and Bebe Barron were forging ahead, using new techniques to create the world's first totally electronic film soundtrack for the classic sci-fi film *Forbidden Planet*. With the aid of unique electronic circuits the couple developed themselves, the soundtrack was recorded employing many of the techniques used in *musique concrète* (splicing, reversing, looping, and so on) to create the stunning sonic backdrop to the film (credited as 'Electronic Tonalities').

Back in the UK, from the mid-'50s onwards, the BBC's Radiophonic Workshop were responsible for bringing these techniques more into the mainstream. The Workshop's purpose was simply to create music and audio accompaniment for BBC programmes. But within that brief, the composers working there exploited the new techniques, including tape splicing and primitive synthesis. If there was one TV programme that catapulted these innovations into everyone's living rooms, it was *Doctor Who*. Although it sounded like the product of a synthesizer and a multitrack recorder, the original 1963 version of the theme tune used neither — it was instead assembled using a combination of tape-splicing, *musique-concrète*, and sound-on-sound techniques. The source sounds were either 'found' recordings or the output from electronic signal generators, processed via nothing more complex than simple tape delays and then painstakingly layered by repeatedly bouncing from tape recorder to tape recorder, adding layers each time and sync'ing the various recorders involved purely by hand and ear, as the BBC had no proper multitrack facilities at the time.

The Mellotron.

Around this time, the BBC took interest in a new product that had just come out — the Mellotron. Often erroneously described as the world's first sampler, this was in fact a playback-only tape-based instrument, and it wasn't even the first of those. The Mellotron in turn derived from another tape-based instrument invented as far back as the late '40s — the Chamberlin. The full story can be found in the review of the new Mellotron in *SOS* August 2002 (see www.soundonsound.com/sos/aug02/articles/mellotron.asp), but here it will suffice to say that its inventor, Harry Chamberlin (a keen organist and home tape recordist) supposedly had the idea when recording himself playing the organ. The story goes that he realised that by making one-note recordings of real instruments and triggering the appropriate ones from a musical keyboard, he could create a keyboard instrument with the potential to play back any sound — an unheard-of idea at the time.

The story of the development of the American Chamberlin into the UK-based Mellotron is confusing, and most versions are hotly disputed, but it is safe to say that the design of the former inspired the more reliable latter! Not surprisingly, the more robust Mellotron proved more popular in the end, and is now revered as a classic instrument, perhaps due to the many influential bands that used it. These included the Beatles, the Kinks, and the Moody Blues in the '60s, and later King Crimson, Yes, Genesis, and Tangerine Dream. More recent users include the eels and Oasis.

However, the Mellotron had potential for use as more than just a musical instrument. In the early '60s, someone at the BBC had the idea of using Mellotrons as sound-effects generators with different effects on each key, and so several were made especially for the Beeb to add sound effects to TV and radio programmes with. From the late '60s onwards, synthesizers, which were at first marketed as being able to create any sound electronically, began to make inroads into the Mellotron's domination. However, the early synths were all monophonic, and so the Mellotron remained popular wherever something capable of playing chords was needed. Gradually, though, the frustrations of using them put people off. They were very heavy and could be unreliable. What's more, the tape-based instrument recordings were not looped — so they lasted a finite eight seconds (at best) and then stopped, often very ungracefully!

Rick Wakeman (of Yes, and also the session Mellotron player on Bowie's 'Space Oddity') was so frustrated by this limitation that he funded the development of Dave Biro's Birotron (shown below). This used continuous tape loops that rotated constantly — when a note was played, the tape was simply pressed against a tape head and playback began at any arbitrary point in the loop. To overcome the wow and flutter and slew that was inevitable when the rotating tape loop made contact with the head, each key had a simple attack parameter to soften the attack and hide the artefacts. But the Birotron wasn't a success, and only a handful were ever made.

However, by this time, playback instruments were being developed in the most unlikely places. Toy manufacturer Mattel — known best for their Barbie doll — were responsible for the Optigan. This used optical disks to play back pre-recorded instrumental sounds, drum patterns and musical accompaniments. The discs had waveforms on them which were read by a light-bar reader inside the machine and then turned into sound, much like the old optical soundtracks used on film. Like the Chamberlin and Mellotron before it, the Optigan's principle market was home organists, and, by buying pre-recorded disks, you could 'acquire' accompaniment in different styles to play along with. In many ways, the Optigan pre-empted the various 'arranger' keyboards of today. But like the Chamberlin and Mellotron before it, the technology used was embryonic and unreliable. The disks were scratchy and the sound quality wasn't good, and as such, the Optigan was not a success.

The very rare Birotron (seen at the now-defunct UK Museum of Synthesizer Technology in the mid-'90s).

Mattel's Optigan. This one was photographed in Damon Albarn's studio in the late '90s.

By the late '70s, the advent of the polyphonic synthesizer pretty much spelled the end of the analogue playback keyboards. The original tape-splicing techniques had also largely been abandoned as being too time-consuming — just a few minutes of *musique concrète* could take weeks of work or more. The last creative and commercial hurrah as far as the old tape-looping techniques were concerned was arguably UK band 10cc's 'I'm Not In Love', in 1975 (the detailed subject of *SOS*'s Classic Tracks feature in June this year).

What is evident throughout this story is that musicians were obviously keen on the idea of capturing and manipulating acoustic sounds, no matter how awkward or time-consuming the methods required. It was no wonder that when the Fairlight appeared, its sampling function was embraced so warmly.

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Higher Rates

Of course, times have moved on since the development of the CD-quality specification in the mid-'80s, and we're now looking at possible rates of 96kHz or even 192kHz for digital recording, which result in ever higher upper frequency responses of over 40kHz and over 90kHz respectively. This fits with some audiophile schools of thought, which maintain that whilst they are not audible in the conventional manner, sounds over 20kHz can nevertheless be 'felt' or perceived in some way. However, we do have to look at this within the context and perspective of current audio recording and reproduction technology.

Most mics can't record frequencies much above 20kHz (maybe 22kHz at best) and most amps and speakers cannot reproduce frequencies much above these figures either. Some pro monitor speakers can extend up to 40 or 50kHz, as can specialised amps, but your average amp/speaker isn't going to get close! Also, there are very few A-D and D-A converters that can accurately handle these rates either unless you start looking at specialised (and expensive) outboard converters. And then there's the fact that many conventional musical sounds don't have frequencies that come anywhere near even the 20kHz upper frequency limit of the CD standard. Knowing this, it starts to seem a bit silly when you see processing power and extra storage space being eaten up to record at these supersonic sample rates — particularly for kick drums and basses. Even sounds such as piano, strings, guitar, and most drums and percussion can benefit little from being recorded at these rates — especially when you factor in the technical limitations of analogue recording and playback mechanisms, such as those found in even modern mics and monitors.

Nevertheless, there is no doubt that there are many in the modern recording community who swear that they have (say) Fender Precision bass samples recorded at 96kHz that sound superior. For the record, my opinion is that once said Fender bass (or whatever) is buried in a track with other instruments, any hypothetical benefits the ultra-high sampling rates might bring are largely going to be lost — especially when the track is mixed down for use on a 44.1kHz CD and played through the average hi-fi... or worse, made into an MP3 and listened to through iPod earphones on a tube train!

However, I don't wish to start sounding too much like an instalment of *Grumpy Old Men*, so let's summarise. In practice, 44.1kHz is more than enough to adequately sample most instruments for most musical (and non-musical) applications for playback on most systems. If you want to sample at higher frequencies, that is, of course, your decision if you have the equipment to do so, although it will of course make your sampler work twice as hard (or more) to achieve whatever sonic improvements it does. Certainly the polyphony will be restricted at the higher rates in hardware samplers, and in software samplers, the host CPU will have to work harder, which will either result in the same restrictions or possibly in worse, more intrusive problems, such as dropouts, clicks or outright crashes.

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Next Month

That says as much as I am going to say on the subject of the horizontal axis in sampling, or the frequency. But what about the vertical axis, or amplitude? That's something for next month. SOS

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SOUND ON SOUND
The World's Best Music Recording Magazine

In this article:

Update: Cubase SX/SL v3.1

Cubase Notes

Published in SOS August 2005

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Technique : Cubase Notes

We investigate the major and minor new features on offer in *Cubase SX/SL* 3.1.

Mark Wherry

Last year's *Cubase SX 3* release was perhaps the most significant version of *Cubase SX* to date, and while it offered a mature feature-set for the most part, there were still a few areas that could have been fine-tuned. However, Steinberg have been listening and, by the time you read this, *Cubase SX* 3.1 should be available as a free upgrade for *SX3* users, with a similar update for *Cubase SL* users to follow.

Get Yourself Connected

Cubase SX v3.1 enables you to define external instruments in the VST Connections window, meaning that you can work with them inside *Cubase* in exactly the same way you would VST Instruments.

One of the headline (*SX*-only) new features is the second phase of Steinberg and Yamaha's Studio Connections initiative, dubbed Studio Connections Audio Integration, making it possible to integrate external MIDI instruments into the *Cubase* environment and use them in the same way as if they were VST Instruments. This is achieved in a similar way to integrating external effects units into the VST Mixer (a feature introduced in *SX3*), via a new External Instruments tab in the VST Connections window. Here you can specify the external MIDI device that will receive the MIDI data, and the inputs on your computer system's audio hardware to which the outputs of the device are connected.

Once the configuration is done, you can choose this external instrument via the

VST Instruments window and *Cubase* will create a mixer Audio Instrument channel for its output. The external instrument is now functionally identical to a VST one (you can route its audio around the VST Mixer, using insert and send effects, for example), with the exception that you can only use one instance of each external instrument in a Project. The external instrument's editor will be a Device Panel, if one has been assigned, and the VST Connections window allows you to make delay adjustments to compensate for any latency.

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Into The Mix

Acknowledging the fact that many people use Key Commands in *Cubase*, it's now possible to carry out certain actions in the VST Mixer via Key Commands. This includes bypassing EQs, sends and inserts, loading and saving mixer and channel settings, configuring metering and toggling visible channel types. Usefully, you can also specify whether Key Command actions should affect all channels in the mixer, with the ability to exclude input and output channels, or just the selected channel.

Other improvements to the mixer include the ability to disable the pan control on an audio-based channel by Alt/Option-Shift-clicking it, and there's also a new Equal Power mode in the Project Setup window's Pan Law settings, where the power of the signal remains constant no matter what the pan position — useful for smoother-sounding pan movements across the stereo or surround field. Another welcome enhancement is the ability to copy and paste the settings of multiple mixer channels in one go. Finally, *SX* now supports *Nuendo*'s optional Dolby Digital and DTS encoders. Once installed, extra options are added in the Export Audio Mixdown for these formats.

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It's The Little Things

As usual, this update is packed with smaller features that are as significant as the headline grabbers — certainly in that they aid workflow and improve usability. The first of these will be welcomed by those who remember 'classic' *Cubase* and the ability to glue multiple parts simultaneously: yes, you can now do this in *SX/SL*: select the parts you want to glue together on the same track and click with the Glue tool, or Alt/Option-click a part with the Glue tool to join that part to all successive parts on the same track.

There's also a Preference for automatic colouring of tracks — something I've definitely wanted — and, on the subject of aesthetics, another new one for enabling cross-hair lines to appear when you're dragging Objects around the Project window.

If you find yourself working with Automation tracks, you might find the new Track Folding features useful. These are accessible from a new Track Folding sub-menu in the Project menu, or via Key Commands. The Toggle Selected Track command shows and hides Automation tracks for the currently selected track, and the Show Used Automation and Hide All Automation commands have been moved into this sub-menu as well. Fold and Unfold Tracks closes or opens all Folder tracks in the Project, and the former option works in conjunction with a new preference called Deep Folding, which closes all Folder tracks within a Folder track. Still on the subject of Automation Tracks, there's now an Extract MIDI Automation feature, for converting controller data into Automation data on a track.

Another of the new features is great for those who work a lot with time signatures and tempo changes, especially when writing to picture. This is the Process Bars window in the Tempo track. Generally speaking, Process Bars is to musical time what Process Tempo is to linear time, and enables you to insert, delete, reinterpret or replace bars within a specific range, keeping track of any tempo and time-signature changes in your Project.

Overall, *Cubase SX*/*SL* 3.1 is a tremendously useful update that will be warmly welcomed by current users of v3. If you haven't upgraded from *SX1* or *SX2* yet, now might be the time. SOS

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SOUND ON SOUND
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In this article:

Using The Malström Synth In Reason

Workshop

Published in SOS August 2005

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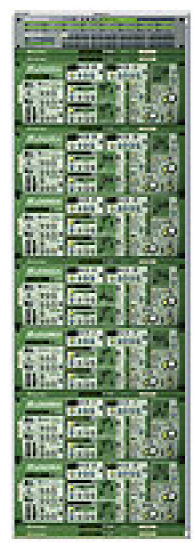
Technique : Reason Notes

The *Malström* device can certainly be used as a conventional synth, but behind its slightly bizarre green front panel there's a powerful graintable sound engine begging to be exploited to the full. We show you how.

Simon Price



Most software studios or sequencing packages now come with variations on subtractive 'analogue' synths, and also samplers. *Reason*, of course, has the *Subtractor* synth, and the *NN19* and *NNXT* samplers. However, *Reason* also has a little gem of a synth that dares to break away from these common sound-generation engines. The *Malström* synth uses a complex synthesis method called Granular Resynthesis, but manages to package it in a way that makes it accessible to anyone familiar with more run-of-the-mill instruments. Learning to understand and control the *Malström* is well worth the effort for the new sounds you'll be able to create, but is also the perfect first step for anyone interested in learning about granular synthesis in general.

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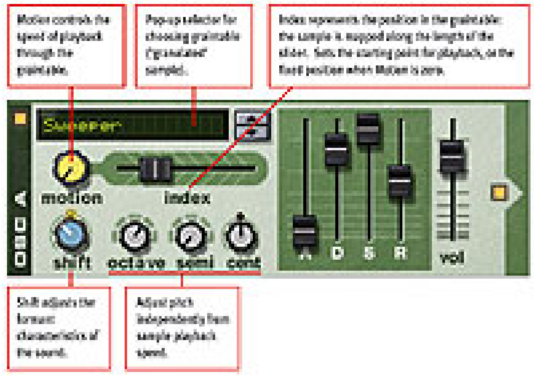
Malström Basics

First off, the *Malström* synth looks a lot like any regular semi-modular soft synth or sampler. There are two oscillator sections, amplitude envelopes, filters, a filter envelope, LFOs, a 'Shaper' and signal-routing options. Much of the synth does operate in the same way as more familiar synths, except for the sound-generation method employed by the oscillators. In fact, by ignoring some parameters you can use the *Malström* as a fairly standard subtractive synth. Create a *Malström* and have a play.

The default patch is a simple sine wave. Clicking on the mini-display that reads 'Sine' in the Osc A section brings up a pop-up list of 'waveforms' (see screen opposite). I've put 'waveforms' in quotes because, as we'll see shortly, they are considerably more than just samples or waves. Anyway, if you choose Sine, Square or Triangle from the list and ignore the parameters marked Index, Shift and Motion, you can operate the *Malström* like a regular synth. However, choose any sound type other than these three basic waves, and you'd at first be forgiven for thinking that the *Malström* was loading up a sample or 'wavetable', something like what you'd find in workstation synths using the 'Sample & Synthesis' method. In fact, the *Malström* is a lot more sophisticated than that.



The controls of a *Malström* Oscillator section.

The *Malström*'s oscillators use short sampled waveforms that have been analysed, pre-processed and converted into a series of small sections (known as grains). These preset samples have some similarity to wavetables or samples, but as they have been 'granulated' they are called graintables. Each 'grain' of sound has some independence from the overall sample, which allows the granular synthesis engine to manipulate the pitch and speed of the sound independently. The best way to hear how the *Malström* works is to pick a graintable of a distinctive sample. If you follow these steps you should get a feel for what is happening and an idea of the possibilities of this synthesis technology.

***** Create a new *Malström* and choose 'Voice: Elektronik' in Osc A's pop-up selector. Osc B is off by default, so ignore that for now. Play the *Malström* from your keyboard and you'll hear the sample (the vocoded word 'electronic') being played in a forward loop. Different keys play the sample at different pitches, but notice that it's always at the same speed. The way each sample loops is preset in the *Malström*. The 'Elektronik' waveform plays in a forward loop, but some others play forwards/backwards, and a few play forward until a loop point, as with most sampler patches. However, you can mess with this basic starting point quite a lot, as you'll see shortly.

***** Try adjusting the Octave knob. Again, notice that the sound is re-pitched but plays at the same speed.

***** Next, try the Motion knob. Turn this up or down slowly and you'll hear that it sets the speed at which the sample plays back, again independently of pitch. Playback still extends across the full range of the sample, but as you get down towards the bottom of the Motion control you effectively won't hear much beyond the starting point. At the minimum setting, playback just loops around a tiny section of the sample.

You may have noticed that when you pitch the oscillator up, as well as the speed staying constant, the characteristic 'chipmunk effect' associated with speeding up a vocal sample is not produced. An advantage of granular synthesis is that the formants in the sound (the resonant frequencies of a sound source that determine some of its character) gain a degree of independence from the parameters of pitch and speed. The Shift control adjusts these formant characteristics of the sound.


The Graintable pop-up.

***** Push up Shift and you'll hear that you can, in fact, add the 'chipmunk' sound without speeding up the sample. The Shift control dramatically alters the tonal and harmonic characteristics of the sound sources, and can add lots of movement to a patch when swept with an LFO.

***** The last of the granular controls is the Index slider. Again, it's easiest to hear what this is doing with the 'Elektronik' sound preset. Reset the Pitch and Shift controls to their default positions and make sure Index is set fully left. Now, turn the Motion control fully anti-clockwise, and hold down middle 'C' on your controller keyboard. You'll just hear a continuous buzzing sound, which is the first grain looping rapidly. Now very slowly sweep the Index control over to the right. You'll hear the whole sampled word play back.

As you'll probably gather, Index represents the position in the sample, with the sample mapped out along the length of the slider. As you sweep the Index fader to the right, each tiny element of the sample is being looped in turn, re-synthesising the original sound in full at whatever speed you move the control. When Motion is at minimum, the Index control determines which grain is playing back, but with any other Motion setting Index determines the starting point where the graintable begins playback each time a key is pressed.

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The *Malström* stereo trick

The *Malström*'s two oscillator sections both produce mono signals, and much of the time these are used to blend two different sounds together. However, there is a Spread knob in the bottom-right corner of the front panel that pans Osc/Filter A to the left and Osc/Filter B to the right. By setting both Oscillators to play slight variations of the same sound, then spreading them left and right, you can create really lush stereo patches. In the example patch above I've given nearly all the controls the same settings in Oscillators A/B and Filters A/B. Osc B is not routed via the Shaper, and the Shaper is off so as not to affect Osc A. The variation between the two sides is achieved by setting the two Index controls fractionally apart. This puts the two oscillators slightly out of step with one another (out of phase), and the whole patch suddenly jumps into a wide stereo spread.

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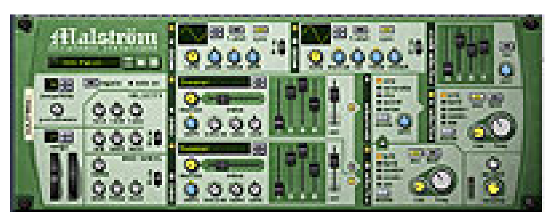
Using The *Malström*

To start using the *Malström*, you don't need to understand all these controls at once. You can just select the Oscillator waveforms and use these as basic starting points — as if the *Malström* were a sampler with a limited number of preset sounds. However, the fun comes from experimenting with the controls and seeing how they interact with one another, then bringing the LFOs ('Mods') into play to create movement and development in a sound. For example, the Index slider sets the start position in the loop, and Motion then sets the speed at which the loop is swept. However, if you start sweeping the Index slider itself, using Mod A, you have two interacting and competing parameters that introduce unpredictability into the patch. Alternatively, by turning Motion right off and using Mod A to control Index you can set exactly how the graintable plays back. The Mods have a huge number of wave shapes, enabling you to control the Index in slow sweeps, single directions, pseudo-random patterns, or rhythmically. The Single Shot button featured on each Mod only triggers the LFO wave once when you hit a key, allowing you to use a Mod like an envelope.


The example patch.

Some of the graintables provided in the *Malström* can be used in a fairly straightforward fashion, like samples — for example, the guitar plucks or the tabla hit. Others, such as the sweeps or FX sounds, are more like a small collection of diverse material all butted together. These graintables give you a chance to focus in and find new waveforms to act as building blocks for your patches. In the example patch in the screen above I've used the 'Sweeper' graintable as my starting point. When left to loop across its entirety, this sound sweeps through a number of harmonics and different shades. This is too busy and distinctive, so I've tried to pick out one small section. I've set the Motion control to 'Off', so that the graintable doesn't play or loop at all of its own accord. I've then set the Index slider to the point in the sample that I'm interested in.

Lastly, I've set Mod A to sweep the Index by a small amount, so that I just get the bit I want looping. Although there's a limited number of graintables in the *Malström*, there are many smaller waves to be picked out using this method.

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Beyond The *Malström*

One feature sadly missing in the *Malström* is the ability to load your own samples. Because the samples used have to undergo analysis and processing to turn them into suitable graintables (and looping directions and points need to be set), Propellerheads obviously decided it would be simpler to provide a number of presets. However, there are other packages using granular synthesis that allow you to load in your own audio samples. Most notable are *Reaktor*, which features synths such as *Triptoniser*, *Traveliser* and *Grainstates* that employ similar techniques to *Malström*, and *Kontakt*, Native Instruments' sampler package. These can load audio from your hard drive and they take only a few seconds to 'granulate' the samples. With any luck, a future version of *Reason* may have a *Malström* with sample-loading capability and controls to adjust the necessary graintable properties, such as grain size and smoothing.

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Bend It, Shape It

Once you've got to grips with the *Malström*'s synthesis engine, the rest of the instrument should be fairly easy to master. The filters and Mods (posh LFOs!) should be self-explanatory, but it's worth mentioning the switchable routing options and the Shaper module. The front-panel design employs arrows as a guide to where signals flow from one section to the next, and buttons designate whether a route is taken or not. For example, Osc A has arrows and switches for going into the Shaper and Filter B. Osc B can only go into Filter B (and then to the outputs) or directly to the outputs. However Filter B can be routed into the Shaper, so Osc B can get in via that route. The Shaper always goes into Filter A, which in turn goes to the outputs. Of course, once you have access to a bit of this kind of flexibility you always wish you had more. For example, it would be nice to have two shapers so that you didn't have to combine Oscillators A and B at this point, especially when routing in external signals (more on external signals in a moment).

You might say that the Shaper is a 'bonus feature' on the *Malström*, but it has become one of the distinctive-sounding features of the instrument, offering several different effects for distorting and smoothing the signals passed through it. My main advice for this feature of the *Malström* is to experiment, and don't go overboard with it, as it has a tendency to 'take over' the patch.

Given that the *Malström* has this distinctive Shaper section, plus a pair of really nice multi-mode filters, it's perhaps not surprising that Propellerheads added external audio inputs to the back panel, so that other *Reason* devices can be routed through the synth. The two inputs come into the *Malström*'s routing scheme at the same stage as Oscillators A and B. In other words, they are routed to the separate filters and only the first input goes through the Shaper. However, as with the Oscillators, Filter B can be routed back into the Shaper (which then takes the signal through to Filter A). This means that you can either send a stereo signal through and not use the Shaper, or use a mono signal and go through as many sections as you like.

This screenshot shows the back panel of a *Combinator* patch in which a *Malström*'s external audio inputs are being used to filter and shape an NNXT. The Malström's filter envelope can be triggered by the same MIDI notes that play the NNXT, or from a Gate input (which can be from the *Malström* itself!). The *Combinator* front panel can also be used to tie together the A and B filter controls for stereo operation.

There are a couple of tricks you can use to bring the Filter Envelope into play on external inputs. One is to connect a CV cable from one of the Mod outputs back into the *Malström*'s own Filter Envelope Gate Input and set a sync'd rhythmic trigger. More recently, *Reason 3* allows you to combine the external device and the *Malström* into a *Combinator* (see screen above). A good reason for doing this is that the same MIDI note that triggers the external sound source can then trigger the filter envelope on the *Malström* at the same time. Finally, another good reason for putting the *Malström* into a *Combinator* is that you can assign a *Combinator* panel knob to control both Malström filters at once, keeping them linked together. SOS

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