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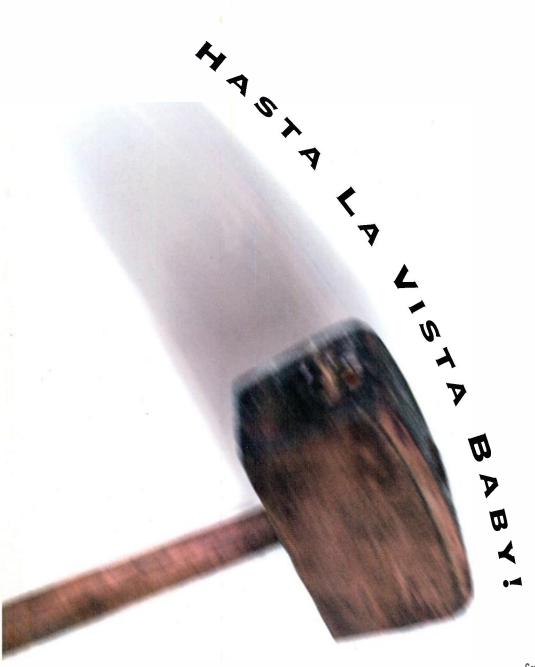## WIN SOUNDCRAFT GHOST MIXER OR EXPANDER MODULE

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leader

As someone whose main instrument is the guitar, I've always felt it ironic that the keyboard synth, despite its enormous range of sounds, is singularly lacking in means to apply true expression to a performance. Admittedly, there are pitch-bend and mod wheels, aftertouch, and sometimes even ribbon controllers, but these are no substitute for what a guitar player can achieve just by bending strings or by picking a note in a different way.

Alternative controllers are available, and two different types are reviewed in this issue, but the technical problems in making them work are not insignificant. Wind controllers, for example, tend to offer a 'halfway house' solution to translating traditional playing techniques into MIDI data, because although it's easy to fit the keys with switches, the process of analysing what goes on at the mouthpiece and then turning that into appropriate controller information is extremely difficult. Breath pressure is straightforward enough, but wind players can also subtly change the pitch of a note, add vibrato, or change the harmonic content of a sound by altering the shape of their mouth. Physical modelling has provided us with a better approximation of this, but we still lack the most effective means of control.

Guitar synths (such as the Roland GR30, reviewed on page 192) represent another technical challenge, because some means has to be devised to measure the pitch of the string before the appropriate MIDI pitch can be called up. A guitar string produces a very complex series of harmonics, making it quite difficult to extract the fundamental pitch. What's more, unlike on a keyboard, where a Note Off message is related to the opening of a switch, a guitar note decays steadily until its pitch can't be detected any more, making it very difficult to play long notes unless a sustain pedal is used. Zeta's MIDI violin, which we also happen to be looking at in this issue (see page 228), faces similar problems — possibly even more, as the start of a violin note can be changed radically by applying different bow techniques.

Despite their problems, modern guitar-to-MIDI devices and MIDI violins are highly advanced pieces of technology, and are capable of excellent results, providing you avoid any playing tricks that might confuse them. They offer the advantages of independent string bend and vibrato, and unlike the vibrato you get with a mod wheel, the rate is controlled entirely by your fingers.

Perhaps the designers of keyboard instruments could benefit from examining more closely how non-keyboard instruments are played, and then apply what they learn to building more appropriate performance controllers for keyboards. Breath controllers, for example, are hugely underrated, but still only control one parameter at a time. A more sophisticated breath controller would be far more useful — one which responded to both pressure and embouchure would be nice. We also need a hardware unit fitted out with knobs to allow controller offsets to be set up — this is a very organic process, and is totally unsuited to button-driven menu access. If the hardware approach is too costly, it should be possible to set up very sophisticated control parameters using a page of software faders within a sequencer package — it just needs somebody to believe that it would be useful.

While I don't pretend to know all the answers, at least I know some of the questions, and I'm convinced that the future of synthesis is dependent on better performance control systems being developed. Until then, beneath all the pizzaz, the synth is still only one evolutionary stage beyond an electric organ played through a wah-wah pedal!

Paul White *Editor*

KORG NS5R

66
Korg NS5R

REVIEWS

FEATURES

WIN SOUNDCRAFT GHOST 24-CHANNEL MIXER p256

in this issue
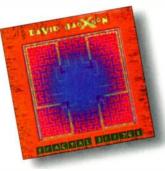
Crosstalk

Beam Me Up

I read with great interest the 'Theremania' article in November 1996's issue of *Sound On Sound*. However, the theremin is certainly not nowadays the only instrument playable without physical contact. Since 1988, we at The Soundbeam Project have been promoting and marketing Soundbeam, an ultrasonic movement-to-MIDI converter, manufactured for us by Robin Wood of EMS in Cornwall. The Soundbeam enables any MIDI instrument to be controlled by a performer's movements in relation to its invisible beam, and has a variable range of between 25cm and 6m. The system was developed in the '80s as a result of research by composer Edward Williams into ways of enabling dancers' movements to play electronic instruments. As well as being widely used in dance, theatre and museum and gallery installations, Soundbeam has established an important role in special-needs education. Over 1000 units are in use worldwide.
Mark Newbold
The Soundbeam Project

Derek Johnson replies: *Thanks for the letter, Mark. When writing reviews there is never enough space to go into as much detail as one would like, and broad generalisations are often the order of the day. I was actually thinking about the Soundbeam as I wrote*

The Soundbeam.

those words, but I decided that I was justified in not mentioning it at that time, since the various flavours of theremin currently being manufactured by the likes of Big Briar (Bob Moog's company) and Longwave Instruments do represent virtually the only readily-available non-contact controller with integral sound source. The Soundbeam, for example, is a controller only, and needs MIDI equipment attached to it to make a sound. Interestingly, Sound On Sound *ran a feature on the Drake Project back in April 1996, and the Soundbeam was referred to as as being a particularly important tool in allowing people with special needs to participate in musical events.*

Ex-Van Der Graaf Generator multi-instrumentalist (and Soundbeam advocate supreme) David Jackson has recently released a CD which features the Soundbeam extensively. Fractal Bridges *(on Fie! Records, FIE9113, distributed by Vital) is mostly improvised, and features David generating backings with the Soundbeam as well as playing sax and flute. In some cases, the beams are used to trigger MIDI sequences, and a rather interesting effect is generated by allowing two beams to cross, so that Soundbeam plays Soundbeam. The album was produced by Peter Hamill (also of Van Der Graaf, who was featured in* SOS *back in February 1994) and recorded at Terra Incognita — that's the studio owned by David Lord, who we spoke to in November 1996. David Jackson uses the technology live — where he interacts with*

beams while playing — and also gives workshops; he's also the author of a handbook, called Soundbeam Artistry, *available from the Soundbeam Project for £12.95. The Project has also produced two videos:* Welcome to Soundbeam *(£12.95) is a general introduction to the system, with an emphasis on special needs, but offering several examples of dance applications, while* Incidental Music *(£24.95) is a video/booklet package that offers an account of a research project by Dr Phil Ellis at Lambert School, Stratford upon Avon, into the effects of Sound Therapy on a number of children with special educational needs.*

The cost of a Soundbeam system depends very much on how many beams you have — up to four can be used, in a master and three slaves configuration, operating on four different MIDI channels — but a basic single-beam system costs in the vicinity of £1000. Incidentally, a residential weekend is being held on April 25 to 27, at Kinnersley Castle, Herefordshire; delegates will be immersed in workshops, clinics and presentations, with a particular focus for those working in a special-needs environment. Cost, including full board, is £189 plus VAT. For further details on all Soundbeam matters, contact The Soundbeam Project at 463 Earlham Road, Norwich NR4 7HL; telephone 01603 507788, fax 01603 507877, or try them on100637.1043 @compuserve.com if you have email.

You may be interested to know that there is another zero-contact sound-generating instrument available, from the USA. Those of you with an interest in synth history will remember Don Buchla and his alternative modular synth series, not to mention his non-keyboard-based control surfaces. Buchla's latest product is the Lightning II, a combination of infra-red controller

and 32-voice synth. Lightning II senses the position and movement of a pair of hand-held wands and turns this information into MIDI signals. The on-board synth, which uses the Kurzweil Multimedia Audio Sample-playback System (MASS) technology, means you don't even need an external synth to generate sounds: Lightning's sound source offers 16 MIDI channels with dynamic voice allocation, 6Mb wavetable memory with 357 sounds, effects and General MIDI compatibility.

The bulk of Lightning II's electronics is housed in a half-rack cabinet, while a remote head, designed to be mounted on a normal mic stand in front of a performer, contains optics and numeric displays. Based on principles of optical triangulation, Lightning II gathers its information by tracking tiny infra-red transmitters that are built into the wands. These wands, which are battery-powered and wireless, provide complete freedom of movement within a performance space that can be as large as 12 feet high by 20 feet wide.

Basically, Lightning II senses the horizontal and vertical position of each wand, for a total of four independent co-ordinates. From this information, a digital signal processor computes instantaneous velocity and acceleration, and performs detailed analysis of gesture. The user defines the relationships between various gestures and potential musical responses. As an example, Lightning

Crosstalk

II's co-ordinates could be mapped to various MIDI controllers on several channels. Spatial pitch-bend wheels, pan pots, level controls and modulation wheels are easily defined, and performance gestures can be analysed for direction and velocity to generate a variety of notes, as well as other musical events. A multi-dimensional zoning capability can be used to create different musical responses in different regions.

Currently, Lightning II is available only in the States — there is no European distribution — but non-US residents can obtain it mail order. Shipping to "just about anywhere in Europe" is $65, with any local taxes or duty obviously the customer's responsibility. Contact Buchla & Associates, PO Box 10205, Berkeley, CA 94709, USA; telephone 001 510 528 4446, fax 001 510 526 1955, email buchla@best.com or visit Buchla's informative web site at http://www.buchla.com for more info.

Apart from the Soundbeam, the Lightning II, and theremins, there isn't much in the way of off-the-shelf, current choice for no-contact controllers, but we must again mention Infusion Systems of Vancouver, Canada, manufacturers of the I-Cube sensor-to-MIDI system. A total of 18 different kinds of phenomena — heat, motion and proximity, for example — can be used to generate a wide range of MIDI events with this system. Obviously, as with the Soundbeam, you need external MIDI equipment to produce a sound. Infusion Systems can be reached at 1320 East Georgia Street, Vancouver, BC, V5L 2A8 Canada, telephone 001 604 253 0747, fax +1 604 253 0747. You can email them on info@infusionsystems.com or visit their web site at http://www.infusionsystems.com if you're interested in more detail on Infusion products.

Zero-contact — and unusual hands-on — controllers are under development at many educational establishments around the world,

but few are likely to result in mass-market products. Worthy of note is Tod Machover at MIT (Massachusetts Institute of Technology) in the USA. Machover's high profile work involves 'hyperinstruments', interactive instruments designed for anyone to play, with evocative names such as Forest, Gesture Wall, Harmonic Driving, Melody Easel and Rhythm Tree. This technology uses unusual surfaces and sensors to generate data that is processed by a computer and transformed, via MIDI, into the final performance; the hyperinstruments were used last summer in the performance of a work entitled Brain Opera, which premiered at New York's Lincoln Centre. The work will also appear on the Internet, where anyone connected can contribute sounds or influence performances. All the information you could possibly want about Brain Opera is available on a dedicated web site, at http://brainop.media.mit.edu/. This provides a front end for the Internet angle of the project, and has a separate, text-only section if you have a slow net connection or just want to get to the facts quickly. Tod Machover and his hyperinstruments were also recently featured in a week's worth of BBC Radio 3's The Music Machine series; this is a 15-minute weekday slot at 5pm that is aimed at schools and colleges, yet covers an amazingly wide range of musical topics, both traditional and cutting-edge.

One other name worth dropping is Christopher Janney, who creates interactive sound and light installations in the strangest places; his Harmonic Runway for Miami International Airport featured a 180-foot corridor composed of 132 sheets of coloured glass and a sound score based on the natural environments of south Florida. Janney's installations have also appeared in the Boston, New York and Paris subway systems. Not surprisingly, Christopher Janney has a net presence, and the URL you need is http://nttad.com/asci/cjwork.html. ❏

Manual Labour

I have recently obtained a second-hand Fostex A8 quarter-inch 8-track tape recorder. Unfortunately, it came without its original user manual. I've also noticed a connector that's apparently for a remote control. I'd quite like to obtain both of these items, and was wondering if you had any idea where I could look.
Steve Pashley
via email

I've just bought a Fostex X26 cassette-based 4-track, but it's missing the manual. Could you tell me where I might be able to find one?
Frankie Savage
Chingford

Derek Johnson replies: *The manual queries are the easiest to deal with: a word with the service department at SCV London (Fostex's UK distributor) reveals that nicely-bound photocopies of manuals for all older Fostex gear can be purchased for a flat fee of £10 each, inclusive of postage. Contact SCV London, 6-24 Southgate Road, London N1 3JJ; the service department's direct line is 0171 254 3366 if you want to give them a call.*

Tracking down an 8030 remote is going to be a little more tricky. The unit is no longer available as a spare from SCV, so your best bet is going to be a trawl around the dealers to see if there are any available second-hand; alternatively, a 'wanted' plea in our Free Ads section can often scare up the most obscure items. You can find a coupon at the beginning of the Free Ads section at the back of the mag. ❏

Quiet Life

I have a comment to make further to Paul D Lehrman's article on making computer-based studios quieter ('Silent Running', in January 1997's SOS). I use one of those old dot-matrix/daisywheel printer hoods that were around a lot before quieter inkjets and the like. They are about a metre square, usually with a perspex door and a built-in fan (which I replaced with a quieter unit). Most importantly, they are soundproofed. The box also has a flat top for my monitors, and there's even a mains socket and a hole for the leads. I picked this up for a fiver from a computer auction; there were lots there and It's proved to be the ideal solution. I can enclose my computer and all my drives, with no lead extensions needed.
Stephen Bennett
via email

Thanks for the tip, Stephen.

shape of things to come

by Derek Johnson

Roland's fresh Canvas

Elsewhere in this issue, you can read our exclusive preview of Roland's new JP8000 physical modelling-based synth, which marks a move into new technological territory for Roland, even if it's a recreation of analogue synthesis. However, it's nice to see that Roland are not abandoning their established platforms. We've already told you about the imminent JV2080 sound module (see last month's news pages), and now there's a further development of the Sound Canvas General MIDI platform on the way, in the shape of the SC88 Pro module. The original SC88, reviewed in *Sound On Sound*'s September 1994 issue, was then Roland's definitive statement of what a GM module could be, offering a comprehensive waveform collection, huge polyphony, 32-part multitimbrality and two sets of

MIDI connections. The new SC88 Pro takes the original module and soups it up, with a total of 20Mb of waveforms instead of 8Mb. The original offered 654 patches and 24 drum kits, while the Pro pumps that up to 1170 patches, plus 39 drum kits. Another new feature is that, in addition to global effects applied to the overall mix, the SC88 has 64 insert effects, which can be applied to any number of multitimbral parts.

Many of the new sounds have been taken from the SRJV80 sound expansion boards developed by Roland for their up-market synths, and several effects offer RSS (Roland Sound Space) 3D processing. At £799 this isn't the cheapest GM/GS module on the market, but Roland seem to have squeezed quite a lot into a desktop package. Watch out for a review soon.

- **A** Roland UK Ltd, Atlantic Close, Swansea Enterprise Park, Swansea SA7 9FJ.
- **T** Brochure Hotline 01792 515020.
- **F** 01792 575042.
- **W** http://www.roland.co.uk

TL Audio take the tube

TL Audio are adding to their range of valve-based equipment with the launch of the VTC — Valve Technology Console. The VTC is a fully modular in-line multitrack mixer that offers TL Audio's valve circuitry in a compact 8-buss console. The desk is available in configurations from 16 to 56 inputs, with options including internal or external patchbay, fader/mute automation, meter bridge and 8-, 16- or 24-channel expanders. Each VTC channel offers four-band EQ with two fully parametric mids, one stereo and six mono auxiliary sends, faders on both channel and monitor signal paths, insert points and tape/direct out sockets. The master section offers six stereo aux returns, comprehensive control room and studio monitor sections, selectable PFL/Solo-In-Place modes, and two independent headphone feeds. Pricing has yet to be fixed, but it is expected that a 32-channel VTC will retail in the £18,000-£19,000 (including VAT) range.

- **A** TL Audio, Letchworth, Herts SG6 1AN.
- **T** 01462 490600.
- **F** 01462 490700.

FAT sounds from 303 clone

The current vogue for MIDI-controlled clones of Roland's legendary TB303 Bassline shows no sign of abating: the latest machine to attempt to recreate 'that' sound is the Freeform Analog Technologies FB383 Freebass, available exclusively from Turnkey. The FB383 has been designed to be as accurate a simulation as possible, with no extras — just the functionality of a TB303. The market apparently doesn't want any more or less than the control offered by a TB303, and it doesn't want to have to pay a lot for it. The result is that the bright yellow Freebass will retail for an extremely attractive £199, including VAT. The FAT Freebass is totally analogue, features a VCO that is continuously variable between square and sawtooth waveforms (a feature *not* available on original TB303s!), and offers controls for filter cutoff, resonance, envelope mode, decay and accent (the latter two are triggerable via MIDI). The module has a dedicated auto-tune facility and an audio input to the filter.

- **A** Turnkey Professional, 114-116 Charing Cross Road, London WC2H 0DT.
- **T** 0171 240 4036.
- **F** 0171 497 0690.
- **E** sales@turnkeypro.compulink.co.uk
- **W** http://www.turnkey.uk.com

Max sound, min pound

New from BBE is what appears to be the most accessible incarnation yet of the Sonic Maximizer psychoacoustic enhancer technology: the two-channel 262 is not only hyper-compact, in a 1/3-rack package, but costs a mere £159, including VAT. Controls are kept to a minimum: each channel features a Lo Contour and Process knob, plus a bypass switch. The 262 has been designed for stage or studio use.

BBE have also launched the DI-100, a direct injection box with built-in Sonic Maximizer; once again, controls are restricted to Lo Contour and Process, but this £120 device allows you to add a little extra bite and fullness to any single instrument or mic input. BBE also make a Sonic Maximizer-less DI box — the DI-10 — for £99.

A Sound Technology plc, Letchworth Point, Letchworth, Herts SG6 1ND.
T 01462 480000.
F 01462 480800.
W http://www.soundtech.co.uk

Quasimidi rave on

Quasimidi just keep on releasing dance machines; the new Rave-O-Lution 309 is a 'two-in-one' rhythm unit that combines a polyphonic drum module with a monophonic bass/lead line synth. The package also features an integrated sequencer with real-time and step record modes. The drum section is divided into four parts — kick, snare, hi-hat and percussion — and each part can be edited with a full complement of parameters, identical to those of the synthesizer section. Each section also has its own collection of 64 patch memories. No less than 128 instrument sounds are available for the bass/lead track, and there's a real-time filter, EQ, LFO and effect controls.

The 309's front panel features a dizzying number of knobs and buttons, all of which transmit over MIDI. The feel is like a mutated TR909 or TR808 welded to a hyped-up TB303 — rather like a Roland MC303, but with even more knobs. The sequencer section offers accent and glide controls, like the TB303, and there are three effects processors on board (offering reverb/delay, modulation and EQ). On the filter front, both the synth and percussion sections benefit from a 24dB/octave filter with cutoff and resonance control, as well as an adjustable 'filter overdrive' control for producing "delightfully nasty, aggressive sounds." Oo-er, missus...

Quasimidi have made expandability a feature of many of their previous products, and this is also the case for the 309: first of all, the single stereo output pair can be enhanced by a further pair of outs and two audio ins. The outputs are freely assignable and the inputs allow you to process an external sound source through the internal synth parameters. Also available later in 1997 will be user-installable sound and waveform expansions.

A Quasimidi Musikelektronik GmbH, Eisenbahnstr. 13, D-35274 Kirchhain, Germany.
T +49 6422 94020.
F +49 6422 940244.
W http://www.quasimidi.com

Opcode's platform news

Opcode have recently released the Studio 64X, a cross-platform MIDI interface/patchbay that's equally at home in a Macintosh or Windows environment. The device automatically detects whether it's attached to a Mac or PC, and can function as a stand-alone MIDI patchbay. The Studio 64X offers four MIDI ins and six MIDI outs, for a total of 64 discrete MIDI channels; all formats of SMPTE sync are also supported. MIDI patchbay functions can be programmed in software, stored in the interface, and then accessed from the front panel; eight programs — four user and four preset — are stored. Also handy is the modem/printer through switch, which gives you access to these peripherals, if attached, without cable swapping. The Studio 64X retails for £269, including VAT.

A MCMXCIX, 9 Hatton Street, London NW8 8PL.
T 0171 723 7221.
F 0171 723 8150.
W http://www.mcm.co.uk/opcode
W http://www.opcode.com

Why did Sutekina cross the road?

Sutekina Music, in London's Denmark Street, has recently moved — across the road. The new custom-built premises at number 10 mark the company's relaunch as Sutekina Pro Audio, a change which reflects its newly-expanded range of state-of-the-art recording equipment and digital technology. In addition to the showroom space at the front of the store, which will feature both keyboards and software, the rear of the new premises has been designated the Digital Domain. Here, customers will find a large recording studio-style area and an all-digital suite, where the latest technology can be seen and heard. The idea behind the Digital Domain is to provide working examples of how audio can stay digital during the entire recording process, bar producing the initial sound and hearing the result over the studio monitors. Sutekina Pro Audio will also offer access to Time & Space and Sounds OK sample CD jukeboxes, and provide a mail-order service.

A Sutekina Pro Audio, 10 Denmark Street, London WC2H 8NA.
T 0171 836 0180.
T Mail Order 0800 371129.
F 0171 379 3398.

shape of things to come

Vestax rev up with V8

Vestax have been a little quiet since the launch of their HDR range of hard disk recording systems, back in 1995. However, they're now back in launch mode, with the announcement of the HDR-V8 Pro Series digital recording system. This is an 8-track recorder in a 2U rack space, with a main processor that's three times faster than previous versions, eight ins and eight outs, simultaneous recording on all eight tracks, and no data compression. As shown at November's AES show in LA, the HDR-V8 was equipped with a 1Gb drive as standard, which can be user-upgraded to 4Gb. The new machine is further enhanced by balanced ins and outs, three auxiliaries per channel and a built-in digital mixer with three-band EQ. As for compatibility with other digital systems, the HDR-V8 can interface with Tascam DA88 and Alesis ADAT. Removable hard drives can be installed, if you like, and other options include remote control, a SMPTE interface, and multiple machine lock.

A Vestax Europe, 18 St Christopher's Road, Haslemere, Surrey GU27 1DQ.
T 01428 653117.
F 01428 661021.

The BM6A monitors.

Dynaudio active

Dynaudio Acoustics have added two pairs of two-way active monitors to their range. The monitors — the BM6A and the BM15A — were previewed at the Los Angeles AES show back in November. The smaller BM6A (£1323.05 per pair, including VAT) features 170mm bass and 26mm HF drivers, and is equipped with a pair of 100W MOSFET amps, with HF clip protection and overload indicators. The BM15A (£1957.55 per pair) features a 240mm bass driver, and a 30mm HF driver; its dual 150W power amps are configured as 150W into 4Ω for the LF driver and 80W into 8Ω for the HF driver. Both monitors have a quoted frequency response of 40Hz-20kHz, and both also offer variable LF EQ and HF trim controls.

A Unity Audio Ltd, Upper Wheeler House, Colliers End, Herts SG11 1ET.
T 01920 822890.
F 01920 822892.
E unityaudio@channel.co.uk

Soundcraft's Phantom power

Soundcraft's support for their Ghost and Ghost LE mixers continues with the launch of a new 24-channel expander module. The expander has an identical channel strip layout to the original consoles, with the same low-noise mic preamp and 4-band parametric EQ. When the expander is added to a Ghost desk (with CPU), the mutes and MIDI note functions are integrated with, and controlled from, the main console's master section. Adding the expander to a 32-channel console gives the user 120 separately-controllable inputs at mixdown; installation simply involves a D-Type ribbon cable.

Incidentally, there's a chance to win a superb 24-input Ghost, including meter bridge and stand, in this issue and the next of *Sound On Sound*; turn to page 256 to try your luck.

A Soundcraft Electronics Ltd, Cranborne House, Cranborne Industrial Estate, Cranborne Road, Potters Bar, Herts, EN6 3JN.
T 01707 665000.
F 01707 660482.
W http://www.soundcraft.com

Jazz saxophonist Courtney Pine (see *SOS* feature in September 1996) is now the proud owner of a Digidesign Pro Tools II Core System, Yamaha 02R digital mixer, Focusrite Red 2 dual parametric EQ, Focusrite Red 3 dual compressor/limiter, and Lexicon PCM80 multi-effects, plus a variety of digital interconnect tools. The system was supplied by pro audio retailer TSC.
T TSC 0171 258 3454.

Mixer manufacturers Allen & Heath, based in Penryn, Cornwall, have been made South region winner of the NatWest/FT Export Excellence Awards. MD Glenn Rogers comments: "This is a great achievement for everyone in the company. Exports now account for 94% of sales revenue, thanks to the hard work of our Penryn team and our distributors across the globe." Allen & Heath went forward with the five other regional winners to the national award ceremony, held on November 20 in London.
T 01326 372070.

Rane's UK distributors Shuttlesound are providing an in-store demo rack (installed in authorised Rane outlets) for the new Mojo range of signal processors. Each contains a MC22 compressor, MQ302 graphic EQ, MX22 and MX23 Mojo Dividers, and MH4 headphone amp.
T 0181 646 7114.
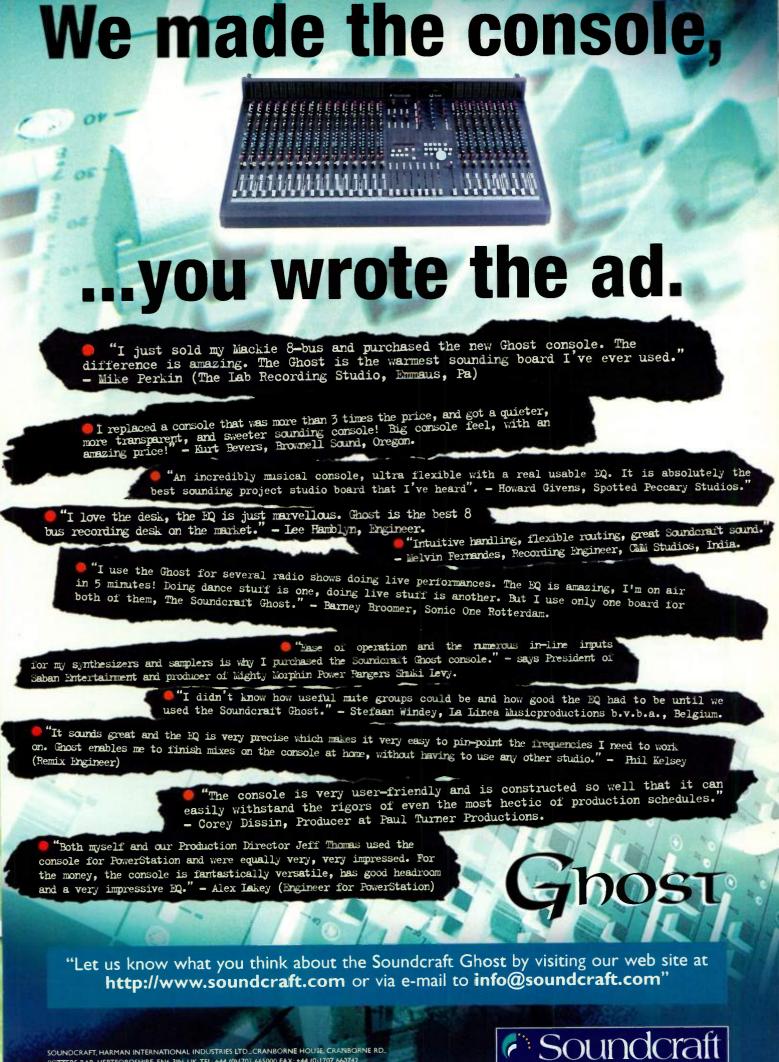
The Arts Council's recent lottery hand-outs include a £9,500,000 award for a National Centre for Pop Music, to be based in Sheffield. The centre will be an art-led, technology-based facility with a mix of changing exhibits, interactive displays, and a cafe and bar.

Digidesign Pro Tools users looking to upgrade to version 4.0 will be interested in the full rules governing free upgrades: registered users of Pro Tools III and Pro Tools Project who purchased their systems (PCI or NuBus) after June 1st 1996 will automatically receive a free update to Pro Tools v4.0. Registered users of Pro Tools with *PowerMix* software who purchased their software after October 1st 1996 will automatically receive a free update to Pro Tools v4.0. All registered users who purchased *Sound Designer II* v2.82, who are registered for Pro Tools v3.2/3.21 software will automatically receive the Pro Tools 4.0 upgrade for free. This offer does not apply to customers who have purchased the *Sound Designer II* v2.82 update/upgrade.
T 01753 653322.
F 01753 654999.

shape of things to come

K Klass go underground

D ance artists and remixers K Klass have recently moved into their new studio — which is housed in a nuclear bunker in North Wales, formerly run by the Royal Observation Corps! The studio is equipped with gear supplied by Larking Audio, centred around a Soundtracs Jade S 48-channel in-line desk with moving fader automation, assignable dynamics and surround sound capability, plus a Dynaudio M3 monitoring system. The large rooms and thick concrete walls make the bunker an ideal location for a studio (as well as offering plenty of room for expansion), and the survival systems — air filtration, contamination tanks and radio gauges — remain in place! As of writing, K Klass are working on a remix of Whitney Houston's new single, as well as material for their next album for Deconstruction.

Another studio to get the Larking Audio touch is that owned by original Boney M singer Liz Mitchell and her husband Thomas Pemberton, who is also her manager and producer. The newly-built Oxfordshire studio houses a Soundtracs Jade 40-input desk, Saturn 824 analogue 24-track, 24 channels of Dolby SR noise reduction, Genelec 1037B powered monitors and a range of outboard equipment. Liz is currently working on her career as a solo artist.

A Larking Audio Ltd, 14-15 Cam Square, Hitchin, Herts SG4 0TZ.
T 01234 772244.
F 01234 772109.
E 100347.1046@compuserve.com

caught in the net

QSOUND *IQ*
QSound Labs' *IQ* is a multi-purpose 3D audio enhancement software designed for the Internet. The software is compatible with audio plug-ins such as *Real Audio*, *Shockwave Audio* and *Xing StreamWorks* for both Netscape *Navigator* and Microsoft *Explorer* web browsers. Using Qsound's *QXpander* stereo-to-surround technology, it provides real-time 3D enhancement; *IQ* also automatically detects mono content and applies mono-to-3D processing. The software also works with any Windows audio playback, enhancing the sound of multimedia titles, WAV files and even system sounds. *IQ* costs US$14.95, and is available from QSound's web site; at present, *IQ* is a PC-only product, with a Mac version currently under development.

W http://www.qsound.ca

DANCE MUSIC RESOURCE PAGES
The Dance Music Resource Pages, the UK dance release resource which we plugged back in November, has added a free new release email service. If you've got a modem and an email account but can't access the broader Internet, this could be the answer to getting the latest release information. To subscribe to the list, send a request to the email address below. The web site is still going strong, and the release listing is still available as a fax service or newsletter.

E juno@deepside.demon.co.uk
W http://www.ndirect.co.uk/dance/music/htm/

ARX SYSTEMS
Australian-based audio manufacturer ARX Systems are now on-line; the site features full product information, user manuals, application notes, reviews, email contacts for various ARX personnel plus general audio information. The company claim to have optimised the site for people with limited-bandwidth access, so visitors won't have to spend time waiting for "a slew of meaningless graphics to download".

W http://www.arx.com.au

HEAVENLY MUSIC
MIDI File company Heavenly Music have updated their web site; in addition to all the latest Heavenly news, downloadable catalogue and free demo files, the site now features a new educational section, with background music composed and produced (using Steinberg's *Cubase*) by 12 year-old Paul, son of Heavenly's Pauly and Joe Ortiz. Cute logo too!

W http://www.ortiz.demon.co.uk

APHEX SYSTEMS
Aphex Systems — we talked to the company's president Marvin Caesar last month — have recently launched their web site. Visitors to the site can gain access to a company profile, marketing literature, product reviews and technical support, plus details of international distributors.

W http://www.aphexsys.com

SOS can be reached on CompuServe; our email address is 100517,1113. Our full Internet address is 100517.1113@compuserve.com.

shape of things to come

DAT'S new!

The new Fostex D15 timecode-chasing DAT machine has been announced by UK distributors SCV London. The D15 replaces the established D10, which brought the ability to read and write SMPTE-encoded DAT to a wider range of people, and pioneered 'instant start' and 'scrub' on DAT by using on-board RAM. The new machine is enhanced by a number of novel features, including a new 4-motor transport (which can rewind a 120-minute tape in as little as 60 seconds), a parallel interface as standard, selectable reference levels, and fine recording-level adjustment for each channel. New timecode facilities, which will be supplied as standard on all machines, allow the user to stripe tape with externally-generated timecode and output LTC or A-time, reference externally to video or word clock and chase to incoming timecode.

The Fostex D15 will retail at just £2931.63, including VAT, complete with all the necessary timecode facilities; an optional RS422 protocol interface for external controllers should be available later in 1997 for approximately £230.

A SCV London, 6-24 Southgate Road, London N1 3JJ.
T 0171 923 1892.
F 0171 241 3644.
E sales@scvlondon.co.uk

Crimewatch SOS

Police are looking for help in identifying and tracing a man they want to interview in connection with a series of deceptions which have resulted in the loss of £35,000 of recording equipment. The suspect used stolen cheques to buy equipment from Active Sound UK in Watford, Music Control in Stoke-on-Trent, Universal Computer Systems in Perivale and Blue Systems in Saffron Waldon. Some equipment was delivered to an office that was leased under a false name in Kilburn High Road, NW6, and some he collected in person. When police were alerted and visited the office, it had been cleaned out. Luckily, one of the collections was made at Kings Cross station, and was captured on CCTV cameras; the suspect is described as a white or latin-looking man, aged between his late 20s and mid 30s, about 5'9" or 5'10" tall, medium build, with long, clean, fairly straight black hair, worn in a long 'Beatles' style. He had a distinctive twitch in his right eye, spoke with a London accent and used the aliases Richard Johnson, John Goodhall and Mark C McDermott. Police believe he may have been working with a male accomplice, but don't have a description of him.

Anyone who has information which could assist police with this enquiry should telephone Hampstead Police Station and speak to DC Louise Cherrington on 0181 733 6664 or Crimestoppers on 0800 555111.

The Wonder of it all

At last November's AES show in LA, Stevie Wonder stopped by the Emu stand to check out the new Emulator 4 keyboard. After a few return trips to the booth, Stevie became so impressed with the E4K that he asked to take it home. Dave Bristow, Emu's vice president of marketing, was pleased to grant this wish, and presented the E4K on behalf of Emu Systems.

A Emu Systems, Suite 6, Adam Ferguson House, Eskmills Park, Musselburgh, Scotland EH21 7PQ.
T 0131 653 6556.
F 0131 665 0473.
W http://www.emu.com

shape of things to come

Croydon's Village fate

The Music Village Group of music retailers is opening a second Digital Village shop in Croydon. The new shop, which is being opened in response to the continuing success of the original Barnet branch, should be fully on-line early in February. The ink has literally just dried on the lease as we go to press, so full details aren't yet available. We do know, however, that the new shop will be located at 562/562A Brighton Road in South Croydon (that's opposite the *Croydon Advertiser* building for those of you in the area). We also know that you'll be able to try and buy the latest hi-tech and pro audio gear in comfortable surroundings. For phone details and precise opening date, keep an eye on the Music Village and Digital Village ads in *Sound On Sound*.

Yamaha's floored genius

Yamaha's new MFC10 may resemble a large guitar effects board, but in fact it's a MIDI foot controller, offering 12 sturdy foot switches and a pedal that can be configured to transmit a wide range of MIDI data in real time. If all you want to do is send out program change or MIDI Stop/Start commands, the MFC10 will cope, but if you're after something a bit more complicated, the new pedal board is well up to it. Note on, Note Off, tempo, Bank Select and sundry control changes can all be transmitted from the MFC10 — and if one foot controller isn't enough,

there are sockets for you to add up to four more external pedals, which can be either continuous controller, latching or trigger-type pedals. The rear panel of the MFC10 also features a WX input, which allows Yamaha's WX11 wind controller to be used to control an entire MIDI system; one other advantage of this setup is that the WX socket removes the need for the BT7 interface/battery pack normally required with WX instruments. The MFC10 has a recommended retail price of £239 including VAT.

> **A** Yamaha Kemble Music (UK), Sherbourne Drive, Tilbrook, Milton Keynes MK7 8BL.
> **T** Information line 01908 369269.
> **F** 01908 368872.
> **W** http://www.yamaha.co.uk/

education corner

ROLAND'S EXTENDED WARRANTY FOR SCHOOLS

Roland have recently introduced a comprehensive two-year warranty scheme (on both parts and labour) for their products purchased by schools and colleges. To qualify, educationalists must return their warranty card directly to Roland, on purchase. To discuss your requirements, you can call Roland's dedicated Education Helpline, on 0941 121300.

Islington Music Workshop Studio

IMW CUT PRICES

Islington Music Workshop have found their courses so popular since the installation of an SSL 4000G+ mixing desk that there are plans to increase the number of courses in 1997. The immediate result of the extra courses is a new pricing policy: course costs will actually be cut by up to 12%.

> **A** Islington Music Workshop, 44 Peartree Street, London EC1V 3SB.
> **T** 0171 608 0231.
> **F** 0171 490 0120.

CITY AND ISLINGTON COLLEGE

City and Islington College recently opened new music facilities, including digital and analogue recording studios, a MIDI suite and four rehearsal rooms. The college currently runs two full-time courses in popular music (BTEC National Diploma and Access to Music for Adults) as well as a range of part-time recording and music technology courses.

> **A** City and Islington College, The Marlborough Building, 383 Holloway Road, London N7 0RN.
> **T** 0171 607 8614.
> **F** 0171 607 6995.

THAMES VALLEY UNIVERSITY

Thames Valley University's range of short courses — called Mini Modules — have been running for three years now. In amongst the 'German for Beginners' and 'Indian Head Massage' courses is a range of music technology courses. These typically run for 20 hours over 10 weeks, and include Sound Sampling for Beginners, MIDI and Sequencing for Beginners, Creative Radio, Radio for DJs, Multimedia, and An Introduction to Music Studio Techniques. Each costs £67.50; concessions are available. Contact the University for details of the next set of courses.

> **A** University Centre for Complementary Learning, Thames Valley University, Slough SL1 1YG.
> **T** 01753 697719.

CITY OF WESTMINSTER COLLEGE

A brand new Soundcraft DC2020 Surround mixer has been installed at The City of Westminster College's Cockpit Theatre. The desk will be used by students on the college's sound engineering courses, for both live recording and post-production projects.

> **T** Soundcraft 01707 665000.
> **A** City of Westminster College, 25 Paddington Green, London W2 1NB.
> **T** 0171 258 2789.

SURREY UNIVERSITY

The music department at Surrey University has recently purchased a DDA CS3 mixing desk for use on its Tonmeister music and sound course and for its ongoing research programme. The desk, which features a multitude of output possibilities and DDA's SIS (Spatial Image System), will be used to make multi-channel surround recordings, and is portable enough for use on mobile recording sessions.

> **T** Surrey University 01483 259317.
> **T** Mark IV Pro Audio 01562 741515.

shape of things to come

Merging technologies

If your MIDI merging needs aren't addressed by commonly available 2:1 devices, check out two new products from Philip Rees. The 5M and 9M offer users the ability to reliably merge five and nine MIDI streams (respectively) into one. While this might seem extreme, merging the outputs from a host of controllers in order to access one multitimbral module would be rather useful for, say, a band of MIDI musicians — a percussion controller, a couple of keyboards, a MIDI guitar and a MIDI horn soon adds up to a lot of MIDI outputs that can be easily managed by a 5M.

Both devices have a single MIDI Out port, which carries the merged data stream, and both require an external power source. The sophisticated data-handling features of these devices include intelligent interlocks on MIDI clock and pitch bend, and both units can readily handle all types of MIDI data, including MIDI Timecode and System Exclusive. The suggested retail price of the 5M is £125.95, while the 9M costs £169.95.

A Philip Rees Music Technology,
Unit 2, Clarendon Court,
Park Street, Charlbury,
Oxford OX7 3PT.
T 01608 811215.
F 01608 811227.

Peak tweak

Bias *Peak*, the digital audio editing software for Apple Mac which we reviewed in our September 1996 issue, has now reached v1.5. The new version features RealAudio file format support, used for preparing audio for the Internet, and support for *Sound Designer II* audio regions. SMDI sampler users will benefit from improved SMDI support. In addition to sending and receiving individual samples, *Peak* now sends and receives multiple samples at once; the software now supports keyrange information on Emu samplers too. Also improved are the pitch change and change duration functions, and the mono-to-stereo transformation facility.

A Natural Audio Ltd, Suite 6,
Kinetic Business Centre,
Theobald Street,
Borehamwood WD6 4SE.
T 0181 207 1717.
F 0181 207 2727.
E 100565.3561
@compuserve.com

LEXICON PRICE/QUALITY BREAKTHROUGH

lexicon

MPX 1

For those of us who need top quality reverb & multi FX but cannot stretch to the expense of the PCM80 & PCM90, Lexicon have released the MPX-1. Featuring the famous 'Lex' chip for reverb and separate DSP processor for multi FX, the MPX-1 brings you all that is good in signal processing for an incredible £1199 inc VAT!

- **Superb Quality Reverb**
- **Up To 5 Effects At Once**
- **Midi Controllable Morphing of Effect**
- **On-Line Help Built-In**
- **SPDIF digital i/o**

Up to 5 simultaneous effects are available including pristine quality stereo pitch shifting, <u>and</u> effects can be "morphed" from one algorithm to another, as pioneered in the Vortex. An all new operating system which includes on-line help and a databasing system for sorting presets, means that achieving quality results is as stress free as possible.

Initial supplies of this exciting new product will be very limited - get your order in now to avoid disappointment.

NEW PRODUCT
£1199
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turnkey
PROFESSIONAL
Tel 0171 240 4036

1066   dbx

DBX build on the success of their highly acclaimed dynamics processing range with the flexible new 1066.

In addition to all the professional features expected from this respected brand, the 1066 offers switching between their patented "Overeasy" soft-knee and hard-knee modes, sidechain switching, ultra-low distortion "Peak Stop Plus" limiting and meticulous US hand-built quality.

A/B this product at Turnkey to hear why DBX remains an industry standard in signal processing.

NEW PRODUCT
£479
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turnkey
PROFESSIONAL
Tel 0171 240 4036

NEW
FILTER   waldorf

The 4 Pole is the legendary Microwave/Wave 24 dB lowpass filter. You are free to apply this filter to any audio signal that you patch into the Filter. So if your favourite sampler or sampleplayer lacks that special quality, you can give it a sonic boost via the 4 Pole Filter.
It is the most powerful (and quite possibly the only) analog filter available today!

NEW PRODUCT
£299
NEW BOXED

VTP-1   DigiTech

Digitech have already taken the pro market by storm with the introduction of the Studio Vocalist, and now the 2 channel VTP1 tube mic amp / EQ looks set to do the same. It combines four bands of semi parametric EQ with a high definition mic amp based around a proprietary valve design. The usual phase reverse, high pass filter, 20dB pad (line inputs also available) and 48v phantom power are provided, together with high quality balanced analog outputs, and AES/EBU and SPDIF digital outputs at either 44.1 or 48 kHz. The ideal front end to any digital recording system.

NEW
£999
NEW BOXED
turnkey
PROFESSIONAL
Tel 0171 240 4036

NEW
Indigo Range   TL Audio

In the production of modern valve audio equipment, the success of TLA equipment has been unrivalled, from the best selling compressor, to the full blown valve mixer. For those of you who don't quite need the full functionality of the existing range, the new 1U high Indigo range has been designed specifically for your needs. Superb warm and fat valve signal processors at an affordable price. Models include a compressor, EQ, mic amp and overdrive unit. Entire range on demo at Turnkey for direct A/B comparison.

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BBE

- **Full Range Process Covering High & Low Frequencies**
- **Large Range of Models for all Applications**
- **Suitable for Studio & Live Use**
- **Killer Sound!**

MAXIMISE YOUR MIX!

No mix is complete without some sort of psychoacoustic processing these days, and the BBE sonic maximizers have a unique angle in this department, with a process quite unlike any other unit. Rather than using harmonic regeneration, the audio spectrum is split into various regions and phase shifted apart, which can have devastating results - particularly at the bottom end. Unlike harmonic regeneration, phase shifting doesn't generate feedback, so they're also great for live use. Check out the full range below - there's sure to be a model to suit you!

362 Sonic Maximizer

The 362 series delivers the sound-improving benefits of "Sonic Maximizer" technology with the convenience of ganged-stereo operation.

362 £199 NEW BOXED

362NR Sonic Maximizer + Noise Reduction

The 362NR combines a full-featured sonic maximizer with an independent, single-ended noise reduction system in a single rack-space unit.

362NR £269 NEW BOXED

362SW Sonic Maximizer + Sub Woofer Control

The 362SW combines a full-featured sonic maximizer and an independent subwoofer control with a variable low-pass filter and output level control.

362SW £269 NEW BOXED

462 Sonic Maximizer

The 462 is perfect for semi-pro applications not requiring balanced jacks. With true dual-mono operation, the 462 is two independent sonic maximizers in one chassis.

462 £269 NEW BOXED

862 Sonic Maximizer

The 862 unit features dual-mono operation with balanced XLR inputs and outputs. It is designed for applications requiring low noise, high headroom and +4dBu input levels.

862 £469 NEW BOXED

HR-GP5
MULTI FX PROCESSOR — SONY

Acclaimed quality multi-effects unit from Sony, brother of the HR-MP5. The same processing split further allows up to 7 effects at once with a fantastic range of reverbs, delays and more "off the wall" effects, plus a superb pre-amp section making it ideal for both studio use and guitarists. It features the same display and operating system as the MP5, so nothing could be easier to use! With our massive discount stocks won't last long, get your order in now!

- **Top Quality Effects Processor**
- **Exclusive End of Line Clearance Deal**

RRP £686 **£249** NEW BOXED

DMT-8 VL
DIGITAL MULTITRACKER — FOSTEX

The personal digital multi-track is here at last, and at a price you won't believe! 8 separate tracks of CD quality audio recorded to half-gig hard-disk (up to 12 minutes). Record 2 tracks simultaneously, jog-shuttle audio and sync to MIDI without losing a track. The 8 channel mixer accepts 2 mics, with a 2 band EQ and 2 aux sends. Separate outputs and S/PDIF optical output. Why buy 4 tracks when you can have 8 ?!

RRP £1499 **£999** NEW BOXED

IT'S BACK!

Vortex — lexicon

Despite the first digital effects units being developed almost 20 years ago, they have all followed a familiar path ever since, offering a conventional series of effects. The Lexicon Vortex breaks the mould in allowing effects to be morphed between in real time and in the process, creating new and previously unheard of effects. (This actually changes the effect algorithm continuously - it is not a simple cross-fade between effects) This unit is ideal for dance music, and anybody requiring out of the ordinary effects, and its real time nature make it an excellent choice for live performers such as guitarists. Effects are also affected by the input signal, so that dynamics can control various different paramters within an effect algorithm. Tempo can be tapped in to control delay times etc...Typical Lexicon quality can of course be taken for granted. Exclusive bulk purchase makes this huge reduction possible on the end of the line. First come first served!

- **Unique morphing of effects**
- **Typical Lexicon quality**
- **Superb realtime control**

RRP £429 **£199** EXCLUSIVE DEAL

MixPad 9 — SAMSON AUDIO

A rugged compact mixer packed with professional features including 2 mic/line and 3 stereo inputs, 2 auxs, 2 stereo returns, 2 band EQ, phantom powering and balance outputs.

With it's superior low noise performance and high 27dBu headroom, this is the perfect mixer for small digital set-ups or high quality submixing in larger systems.

With this Turnkey megadeal no other mixer offers these features at this price. Limited stocks only.

RRP £169 **£129** NEW BOXED

Roland JP8000
Analogue Modelling Synthesizer

Though Korg have been making some inroads lately, notably with the Prophecy, Roland is still the name to drop in dance circles, so the announcement of a new, analogue-style Roland megasynth stirs the blood like little else. SOS's BRETT MITCHELL gets his exclusive hands on it...

The demand for more performance-orientated synths is seemingly showing no signs of waning. In recent years we've seen just about every major and not-so-major manufacturer implement some form of increased real-time control on their new instruments. Ironically,

an analogue sound, and an analogue type of control. What they also did was to take existing controllers, such as ribbons and knobs, and greatly expand on their traditional uses by taking advantage of the new high-speed DSP chips that powered their synth engines. The result was, and is, sound manipulation and real-time control trickery that could only be dreamed about with traditional analogue synths.

This is where Roland's new JP8000 comes in: it, too, features a form of modelling synthesis, a multitude of front-panel knobs and sliders, and some real-time control innovations we've never seen before. It's a true synth, and in my opinion the finest-sounding machine Roland have made for quite some time.

ON THE OUTSIDE

The JP8000 is finished in a spacey, metallic-blue plastic casing with a rather futuristic look. The blue surface has a velvety kind of feel which is quite appealing, though I did notice over the course of the review that this velvety surface is also very receptive

backlit LCD, and much orange and white text surrounding all the above. This initially makes the JP8000 appear overwhelming, and perhaps a bit confusing — like the first time you saw a Jupiter 8, perhaps? Once you become familiar with the layout, it all makes sense and is, in practice, quite intuitive and easy to use. To the left of the keyboard are the octave up and down buttons, traditional Roland-style Bender/Mod wheel, LFO 2 functions, and various controller assignment buttons. I've never really been very enthusiastic about the integrated Roland Bender/Mod wheel, especially the Mod part of it, which has a very short range and is difficult to control. Roland have more than made up for it, however, by implementing a very nice ribbon controller which, as we shall see in a moment, has been improved by the fact that you can assign any number of real-time controls to be governed by it.

At the rear is a standard, detachable mains connector, power on/off switch, quarter-inch left and right audio outs, hold (sustain) and control pedal inputs, and

most of these new real-time controllers have been existing control sources from years gone by, such as ribbon controllers, knobs and sliders and, to some degree, even breath controllers. This demand for more real-time control, combined with an undying love for all things analogue-sounding, has spawned some very interesting new instruments, such as Clavia's Nord Lead and the Korg Prophecy. These instruments set out to give the people what they wanted —

to the inevitable barrage of finger scum — so I'm not sure how it will look after years of use. Its overall weight, which is just under 8kg, is only around a third of the weight of the original (Jupiter) JP8, which should make it popular with gigging musicians, quite apart from its performance-orientated features. Above the 49-note, velocity-sensitive keyboard is a crowd of knobs and sliders dedicated to controlling sound parameters, a small (2-line x16-character)

MIDI In and Out connectors — but curiously for a synth of this price, there's no MIDI Thru.

SYNTHESIS ENGINE

The JP8000 uses a new form of synthesis called Analogue Modelling, which seems to be based on a similar concept to a certain red Scandinavian synth released a couple of years ago. Information on the finer details of Analogue Modelling are a little thin on ▶

At TDK, we believe that repetition puts an end to all art. That's why we know you can only accept the best for recording, editing, mastering, presenting and archiving your performance. For this reason, we developed TDK STUDIO products, a range of digital audio media providing flawless recording characteristics and uncompromising reliability. From now on, there will be no repetition due to technical failure of recording media, so you can concentrate on your art first time, every time.

TDK CD-R STUDIO

Probably the first CD-R to overcome compatibility problems that can occur when switching between recorders or players. With its proprietary dye recording layer, TDK's CD-R STUDIO ensures absolute accurate pit forming and an extremely low error rate with all 1 x and 2 x drives. Quality 100% guaranteed. TDK's CD-R STUDIO: Playing length 74 minutes.

TDK MD STUDIO

Looking for a compact and 100% reliable medium for jingles and masters? Then go on air with TDK's MD STUDIO. Developed not only for professional broadcast applications but also for home use, MD STUDIO combines an extremely low error rate with an especially robust cartridge design. Quality 100% guaranteed. TDK's MD STUDIO: Playing length 74 minutes.

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Internet: http://www.tdk-europe.com

⚙TDK

Roland JP8000

▶ the ground: all we're told is that it "processes all the basic waveforms through the built-in DSP oscillator calculations". Limited polyphony is a problem that seems to plague a lot of the newer modelling-based instruments but, thankfully, the JP8000 can generate its sounds with a respectable eight notes of polyphony.

The synth's main operational mode is Performance mode, and it comes equipped with 64 preset and 64 user Performances. Within each of these Performances are two Parts, Upper and Lower, which can be configured in either Single, Dual or Split keyboard modes. Each of the Parts is then assigned to a Patch, of which there are 128 user and 128 preset. Each Patch, in turn, has a conventional 2-oscillator, analogue synth-style construction (Oscillator-Filter-Amplifier).

Oscillator 1 offers a choice of seven basic waveforms (Saw, Square, Triangle, Noise, Feedback Osc, Super Saw, Triangle Mod). The first four of the waves are quite conventional and will be familar to most readers, but the others are a little different: Super Saw consists of seven sawtooth waveforms detuned against each other, Triangle Mod is a modification of a triangle wave with a large number of overtones, and Feedback Osc is said to be geared towards creating guitar feedback sounds — though in practice I found it useful for quite a wide variety of timbres. Each of these waves then offers two additional controls, which differ for each waveform. So, for example, the Square wave has adjustable pulse width and pulse width modulation depth, while the Feedback Osc has adjustable harmonics and feedback amount.

Oscillator 2 is a much simpler affair, offering just Saw, Triangle and Square waves, also with adjustable pulse width and pulse width modulation. Osc 2 may be tuned over a four-octave range, as well as fine tuned, and Osc 2 can be synced to Osc 1.

From here we move to the Osc Common section, where you can set the relative levels of the oscillators against each other, and apply ring modulation and variable amounts of cross modulation, as well as pitch env and LFO to pitch controls. It is possible to apply the pitch envelope and LFO 1 simultaneously to Osc 1 and 2, Osc 2 alone, or to cross-modulation.

It would be an understatement to say that the sonic possibilities of the oscillator sections alone are comprehensive. The combination of all the Osc 1 waves, combined with oscillator sync, cross modulation and ring modulation should keep you in new sounds for quite some time, but — perhaps most importantly — it's also going to generate sounds the like of which you've never heard before.

Moving onto the Filter section, this has 12dB or 24dB/octave high-pass, low-pass or band-pass filters, all with adjustable resonance. The filter has a four-stage ADSR (Attack, Decay, Sustain, Release) envelope, in addition to key follow, LFO1 depth and env amount. I've no complaints about the filters, which work a treat in all their modes, and even simulate filter oscillation at extreme settings.

The Amp section is an (also typical) ADSR affair, with an overall level control and pan

"...this is one very accomplished synth engine..."

pot; autopanning is possible via LFO 1. The JP8000 has two LFOs: LFO 1 offers Saw, Triangle, Square and Random waves with adjustable rate and fade time, and largely takes care of all the internal modualtion duties. LFO 2 is directly linked to the mod lever and can be assigned to Pitch, Filter or Amp, with adjustable rate and depth.

Finally, just before we get to hear the sound emerge from the outputs, there's a simple bass and treble control, chorus effect and delay. The chorus is controlled from a basic level pot, with a choice of 11 effect variations: fast and slow choruses, flanges and phasers. The delay section has three pots dedicated to controlling delay time (maximum 1250ms), feedback and level, in real time, and there's a choice of three panning delays plus short and long mono delays. The various chorus and delay effects are selected by way of the LCD, though,

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BRIEF SPEC

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- **Sound Generation Method: Analogue Modelling.**
- **Sound Source: 2 oscillators (max), sub-oscillator, noise generator, 8-note polyphonic.**
- **Keyboard: 49 notes, velocity sensitive.**
- **Effects: 2 (11 chorus/flanger variations plus 5 delays).**
- **Programs: 128 Performances, 256 Patches.**
- **Controllers: Pitch/Modulation wheel, Ribbon Controller.**
- **Control Inputs: Assignable footswitch, assignable footpedal.**
- **Outputs: L/Mono, R, Headphones.**
- **MIDI: In and Out.**
- **Display: 2-line x 16-character LCD.**
- **Dimensions (mm): 925(W) x 349(D) x 113(H).**
- **Weight: 8kg.**

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unfortunately, delay times are not displayed in ms. All is not lost, however, since (just as with the LFO) chorus rate and delay time are both syncable to MIDI clock or the internal arpeggiator at a variety of intervals.

I'm sure you'll have realised by now that this is one very accomplished synth engine, capable of a very wide variety of sounds. Within the factory presets is the standard fare of Minimoog and 303 bass impersonations, biting lead sounds, and big, rich pads that could be coming from a Prophet 5 — or, indeed, a Jupiter 8. The most noticeable sonic attribute of the JP800 is its very 'sharp' sound, which has more in common with genuine analogue instruments than it does with the (sample-based) JD range of instruments. I've always felt those instruments have a 'glazed' edge, but thankfully this is not the case with the JP8000, which has a very up-front sound.

The JP8000 is effectively a bi-timbral synth, and each of the upper and lower Parts may be assigned their own individual MIDI channels, in addition to a global MIDI channel, to facilitate program changes and so on between Performances. It is also possible to select chorus, delay and LFO sync on/off for each of the Parts, and Patches or Performances can be saved individually or *en masse* to a suitable SysEx device.

IN CONTROL

The JP has been designed very much as a performance instrument. The level and flexibility of real-time control is perhaps greater than we've seen before on any synth. All front-panel knobs and sliders receive and transmit their own MIDI controllers, for recording real-time changes into a sequencer. Several real-time controls may also be simultaneously assigned to the ribbon controller. For example, you could set up the controller so that when you slide your finger to the left, frequency and resonance ramp to their full settings, returning to their original setting when you move your finger to the right, while Osc 1 feedback amount and harmonics move to *their* full settings at the same time. Individual parameters can be adjusted as little or as much as you like, and assigning parameters to the ribbon controller couldn't be easier. Multiple parameters may also be simultaneously modulated by keyboard velocity, in much the same way, and just in case this still isn't enough, an individual parameter may be controlled from the control pedal. But there is no doubt that the most usable controller is the ribbon. I went mad with this, assigning multiple controllers working in different directions, and came up with some truly amazing morphing textures.

SECOND THAT MOTION

Also on the real-time control front, Roland have come up with a very innovative method of adjusting multiple knobs and sliders (apart from assigning them to the ribbon). It's called Motion Control, and it's basically a sequencer for recording knob and slider movements. Up to eight bars of multiple slider movements may be recorded into a total of four locations. Multiple takes may be recorded by cycling around a pre-chosen section — so, for example, you could first record the movement of the Cutoff Frequency control, then perhaps the Resonance control, and then adjust the Pulse Width on Osc 2. It's also possible to delete individual knob and slider movements at a later date, should you decide you don't like them. Once you've recorded your motions, it's then possible to trigger the sequence at any time, via dedicated front-panel buttons, though you can also choose to re-trigger a sequence every time you play a new key. This is an excellent function and will enable you to make some really complicated real-time changes without lifting... well, alright, one finger.

The JP8000 has an onboard arpeggiator that supplies all the expected Up, Down and Octave variations, as well as a Hold function and the ability to sync to MIDI clock and transmit arpeggiated notes via its MIDI Out port. As on their recent MC303, Roland have also supplied a selection of Beat Patterns which alter the gate time and accent of the arpeggio in real time. Further to this, they've built in their RPS phrase synthesizer, familiar from other Roland instruments, which supplies a total of 48 preset patterns (musical phrases) which can be triggered from a single keystroke. It is possible to replace these with your own patterns, which may be recorded in real time and then quantised, if necessary.

CONCLUSION

After spending some time with the JP8000, you begin to realise that Roland have perhaps been sitting back and watching the competition, while thinking about some genuinely new and innovative ways to improve on existing instruments. There's little doubt that the JP8000 has been inspired by Clavia's Nord Lead and also, to some extent, by the Korg Prophecy, as it

seems to share many of the features of both these instruments. However, it also has significant innovations of its own to offer, notably the flexible way in which parameters can be assigned to the ribbon controller, the arpeggio Beat Patterns, and the Motion Control function. But, as I've mentioned, the most striking thing about the JP8000 is its overall 'sound' and the ease with which new textures can be created. The JP8000 will appeal to a broad band of users who are after that traditional analogue sound, but those who are involved in making dance or electronic music, particularly, will be foaming at the mouth.

Watch out for the usual in-depth SOS review in the very near future.

Iomega Jaz

Removable 1Gb Hard Drive

Iomega's removable Zip drive has been a huge success for cost-effective data backup. Now the much larger capacity Jaz, which is fast enough for direct to disk audio use, looks set to continue the Iomega success story. PAUL WHITE enters the Jaz age...

When it comes to computers and their peripherals, we're pretty well used to prices falling and capacity increasing, but even so, Iomega's nifty 100Mb Zip drive caused a lot of excitement when it was first announced back in 1994: almost overnight, the clunky and expensive small-capacity Syquest drives that we all

thought were so great at the time were consigned to history. So popular was the Zip drive that it's probably fair to say that it's only in recent months that supply has been able to meet demand, and despite reservations over reliability, both the drive and the media have proven to be adequately robust so far.

Though the Zip isn't fast enough for direct-to-disk audio (and 100Mb isn't really big enough for serious jobs either), it excels

as a storage medium for audio samples and for general large-scale backup work. In some ways, keeping audio samples on multiple 100Mb cartridges feels more secure than the 'all eggs in one basket' approach of using multi-gigabyte hard drives. Even so, Iomega were aware of the need for a faster, large-capacity drive to meet the demands of the growing multimedia market, so the Jaz was launched, hot on the heels of Zip.

The Zip drive medium is, in effect, a precision floppy disk mounted within a 3.5-inch rigid shell. The slightly thicker 3.5-inch Jaz cartridge houses a rigid, dual-platter mechanism incorporating an 'anti-rattle' device to help protect the cartridge during transportation. The drive spins at 5400rpm (revolutions per minute) and has a formatted capacity of 1070Mb (over 1 Gigabyte) with a 12ms average seek time. With a sustained data transfer rate of between 3.77 and 6.73Mb per second, Jaz rivals current fast hard drives in speed, making it eminently suitable for multitrack digital audio. Several hard disk recording systems are available with Jaz drives as an option: Roland's VS880 digital 'studio in a box', for example, can be supplied with an internal Jaz drive if required. While external Jaz drives are SCSI only, both SCSI and IDE internal versions are available.

ALL THAT JAZ

The Jaz drive itself is tiny — about the size of a modem — and uses the new high-density, 50-pin SCSI connectors. A SCSI lead is provided, along with a 50-pin to 25-pin SCSI adaptor, so that I could hook up the drive to my antique Mac Centris 650. Cosmetically, the Jaz looks not unlike the Zip, the main difference being that it's green rather than blue. Also, very importantly, the Jaz must be mounted flat — it can't be used on its side.

Power for the drive comes from the increasingly familiar (but not increasingly popular!) external PSU, and the majority of the installation software comes on the cartridge included with the drive, though the package

also includes two floppies (one Mac, one PC), to get the installation process rolling. Installation routines are supplied for both Mac and PC. Also on the cartridge are a number of utility programs, including *Jaz Tools*, designed to assist with cataloguing and backing up files.

A recessed numerical selector switch on the rear panel of the green plastic case allows the SCSI ID to be set to any required value; termination can either be set by the user or left for the Jaz to configure automatically. The mean time between failures is quoted as being 250,000 hours — which equates to over 28 years of normal use. Blank Jaz cartridges are now well under £100 each, and when you consider that this piece is been written by a man still smarting from paying £2500 for a 650Mb external drive (and not as many years ago as you might think!), this is indeed a bargain.

INSTALLATION

Installation is straightforward enough, and involves inserting the *Jaz Tools* disk, then running the Mac installer from floppy. Installation must be undertaken with the extensions switched off (start the Mac holding down Shift), then the Mac must be restarted after installation is complete. The Jaz drive icon will now appear on the desktop. The only problem I encountered was that the drive sometimes tried to spit out the disk before it got around to reading it, so I had to physically hold it in for a second or two until it was accepted. Because the *Tools* disk contains essential installation stuff, it's as well not to wipe this, though there is plenty of room left on the disk for new data.

TESTING TIMES

Trouble-free though the installation was, running the Jaz with Pro Tools III hardware

was less straightforward — the Jaz driver couldn't find the drive. In Pro Tools III, the audio drives connect directly to the Digidesign card, not to the Mac's SCSI port, and in the natural scheme of things, the Digidesign disks mount pretty late on in the boot-up sequence. I initially thought that my problem was down to the Jaz driver trying to load before the disks were mounted by the Digidesign software, but it turned out that the drivers just don't hit it off, no matter what order they load in.

A call to Digidesign's tech support revealed that the Jaz drive was not yet approved for use with Pro Tools III, though Digidesign are apparently still doing tests on it. Even so, I was told that it is possible to mount the drive using the Mt. Digi Control Panel widget rather than its own driver — which was true — but from that point on, I found I couldn't copy any data from any Digidesign-connected drive to any other without the system freezing solid. Even after removing the Jaz drive and all traces of its occupation, I still couldn't copy between my two conventional Digidesign-connected drives without locking up the system again!

It turned out that something in the install procedure had corrupted my Digidesign extension, so after ditching the Jaz driver and reinstalling the Digidesign extension, I found the system was back to normal — and the Jaz drive mounted with no problem, providing I let Digidesign do the mounting. Of course, Pro Tools II hardware — which uses the Mac's SCSI port — will work OK (even my old Sound Accelerator card seemed quite happy), and for Power Mac users, the internal audio will work fine, but it remains to be seen how compatible Pro Tools III is at the moment. Even so, I decided to plough on with the tests to see what happened.

Working on a typical song-length, 8-track composition, the Jaz seemed to perform just as well as my fixed drive, with no evident glitching, complaining or dire warning messages. What's more, the Jaz runs a lot quieter than my other hard drives — there's a little bit of a whine as it gets up to speed, but after that, the noise level is surprisingly low. When you finish work, dragging the drive icon to the trash can ejects the disk automatically.

SUMMARY

Because Digidesign don't yet recommend the Jaz drive for use with Pro Tools III, it would be foolish for me to suggest you rush out and buy one for that particular purpose, even though my limited tests showed up no problems once the installation hassles had been sorted out. For any other audio application where the Jaz connects directly to the SCSI port of the Mac, things should be more straightforward, but because anything associated with computers always includes a propensity for conflicts and incompatibilities, always check with the manufacturers of both your software and computer, just to be on the safe side. I only tested the drive up to eight tracks of audio; I don't know what its practical limit is, but for most private users, this will probably be enough anyway.

As a sample-archiving system, the Jaz has a lot going for it, though as I mentioned earlier, I feel more secure spreading my sample library over multiple Zip drives — but then you can always buy two Jaz cartridges and keep a backup.

The bottom line is that the Jaz drive is great value — it costs little more than a regular fixed drive — it's quiet, it's fast enough for audio, and the media is sensibly priced, so providing it's compatible with your system, it seems to be the best removable option at the moment. SOS

TL Audio Crimson

3011 4-Band & 3012 Parametric EQs

Do TL Audio's designs stand up without valves, or is it a case of tubeless tires? PAUL WHITE endeavours to find out.

TL Audio have made a big impression creating high-quality, tube-based audio processors, and their relatively recent Indigo range (see reviews in the August, September and November issues of *SOS*) brings tube performance within the reach of most serious project studio owners. However, there's still a pool of end users for whom even the Indigo range is too expensive, and there are even people out there who don't particularly like the sound of valve gear. To address this market area, TL Audio have launched the Crimson series — a range of processors almost identical to the Indigo range, but incorporating all solid-state circuitry.

The two models on review are the 3011 2-channel equaliser and the 3012 2-channel parametric. There are five units in the series at the moment, the others being the 3001 quad mic preamp, the 3021 2-channel compressor and the 3041 2-channel, 100W power amplifier (due for release in Spring). All five Crimson models are built into 1U, mains-powered boxes, and are fitted with both balanced XLR and unbalanced jack audio connectors — even the power amplifier.

COMMON FEATURES

As with the Indigo series, both units run on mains power, and both unbalanced jacks and balanced XLRs are provided for the audio inputs and outputs. The front panel, high-impedance instrument jacks have been retained, and these have dual sensitivity to allow them to work with guitars, keyboards and so on. A continuously-variable front-panel gain control (affecting all inputs), with +/- 20dB of range, is available for level matching, and all three inputs (jack, XLR and instrument jack) may be used simultaneously, though there's no independent control over their levels.

Even though they don't contain valves, these are very heavy units, but unlike the Indigo series, there are no ventilation slots. Even so, they run very cool indeed. The front panels are a sort of 'well-oxygenated blood' colour with white and orange-brown legend, while the rest of the case, including the rear panel, is plated. Mains comes in via an IEC detachable mains lead, and the operating voltage may be selected using a recessed switch adjacent to the mains connector.

EQ3011 DUAL 4-BAND EQUALISER

The EQ3011 is a dual, 4-band equaliser with shelving high and low controls and two sweep

mid-range controls. It's more like a console EQ than most stand-alone equalisers (and that may well be where it started its days). At the start of the signal chain are the three input types mentioned earlier, and the gain control. The Hi instrument setting is for use with sources such as electric guitars and basses fitted with passive pickups, though you can plug in high-impedance mics as well, if you need to. Each channel has its own EQ In button with green status LED, and the mains switch is located at the extreme right of the front panel, also with a green LED. A Peak LED lets you know when you're getting close to clipping.

There are switchable LF and HF frequencies for the shelving filters (80Hz/120Hz and 8kHz/12kHz), and as with the two mid-band controls, the available range is plus or minus 12dB. Unlike the Indigo EQ, which has switchable mid-range frequencies, the Crimson's mids are fully variable, which to my mind is rather more flexible, albeit at the expense of accurate stereo matching. The mid-range bandwidths are preset to provide a fairly wide, musical EQ rather than for precision surgery. If you need to do detailed work, the parametric EQ

— which I'll come onto in a moment — is probably going to be more useful to you.

PERFORMANCE

The 3011 drives like a typical desk EQ, but I have to admit that even without valves, it sounds a lot better than most. The fully-variable controls make it slightly more flexible than its Indigo twin, but the downside is that you lose out on the subtle valve coloration. The sound is tight, detailed and free from harshness or phasiness, giving it more of a classic feel, but given that the parametric costs exactly the same, I think I'd be tempted to send my money in that direction, purely because there's so much more control. Even so, this is a very nice, easy-to-use EQ.

3012 PARAMETRIC EQUALISER

While there are slight differences between the control functions of the Crimson and Indigo 4-band equalisers, the parametrics seems

identical in all respects other than the lack of valves. The 3012 is basically a dual-channel, 2-band device, though both channels may be cascaded so that the unit can be used as a single-channel, 4-band parametric EQ when required. In single-channel mode, the output of

Equalisers

channel A feeds the input of channel B, though output A still remains active.

All four bands have a +/-15dB cut/boost range and the frequencies are continuously variable, as you'd expect. Band 2 has a 'divide by 10' range switch, while band 3 has a 'times 10' range switch, in order to provide optimum spectral coverage in both 2-band and 4-band modes of operation. A single button switches between 2-band stereo and 4-band mono operation, and once again, each channel has its own In/Out switch, complete with green status LED.

PERFORMANCE

The 3012 is definitely the most flexible of the two units I've looked at here (though that's not to demean the performance of the 3011 in any way). Most of the time I tend to need stereo EQ, but even though the 3012 only provides two bands per channel in this mode, there's still plenty of scope for spectral tailoring —

and to be honest, I know very few people who can make constructive use of more than two, or perhaps three parametric bands at one time. The 3012 has a clean, shimmery top end that doesn't lapse into harshness as soon as you pile on more than 2-3dB of boost, and the low end manages to keep the bass sounding tight and well integrated.

Used over a full mix, this equaliser lets you add air and detail to the top end without wrecking the overall feel or tonality, and by the same token, if the bass end needs a bit of help, this can be achieved without making it sound flabby. Similarly, if you have a nasty peak to deal with, simply wind up the Q and dig in — the 3012 can handle it.

When applied to single tracks or instruments, the 3012 has the ability to shape the sound in a very natural way, but at the same time, there's more than enough range to radically change a sound if that's what you need, especially using higher Q settings. As with the 3011, there is a slight sense of missing something — at least if you've had a chance to hear the tube-powered

Indigos — but tubes aside, both these equalisers are very competent performers.

SUMMARY

Choosing Crimson over Indigo will save around £200 per unit, and though my own preference would be to hold out for the Indigos, the Crimsons deliver a quality equaliser feel at a price not far above more run-of-the mill boxes. What's more, anyone involved in live performance might feel more secure if they don't have to worry whether the valves have survived the last trip or not!

From a marketing viewpoint, the Crimson range makes a lot of sense, as it allows TL Audio to offer high-end, mid-price and near-budget priced studio gear without having to compromise their sonic ideals. As with the rest of the range, the Crimsons look great, they're good at what they do and they're affordable. If you can't stretch to the luxury of valves, the Crimsons might well suit both your ears and your pocket.   **SOS**

£ **3011/3012 £468.82 each inc VAT.**
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T **01462 490600.**
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Emagic ZAP

Lossless Audio Compression Utility for Macintosh

PAUL WHITE tries Emagic's new audio file compression utility while balancing the implications of time and space.

Digital audio recorded to hard disk is an established reality — it's just that it's an established reality that also happens to take up an enormous amount of disk space! Emagic — of *Logic* fame — have just launched a simple utility for the Macintosh (a Windows version is in the pipeline, but is not yet available) called *ZAP* (Zero-loss Audio Packer), and while it won't solve the problem completely, it does enable audio files to be compressed to around half their original size in a totally loss-free manner. This is obviously a major saving in disk space — so what's the catch?

The answer is that there *is* no catch in *ZAP*'s compression process — this isn't like the data-changing compression systems used in DCC or MiniDisc, where less important information is discarded; it's more like the audio equivalent of *ZipIt*, *Disk Doubler* or *Stuffit*, programs used to compress computer files or applications. What you get back when you un-*ZAP* (for want of a better term) is exactly the same file you started with. Though *ZAP* is intended mainly for users of *Logic Audio*, it can compress and decompress files in *Sound Designer II* format (mono or interleaved stereo), 16-bit, uncompressed AIFF and both mono and stereo 16-bit WAV files.

Around 1.5Mb of free RAM is needed to run *ZAP*, which will work on any 68020 Mac or faster, provided the operating system is at least System 7.0. A Power Mac native version of the code is included in the program. Installation is from floppy, and this being the music business, the software is protected by a disk install system — you can install up to two times from the master disk, and you can also deinstall and restore one installer to the disk if you need to reformat or change drives. Once installed, the software is authorised to run from one specific hard drive.

USING ZAP

ZAP's user preference box lets you determine whether the original files should be trashed after compression or not, and whether or not you want your compressed files to be self-expanding — which means that you don't then need *ZAP* to restore them to their former glory. So, why not make all files self-extracting? The answer is that a self-extracting file contains a mini-program to unravel it, so the file size is around 30K larger than it would be if it were not self-extracting. On large files, this isn't much of a penalty, but if you're into saving hundreds of short files, it might be significant.
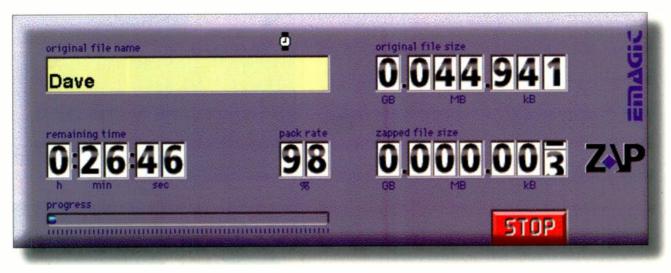
Self-extracting files made using the Mac version of *ZAP* will only be able to self-extract when used on a Mac, though you can decode them if you un-*ZAP* the file from within *ZAP* Windows when this becomes available (as opposed to just double-clicking on the file). The easiest way to compress a file, according to the manual, is to drag the file icon onto the *ZAP* icon — compression should then follow automatically. You can also go via the menu within *ZAP* if that makes you feel better. In

fact, I couldn't get 'drag and drop' to work at all, though working from the menu presented no problems. *ZAP* can also compress multiple files, either individually or as a single archive, depending on the preference settings. A neat feature is that *ZAP* works in the background, so you don't have to stop using the computer to let it do its thing. If you do compress a bunch of files as a single archive, the downside is that you have to uncompress the whole archive to get them back again, whereas if you choose to have the files compressed on an individual basis, they can also be retrieved individually. If you select the audible warning to let you know *ZAP* has finished unpacking a file, you are greeted by a very *Wayne's World*-ish 'Excellent!'.

INSIDE ZAP

ZAP works using a principle called 'non-linear redundancy elimination', which involves a statistical analysis of the audio material to locate

redundant material. The redundant material isn't discarded, but rather is digitally 'described' in more economic terms than simply storing the number for each sample. Totally uncorrelated material, such as white noise, does not tend to get any smaller if ZAP-ped, but more coherent waveforms can be reduced inside by up to 50% or so. Speech with pauses may be often be compressed beyond a 60% saving. Because the process depends on the nature of the sound file data, it isn't possible to say exactly how much space you'll save until you've done the compression, but then this is also true of the computer data compression systems used to compress graphics or text files. If a ZAP file becomes corrupted in any way, ZAP will announce the fact when you try to decompress it — you won't be left working with a dodgy file without being informed.

REASONS TO ZAP

One of the main reasons to ZAP files is so that they can be archived without taking up so much space, but it's also beneficial to ZAP audio files destined for transmission over the Internet. With the proviso that self-extracting Mac-created archives can't self-extract under Windows, this could save a lot of time when transmitting audio files that must not be

compromised in quality. It's useful to note that the self-extraction code is written for 68xxx machines to ensure maximum compatibility, while the native ZAP application works rather faster on Power Macs. For this reason, Power Mac owners are better off opening files from within ZAP, rather than relying on the 'double-click to self extract' operation.

IMPRESSIONS OF ZAP

One of the disconcerting things about ZAP is that it estimates the time the operation will take, and then seems to keep amending it — upwards. On my admittedly old 68040 Centris 650, a four-minute stereo SDII file came up with an initial estimate of around 15 minutes, but after five minutes or so had elapsed, this had increased to around 23 minutes, and showed little sign of going down again! Eventually, I went into the lounge to watch *Drop the Dead Donkey* — well, you have to, don't you — and when I came back, the job was done. I must confess that I didn't have the patience to do it again to get an idea of exactly how long it took — suffice it to say that it's long enough to be boring. UnZAP-ping is slightly faster, but still takes significantly longer than simply copying a file of the same size — around 20 to 25 minutes in this case.

On a stereo file, I saved around 35% on the size of the original file.

There's no doubt that, other than the puzzling reluctance to 'drag and drop', ZAP does as it claims, and the reconstituted file is an exact duplicate of the original — but unless you have one of the latest light-speed Macs, I feel the process is simply too slow for most situations. Batch processing for long-term archiving is feasible, assuming you have something else to do in the meantime (background processing slows ZAP down even further), but for use during sessions, a super-fast Mac is a necessity, not a luxury. I accept that the amount of processing taking place means ZAP-ping isn't a trivial task, but in my own situation, where I'm working on a Mac that's starting to show its age, and where my files are often of album length, I think I prefer to waste disk space rather than time. **SOS**

Audio Toys Inc

Preamp/Mixer

HUGH ROBJOHNS takes a look at yet another studio unit designed to partner the new generation of digital 8-tracks, this time from a company new to the UK market.

The most popular recording format today is the 8-track digital tape machine. Professionals in broadcast and post-production generally seem to use the DA88 format, while the musical fraternity seems to prefer the ADAT system. In both cases, one of the most common problems is that of finding a suitable microphone preamplifier which can feed all eight channels of the machine, as well as letting you build a monitor mix to hear what's going on.

Audio Toys Inc of Columbia, USA, have produced a purpose-designed preamp/mixer to meet these very demands, using circuitry derived from their Paragon series of mixing consoles. The 8MX2 provides eight channels of microphone preamplifiers (with peak limiters), feeding directly to the multitrack tape machine, plus a stereo monitoring system with feeds from the mic signals (pre- or post-limiters), the multitrack returns, or a stereo tape machine return, plus a stereo recording mixer with its own level and pan controls.

THE BOX

The 8MX2 is housed in a sturdy, 19-inch, 1U rackmounting box, with a bright and distinctive red front panel. The rear of the unit is festooned with socketry — eight XLR microphone inputs to the right, six quarter-inch jacks in the centre, and four D-sub connectors towards the left, finishing the collection with an IEC mains socket on the extreme left. The review model did not carry a CE approval mark.

The electronically-balanced mic inputs each have two miniature toggle switches associated with them. The first provides 48V phantom power, and the second provides an earth-lift facility — particularly useful if the 8MX2 shares microphone splits with another mixer. The microphone inputs have 20dB of headroom and maximum input levels of +24dBu.

The collection of six TRS jack sockets provide the monitoring system inputs and outputs: loudspeaker/headphone monitoring and mixer outputs for recording on a 2-track machine, plus a 2-track return — all in stereo, of course, and all electronically balanced. The two 25-pin D-sub connectors provide balanced line-level inputs and monitoring returns, and are wired for compatibility with Tascam DA88s and Fostex CX8/RD8 machines, although suitable cables are also available for interfacing the 8MX2 with Sony PCM800s, Panasonic MDA1s, Roland DM800 and Alesis ADAT machines, amongst others. Two 9-pin D-sub connectors provide a means of interconnecting the monitoring busses between multiple 8MX2 units, to allow elaborate systems to be constructed — offering, say, 24-track recording capabilities whilst allowing easy monitoring from a master unit.

TWIDDLY KNOBS

The front of the 8MX2 is a mass of dual-concentric rotary controls (18, to be precise) and small push buttons (27, no less!). Unfortunately, squeezing this number of controls onto the limited real-estate of a 1U box has resulted in rather cramped and confusing labelling, although once you've become familiar with the control functions, the labels become much less important.

Each of the eight channels has a pair of dual-concentric rotary controls and four push buttons. The microphone input gain is set by the blue-capped inner knob of the lower control, which has a fine, 41-position detented action and a range of 0-65dB. In most

applications I'm sure this gain range will be perfectly acceptable, but personally I would have preferred a slightly wider gain range to accommodate quieter sources or the more distant microphone techniques. As the input is capable of handling signals up to +24dBu, an input pad is not necessary. The bottom two push buttons relate to the microphone signal: the upper of the two reverses the polarity of the microphone signal, and the lower one makes the channel available on the monitoring system.

The outer knob of the lower pair of controls sets the peak limiter threshold for the channel, and since overloads on digital formats are so unpleasant, the inclusion of independent peak limiters on each channel is an excellent idea. The threshold is continuously variable between +24dBu (effectively out of circuit) and +4dBu, and the ratio is said to be infinity:1. No information was given about the attack and release times, but they are clearly extremely short — in practice, the limiter provides a useful and effective overload-prevention system, although care should be taken to set it such that it really does only operate in emergency situations.

The upper set of concentric controls provides level and pan facilities for the stereo mixer output, which are pretty self explanatory. A push button directly below these knobs routes the signal to the mix buss, whilst another to the left selects the input to the level control. The default is the microphone signal, but pressing the Return button selects the corresponding multitrack return signal, allowing the 8MX2 to be used to handle overdubs as well as straight recordings.

8MX2

STEREO MIXER

The stereo mixer couldn't be simpler: the eight inputs are selected between multitrack returns and microphone signals (after the recording gain and limiter thresholds have been set), panned and balanced appropriately, and passed on to the master level and balance control at the right-hand side of the unit.

I don't understand why Audio Toys Inc felt it necessary to include a balance control on the stereo recording feed — it seems to me that this would have been more appropriate on the headphone/loudspeaker monitor output, which presently has no such facility — strange...

With the 8MX2 as supplied, the 2-track return is only available to the monitoring system, but a couple of internal jumpers allow the return signal to feed into the mix buss if desired. In this mode, the return is at unity gain (there is no provision for any level control) and the idea is to make use of this input as an effects return for the stereo mix. This could prove to be a useful facility for a reverb return, although care should obviously

be taken to avoid returning the output of the stereo mix recorder back into the mix buss.

MONITOR SECTION

The monitoring section on the extreme right of the front panel consists of a pair of LED bargraph meters, a headphone socket, a monitor level control and three push buttons. The level control speaks for itself, but the metering is a little unusual. One 10-segment bargraph provides a level meter, while the other shows the amount of gain reduction applied by the selected channel's limiter.

The basic arrangement is that the monitoring section allows the user to audition each input channel individually, to set up the gain and limiter. In this case, the level meter shows the output level for that channel (from -3 to +24dBu in 3dB steps), whilst the other meter (labelled Atten) shows any gain reduction from the limiter (scaled from -2 to -20dB). A push button to the left of the section causes the channel cue signal to be taken from either pre- or post-limiter processing or from the multitrack return signal.

When no channel is selected to feed the cue monitor system, it defaults to the output of the stereo mixer, the level meter showing a mono sum (left plus right minus 6dB) of its output. A push button adjacent to the monitor level control allows the 2-track return to be monitored in stereo.

The manual carries warnings that the

headphone output is only suitable for driving high-impedance headphones, and not the common domestic 8Ω types. I don't understand why, in a mains-powered unit, the headphone amplifiers could not be designed to drive all headphone types, and I didn't have any low-impedance models to find out what happened! With 400Ω and 2kΩ headphones, however, there was plenty of output level.

CONCLUSION

The 8MX2 is a useful piece of equipment for any DA88 owner. If you wish to make multitrack recordings direct to DA88, but with the advantage of a stereo monitor mix, this is the tool for the job. Generally well designed and with good sonics, it is really only let down by one or two bizarre design features, such as the balance control on the stereo mixer output. In use, it proved reliable, quiet, easy to set up and, above all, transparent (assuming that the limiters were not active). I found the 8MX2 an indispensable partner for the DA88. **SOS**

LA Audio GCX2

2-Channel Compressor/Gate

This UK-designed and built processor offers two channels of easy-to-use compression and gating for under £200. PAUL WHITE's backing Britain.

Over the past few years, the amount of competition in the audio processor market has succeeded in forcing prices down to rock bottom, and with budget brands such as Behringer soaking up a hefty share of the low-end business, companies normally associated with mid and high-end products have also shifted their corporate gaze to the mass market. One such company is LA Audio, who started out building products like the Classic Compressor, then designed lower-cost spin-offs such as the 4x4 compressor/gate (reviewed *SOS* July 1996) and the resplendently green LA Lite range (reviewed *SOS* December 1994 and November 1995). The new GCX2 compressor/gate, reviewed here, represents a further significant step down in cost: LA Audio's engineers have evidently concentrated on trimming away unnecessary features and making their designs even more cost-effective to fabricate.

The GCX2 could be viewed very much as an entry-level processor, though (for the benefit of the more discerning user) it still features balanced XLR ins and outs at +4dBu, as well as balanced TRS jacks at -10dBV, plus TRS side-chain insert jacks. On a personal level, I'd have preferred to see the jack ins and outs running unbalanced, simply because most consoles have unbalanced insert points, and running unbalanced to balanced (and vice

versa) often results in a loss of signal level. However, the GCX2's output stage is designed so as to avoid this problem.

FEATURES

Essentially, the GCX2 is a two-channel processor, where each channel comprises one stage of gating and one stage of compression. Both the compressor and the gate operate from the same side-chain RMS level detector, which in turn monitors the input signal level. This is a slightly unusual arrangement, as most compressors derive their control information from the VCA's output rather than from the input, but as this is how some of the older vintage models got their dramatic sound, LA Audio may have adopted it for this very reason. The 6-LED gain-reduction meter monitors the compressor action, while a single green LED shows what the gate is up to. When the gate is bypassed, the green LED dims but continues to monitor the gate action.

Like other LA compressors, the circuitry uses a soft-knee circuit combined with a variable ratio, so that as the signal approaches the threshold level, the gain reduction ratio increases progressively to its set level, rather than coming in all at once. As a rule, this leads to a smoother compressed sound. Threshold adjusts the level at which compression starts, while Ratio is adjustable from 1:1 (no compression at all), right up to 20:1 (heavy limiting). Both the gate and compressor have their own Bypass buttons, but the controls are far simpler than you'd expect to find on most studio processors. For example, there's no attack or release control — just an 'auto' setting with switchable Slow and Fast response times. In Slow mode, the attack time varies from 10ms to 100ms, depending on the transient nature of the input signal, while the release time is fixed at 3s. In Fast mode, the attack can range from just 5ms to 70ms, with the release set at 0.5s. The usual output gain control is provided to make up for any gain

loss due to compression, and the centrally-mounted Dual/Stereo button allows both compressor channels to be linked for stereo operation, when required. In this mode, both channels are set up using the compressor controls of channel 1. The gate section combines the gate side-chains in such a way that both channels open if either side is triggered.

If the compressor is straightforward (and it is), the gate is even simpler, featuring just two knobs and a Bypass button. The Threshold control sets the level below which gating takes place, while the Release control adjusts the gate release time from Fast to Slow. In Fast mode, the release time is a snappy 30ms (suitable for drums), while in Slow mode, the release time is a leisurely 3s, to ensure that gently-decaying sounds aren't snuffed out before their allotted time. The gate range is fixed at 80dB attenuation — which is as close to complete muting as makes no difference — and even though the gate attack time is preset to fast, I didn't notice any clicking when gating non-percussive

sounds. Conversely, on digitally-recorded drums, the gate opened fast enough to keep the transient edge of the sound intact.

PERFORMANCE

Because of the simplified control system LA Audio have adopted for the GCX2, the unit is very easy to set up, yet you rarely feel that you've lost control in an area where you really needed it. For general-purpose gating and compressor applications, the GCX2 provides all the control you need, though for more specialist gating tricks — such as creating gated drum sounds, using external keying, or working with sounds where side-chain filtering is needed — you really need the 4x4 or a dedicated full-function gate such as Drawmer's DS201.

Modest amounts of gain reduction can be achieved without incurring any obvious side effects, but if you start to exceed 6 or 8dB of gain reduction, the sound begins to take on a warm, obviously-compressed character, where the compression moves out of corrective territory and instead becomes a creative effect. All the usual sound sources, including voice, bass guitar and acoustic guitar, were wheeled out for testing, and all came out pretty well. Performing the same tests with a more esoteric compressor (costing almost three times as much) produced a better-focused result, with less effect on the transient definition of the material being processed, but considering the very low cost of the GCX2, it performed extremely well and produced warm, musical results.

The gate side of the GCX2 is positive and straightforward to set up, and in most situations, that's all you could ask for. Indeed, I can see the GCX2 being taken up by the live sound user, precisely because you don't have to spend ages fiddling with it. It doesn't have the flexibility of a gate with side-chain filtering or one with separate attack, hold and release times, but for routine clean-up work, it gives no cause for complaint.

SUMMARY

While those of us who have been recording for a long time prefer to buy fewer pieces of really high-quality gear that will deliver first class results and will last a few years without becoming

obsolete, I recognise that there is a need for a 'more bang for the buck' range of outboard gear that combines respectable audio performance with ease of use. Seen in this context, the GCX2 offers extraordinarily good value for money, and it's capable of producing fine results, providing you don't ask it to do anything too far out of the ordinary. Indeed, for those not experienced in setting up compressors and gates, there's a better chance of getting good results from a box like this than from something that provides all possible functions and facilities.

To sum up, the GCX2 is undoubtedly a good compromise between circuit quality, ease of use and build quality. The styling is a great

improvement on most budget equipment, and the inclusion of properly-balanced XLR inputs and outputs is appreciated. Indeed, LA Audio are to be applauded for bringing to market a home-designed, home-built product that is able to compete on price with mass-produced, built-in-China imports. **SOS**

Westlake LC

Nearfield Monitors

PAUL WHITE lends an ear to Westlake's new nearfield monitors and discovers that they have a distinctively Californian sound.

Westlake Audio claim 25 years of experience in the field of monitor design, but this is the first time, to my knowledge, that they've made a deliberate move to enter the project studio market, with the passive, nearfield LC 6.75 monitors.

FACTS & FIGURES

Surprisingly little technical information was supplied with these speakers (a single-sheet brochure), but I *can* tell you that that the bass driver is 6.5 inches in diameter, and is augmented by a 3/4-inch soft-dome tweeter.
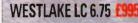
The LC 6.75s have a rated continuous power handling of 80W, are quite sensitive, at 88dB/Watt @1m, and have a frequency response of 60Hz-18kHz (quoted at the standard -3dB limits). The cabinets are constructed from MDF (Medium Density Fibreboard) and finished in black textured paint, and feature dual bass ports directly below the tweeter. A 'step' in the front panel is designed to time-align the tweeter with the bass unit, though no mention is made in the documentation of the crossover frequency or slope characteristics.

This is a strictly passive monitor, with a nominal impedance of 7Ω, though it can dip to as low as 5Ω at some frequencies, so an amplifier capable of working comfortably into 4Ω is to be recommended. I used the well-regarded AVI integrated hi-fi amp (rated at around 60W per channel) for these tests and found I had plenty of power in hand.

The bass/mid unit seems nicely damped, using a doped paper cone in a roll surround, and the HF end is covered by a fabric dome

tweeter set into a rebate in the baffle, so as to provide flush mounting. Judging by the size of the dust cap, the voice coil is probably 1.5 inches or less in diameter. A removable grille cloth and frame is fitted as standard, and this seems to have little or no detrimental effect on the sound, though most studio users will probably prefer to remove it.

Connection is via a pair of conventional binding posts set into a small recess in the rear panel, and these can accept bare wires, banana plugs or spade terminals. With overall measurements of 16 x 8 x 10 inches (as these are American speakers, there are no metric equivalents provided), these are relatively

"This is a strictly passive monitor, with a nominal impedance of 7 Ohms, though it can dip to as low as 5 Ohms at some frequencies, so an amplifier capable of working comfortably into 4 Ohms is to be recommended"

6.75

compact monitors, and at 22lbs each, they feel substantial while remaining portable.

LISTENING TESTS

All my listening tests were conducted using the same, very familiar set of CDs I use to test virtually all the monitors that find their way here, and by way of a benchmark, I also had a pair of ATC SCM10 monitors to hand.

The stereo imaging of the LC 6.75s immediately leaps out as being excellent, while the off-axis response is particularly good, ensuring a usefully wide sweet spot. There's plenty of level, even with a quite modestly-powered amplifier.

At the low-frequency end, you wouldn't expect any really deep bass from monitors of this size, but at moderate listening volumes, you do get a reasonable kick out of them. However, I felt the rendition of bass-heavy transients left something to be desired. Whenever I read a hi-fi review that rants on about "pace, rhythm and timing" in the context of loudspeaker appraisal, I tend to think they're just talking their usual load of techno-cobblers, but the LC 6.75s just didn't seem quite together when delivering kick drum and hard bass guitar sounds. This low-frequency characteristic is relatively subtle, but less benign is an obvious sense of brightness, which seems to be centred between 4 and 5kHz. The subjective impression is almost as if aural enhancement has been used on the monitoring system, and though this makes it easy to pick out detail, it definitely imparts a harsh, almost sibilant characteristic to the test material, and this makes it difficult to believe what you're hearing. I don't know if this particular sound quality is an artifact of the crossover design or whether it's the result of deliberate voicing designed to achieve the 'Westlake' sound — West Coast US studios do seem to go in for bright monitoring, after all — but I'm afraid I wouldn't feel happy working with it. It's almost as though Westlake have used the NS10 as their role model, then decided to build a hi-fi equivalent but with similar overall characteristics. Of course, if you're one of the thousands of engineers who like working with NS10s, this may be exactly what you're looking for.

SUMMARY

The LC 6.75 is well engineered, it delivers adequate level and it has excellent imaging, but I've always felt that a studio reference monitor should sound as flat and accurate as possible. Clearly, the LC 6.75 is designed to have a certain kind of tonal character, and I accept that slight tonal tweaking is all part of making a monitor appeal to the market, but my own view is that Westlake have gone beyond the point at which the speaker is useful as a meaningful reference. In fact, I was so worried by this that I asked two other *SOS* contributors to join in the listening tests, and they very quickly arrived at the same conclusions without being prompted by me. Considering that these monitors cost almost £1000 per pair — not cheap for passive nearfield monitors — my personal recommendation would be to check other models in the price range first, including the Alesis Monitor 2s and Dynaudio BM10s, both at around £800, or put the same money into something like a pair of Event active 2020s (reviewed July 1996), which I feel out-perform the Westlakes on all counts.

Waldorf Pulse Plus

Monophonic Synthesizer

One of the hit synths of 1996, Waldorf's Pulse is now available in a slightly upgraded version, incorporating an external audio input and CV/gate interface. PAUL NAGLE sets it squizzing and phzweeing to find out exactly what has put the Plus in the Pulse.

I t's only fair that I come clean at the outset and declare: I love the Waldorf Pulse. Since I reviewed it a year ago (see *SOS* February '96), it has become a vital component of my studio. The reasons are simple: it has a tremendous bass end, two fast, snappy envelopes, three powerful digitally-controlled analogue oscillators, and two low-frequency oscillators with a remarkably wide frequency range (from 0.0008Hz up to 261Hz — equivalent to MIDI note C3). If that's not enough, the mixer

At first glance, the Pulse Plus doesn't seem radically different. The restful green LED that indicates the position in the programming matrix has been replaced by a dazzling red affair, familiar to Microwave users. An additional level for external audio has been added, and the lettering of the Global section reworded to allow for an extended range of parameters. But it is on the rear of the unit that the most changes are evident: six extra quarter-inch jack sockets give a wealth of new connection options.

SOFTWARE ENHANCEMENTS

In typical Waldorf style, the Pulse's operating system has undergone several upgrades since its release: at the time of the review, it stood at version 1.42. New features include MIDI synchronisation for LFO 1, enhancements to the arpeggiator, and a few fixes. The manual has been overhauled and now includes some helpful hints, as well as a section at the end detailing the extras in the Plus version.

Three LFO waveforms are available for MIDI sync: triangle, sawtooth and square. When controlled externally, LFO speed becomes selectable from a stately eight bars' duration up to 32nd notes, including dotted values. I hope that a future update will allow the Sample & Hold waveform to be MIDI-sync'ed too, with

10) according to their incoming order. Both note and MIDI Clock data are sent via the MIDI Out socket, and arpeggiator settings are memorised on a patch-by-patch basis.

CV OR NOT CV?

Many recent analogue synthesizers have included a CV/Gate interface, but few come close to the extensive implementation that appears on the Pulse Plus, which boasts no

The Pulse Plus's various extra inputs and outputs.

section can be set to overdrive the analogue cascade filter for a wide range of tone colours. An arpeggiator, a flexible modulation matrix and a separate MIDI controller for each parameter all make the Pulse a dream for performance, programming or sequencing.

Not content to rest on their laurels, Waldorf have now released the Pulse Plus, which is equipped with a comprehensive CV/Gate interface and an external audio input to the filter. In this quick tour, I'll look mainly at the new features, so check out the aforementioned *SOS* Pulse review for the full picture.

perhaps the extra sweetener of LFO phase-shifting to better align blips against a sequenced rhythm section.

The arpeggiator is now very respectable, encompassing a maximum range of 10 octaves, selectable from whole to 32nd notes, with triplets and dotted values available for each. Sixteen preset pattern variations extend the possibilities still further. In addition to the usual up, down, up/down and random options, three new Assign modes have been added which replay notes (up to a limit of

fewer than eight parameters with which to fine-tune those pre-MIDI incomings and outgoings. At the most basic level, the CV/Gate Out jacks re-send the notes that drive the Pulse, as either Oct/Volt (as used by Moog, Roland, Sequential,

and so on) or Hz/Volt signals (as used by the Korg MS series, Yamaha CS monosynths, and the like). Four selectable triggering modes should cater for all but the most obscure old synths that may be lurking in your attic. Better still, the interface can be assigned a discrete MIDI channel so that it is addressed independently of the Pulse, providing, in effect, the same functionality as a dedicated unit. The fun doesn't end there, because a second CV output is available: CV2. Its value can be controlled by any of the Pulse's 15 modulation sources; handy if your ageing synth has patch inputs, as in the case of the Korg MS series. A new modulation matrix entry is used to specify source and level. The CV and Gate inputs allow the Pulse to be controlled via pre-MIDI synthesizers or analogue sequencers, providing they adhere to the Oct/Volt standard. When Waldorf decide to add a feature, they don't do it by halves!

AUDIO INPUT

The audio input allows an external signal to be routed through the filter and amplifier sections. Furthermore, since the audio input has a unique controller assigned to it, it is possible to vary the level dynamically via MIDI, as well as swirl it around in the stereo field via the Pulse's twin outputs. When using an external signal, you need to trigger the amplifier envelope in order to hear anything — this can function as a simple but effective noise gate for chopping up audio signals. Other interesting possibilities occur if you connect a polyphonic synthesizer and modulate the Pulse's filter with Sample & Hold. Setting the LFO rate to 0 means that a new cutoff value is generated for each incoming MIDI note, which almost makes up for the lack of MIDI-sync'ed S&H. This is great fun for those fast and wibbly sequenced chords, and if I didn't have this review to write I'd still be playing with it, squeezing some impressive squelches from my otherwise non-squelchy Wavestation.

CONCLUSION

I've had almost a year to appreciate the original Pulse, and I think I'm now brave enough to say that this is the best analogue monosynth I've ever owned — even including my Minimoog. I know that this amounts to nothing less than heresy for many seasoned synth players, and I do concede that the Mini has that instant accessibility factor that the Pulse's matrix and six knobs cannot replace. Nevertheless, I find the Pulse to be more versatile, and it integrates so well into my MIDI studio that I can only urge you to try one out and judge my sanity for yourself.

I think that if I had to make the choice between a Pulse and Pulse Plus, I'd pay the extra £100 and get the Plus version, mainly for the filter input. If I still had any old monosynths lying around, the CV/Gate interface would be a great bonus — far better in my cluttered studio than having a separate unit. But decisiveness was never my strongest point (or was it?) and as a result I now have something in common with Dr Who: I have two Pulses! SOS

SyQuest EZFlyer 230

Removable External 230Mb Drive

PAUL WHITE tries out the SCSI version of this popular drive, and discovers lots of potential applications in the studio.

Over the last couple of years, we have seen a proliferation of low-cost, medium-to-high capacity removable drives such as the Iomega Zip and Jaz drives (reviewed in *SOS* December '95 and on page 38 of this issue respectively), the SyQuest EZ135 (see *SOS* July '96), and now its successor, the EZFlyer reviewed here. The SyQuest EZ135 proved to be fast enough for direct-to-disk audio recording, and I use one in my own setup for storing samples or short *Logic Audio* sound files. However, the EZ135 has been overtaken by its newer sibling, which offers almost double the storage capacity for very little more cost. Fortunately, as a concession to those

who've been trampled beneath the stampeding feet of advancing technology, the EZFlyer can both read from and write to EZ135 cartridges.

FLYER INFO

With a total capacity of 230Mb (closer to 217Mb when formatted), the EZFlyer drive relies on a 3.5-inch cartridge containing a rigid platter, based on Winchester technology; in most respects, it's indistinguishable from the EZ135 disk. With an average access time of 13.5ms, the SCSI version of the drive (reviewed here) has a transfer rate of 2.4Mb/second (4Mb per second burst), while the parallel port version manages only 1.25 Mb per second, though this is still very respectable, and is comparable to many fixed hard drives. This degree of speed makes the drive suitable for real-time audio applications, some video applications, and for any general-purpose backup purpose where medium-length files need to be stored — for

example, samples, MIDI sequence data, or audio files from MIDI + Audio sequencers.

Packaged with the drive is a set of floppy disks containing both PC and Mac installation software, a 25-pin SCSI cable, one cartridge, and a terminator. The mounting software is from Silver Lining, and new cartridges are supplied ready-formatted for either Mac or PC, depending what you ask for. The supplied cartridge also comes with a few bits of free Mac software including some games demos, though due to problems installing my drivers, these were wiped in a reformatting operation before I could check them out. Power, incidentally, comes from an external adaptor that automatically adapts to any local mains voltage between 90 and 270V.

IN USE

Unlike the early SyQuest drives, which were both noisy and cumbersome, the EZFlyer is small, neat, and very importantly, quiet. There's a brief high-pitched whine as the disk runs up to speed, but after that, the noise disappears. If the disk isn't addressed for several minutes, it winds down again and dozes until you need it. For those already familiar with the EZ135, the EZFlyer looks quite different, and also features a simpler loading mechanism. Rather than use the locking lever system, the disk simply slides in and a door closes after it. However, you do have to make sure that the disk is pushed all the way in, otherwise it may not locate properly. Ejection is mechanical, so other than having to wait a few seconds for the disk to run up to speed and mount itself on your desktop, using it isn't unlike using a floppy drive. Even so, because of the mounting and demounting times, using this drive could be frustrating in situations where you have to do a lot of disk swapping. In fact, Iomega's Zip drive is far quicker in this respect, though the actual data transfer rate of the drive is significantly slower.

As with the EZ135, once the cartridge is inserted and mounted, the EZFlyer operates much like any other hard drive, and works OK with Digidesign's *Sound Accelerator* and ProTools II hardware. It also interfaces with SCSI-equipped samplers, though you'll almost certainly have to reformat the disk using the drive management menu in the sampler itself. The disk cartridge packaging is reassuringly protective, and has a transparent outer sleeve beneath which you can fix your own label.

SUMMARY

The SyQuest EZFlyer is a very nice drive; it's quiet and cost-effective, but for serious audio

work, the higher-capacity Iomega 1Gb Jaz drive is a more practical option, and is not hugely more expensive. For my money, the EZFlyer is most useful for archiving samples or for general file backup. For those who already have an EZ135, the ability to read and write EZ135 disks is most welcome. I feel the load/unload operation of the drive has improved over the 135, though the disk eject is a touch on the violent side.

Weaknesses are few, and my only real concern is the time it takes for the drive to mount and demount — around half a minute on my old Mac IIcx. I also had trouble installing the Mac drivers — the self-extracting archive containing them refused to self-extract until some fairly heavy-duty data descrambling utilities had been used on it. Then again, the fact that the unit had already been used before I got it means that something untoward may have happened to the disks in the hands of the previous user, so SyQuest can have the benefit of the doubt here.

What else can I say? This is a neat, cost-effective solution to medium-size file storage and it's fast enough for real-time audio into the bargain, but don't expect it to stay state-of-the-art for long — real life isn't like that anymore where computer peripherals are concerned. **SOS**

Music
ALL ABOUT LIBRARY MUSIC
to go

A surprising amount of the music you hear on TV and radio is not specially commissioned but is taken instead from libraries of ready-written themes. NIGEL BEAHAM-POWELL explains how the library music system works and gives some pointers on getting into it.

Most readers of *SOS* will have heard of 'library' music, but not many will know about how it's used and the amount of money it generates within the UK and abroad.

Around 25-30% of all music used on TV, radio and cable/satellite in the UK is library music, which is sometimes known by its American term, production music. This includes well-known TV themes (*News at Ten* and *Grandstand*), music beds for commercials (Persil), radio station idents and themes (the Radio 4 evergreen *Desert Island Discs*) and about a quarter of the total incidental music within television programmes generally.

Around 70% of the music used in corporate videos comes from the music libraries, and there is a fast-expanding multimedia market which also uses music from the same sources. The total amount of money generated from royalti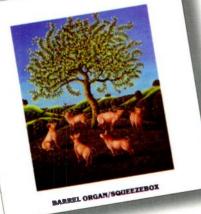es in the UK by library music is between £8 and £10 million. Outside the UK, music written by the main libraries based here has about 20% of the global market share.

WHO ARE THEY?

There are about 80 companies in the UK who make library CDs, ranging from specialist companies who just record military music, for example, to KPM, which is one of the largest and most successful and which is owned by EMI. KPM produce 35-40 CDs a year, and have a back catalogue of over 400 discs. There are many private companies in between, such as De Wolfe and Carlin Production Music, and a series of branded libraries, such as Chappell, Bruton, Killer Tracks and First Com, which come under the Zomba umbrella. One of the five music majors, BMG, runs the highly-successful Atmosphere Music.

WHO COMMISSIONS COMPOSERS?

Most companies have arrangements with composers, and although very few composer/writers are on record company-style contracts, it is not unusual to find composers who produce a library album a year, and who have a long history of working with the same company.

So what sort of music are they on the lookout for? Every style you can possibly imagine, and some you might not — such as barrel organ or patriotic themes from around the world. The major libraries try to cover most areas, from French accordion music to African tribal, minimalist to rave. They aim to have a complete range so that the people who use them can go to them for everything, and are also able to recognise a house style. The smaller companies focus their efforts on exploiting a niche in the marketplace.

By now you might be wondering who these companies commission and how you might get in. Well, there are always opportunities, but they prefer to use composers with a proven track record. It makes sense for companies to use composers who are say, writing music on a weekly basis for TV ads, for a commercial music project. Similarly, if they want serious contemporary music in a minimalist idiom, they will approach someone who has already done it, or who is adept at different orchestral styles. In my experience, it is better to approach a company with a project rather than send in a random cassette of pastiche music. KPM, for example, receive a ▶

CARLIN 138

24 HOUR TELEVISION

CARLIN 144

LIGHT-HEARTED & FUN

CARLIN 230

WILDLIFE

CARLIN 111

BARREL ORGAN/SQUEEZEBOX

ALL ABOUT LIBRARY MUSIC

24 HOUR TELEVISION

LIGHT-HEARTED

WILDLIFE

Nigel Beaham-Powell.

▶ dozen cassettes a week, and they say they listen to every one, but it is relatively rare that this leads to a commission. Carlin Production Music always listen to tapes, and they've found pieces which have been used on compilation projects, but once again it would be extremely unusual to send in a cassette and get an immediate commission for a whole CD. All libraries tend to plan a year or so in advance, so there will always be quite a time lag before a commissioned work gets a CD pressing. Most of the libraries produce catalogues of their work. You can find these in post-production houses, or even obtain them from the libraries themselves. If, for example, you have a talent and a track record in writing sports fanfares or themes, you can check a catalogue of a particular library and see if there's a gap, or something that sounds out of date. If there is, approach the company with some ideas. A bit of preliminary research goes a long way.

Luck, talent and who you know are always pertinent. The standard of library music is high, and getting better. Many composers who are well known in the TV and film industry write very good library music, and it requires pretty sophisticated skills. Generally speaking, the libraries are on the lookout for composers with experience and a reputation, whether it be in the latest dance production techniques or silent movie piano styles.

HOW DO YOU GET PAID?

Most composers do a 50/50 deal on all royalties with the library company. If you were writing a large orchestral project, you'd expect the cost of the orchestra, the recording studio and post production to be paid for by the the library. If you were working on an all-electronic production, it would be up to you to negotiate some money up-front for the recording costs. Some composers, however, just work on a royalty split. The more successful you are, the more likely you'd be to get some kind of retainer, but the real money in either instance comes from usage of the music.

As to how the royalties are delivered, a composer will sign a contract with the library company detailing specific tracks with titles and the agreed royalty split. All mechanical royalties generated by use of that CD will be payable to the composer from the library company at a specific time in the year — usually at the end of September. Your share of the performing royalties — which, in the main, derive from broadcasting (TV, radio, cable/satellite) — will come as a result of membership of the Performing Right Society (PRS) and will arrive between four and nine months after UK usage, and anything up to two years after foreign usage.

HOW IS THE MUSIC USED?

A lot of library music is used in post-production suites all over the country. They will be sent masses of CDs from all the different libraries, and it's not unusual to see floor-to-ceiling racks crammed with library CDs. These are sent free in the hope that an editor or producer requiring music will use them. If music from a library CD is used in a TV or video production, the producer will fill in a cue sheet/licence application form, which has to be sent to the Mechanical Copyright Protection Society (MCPS), together with a cheque at the required rate. This means thousands of forms are used every day.

Here are some examples of the UK rates per 30 seconds of music:

• Non broadcast audio-visual (corporate video):	£19
• Terrestrial TV (BBC, ITV):	£40
• Advertisements (full ITV network)	£1,173
• Radio:	£19
• Satellite TV:	£29

In addition, there are many variations, depending on the kind of use to which the music is put.

MCPS

MCPS protects and administers the copyright in production music libraries' works, and the sound recordings through which these are made available. All those who wish to use production music discs must sign the appropriate MCPS code of conduct, giving them authority to use the discs as referred to in the rate card. The code binds the user company to abide by certain reporting procedures, to ensure that the necessary licence for the use of the music is issued and the proper royalty and other fees paid. A valid licence must be obtained and the relevant fees paid before the production is transmitted, broadcast, shown, exhibited, distributed, hired, exploited or used in any way, except for use strictly necessary for the purpose of completing the production. Phew!

LIBRARY BOOKS

Library music companies can be found listed in commercial music directories such as the *Showcase International Music Book* and *The White Book International Production Directory* (both available from *SOS* Mail Order; see the 'Music Business' heading on the Mail Order pages in this issue for pricing and ordering information). Here's a few of the prominent libraries and relevant industry organisations to get you started.

• Atmosphere Music Ltd: 65 Maltings Place, Bagleys Lane, London SW6 3AR. Tel: 0171 371 5888. Fax: 0171 384 2744.
• Bruton Music: 165-167 High Road, Willesden, London NW10 2SG. Tel: 0181 967 1100. Fax: 0181 451 1307.

• Carlin Production Music: Iron Bridge House, 3 Bridge Approach, Chalk Farm, London NW1 8BD. Tel: 0171 734 3251. Fax: 0171 439 2391.
• Chappell Recorded Music Library: 165-167 High Road, Willesden, London NW10 2SG. Tel: 0181 967 0150. Fax: 0181 451 1307.
• De Wolfe Music Ltd: 80-88 Wardour Street, London W1B 3LF. Tel: 0171 439 8481. Fax: 0171 437 2744.
• KPM Music Ltd: 127 Charing Cross Road, London WC2H 0EA. Tel: 0171 412 9111. Fax: 0171 413 0061.
• MCPS (Mechanical Copyright Protection Society): Elgar House, 41 Streatham High Road, London SW16 1ER. Tel: 0181 664 4400. Fax: 0181 769 8792.
• PRS (Performing Right Society): 29-33 Berners Street, London W1P 4AA. Tel: 0171 580 5544. Fax: 0171 631 4138.

You might be wondering why people don't use the music without paying. Well, it's against the law — and the majority of people are honest.

MAKING THE CHOICE

As I've explained, there are lots of CDs out there, so how do the people who need music for their productions make a choice from all the material available? Quality will out; the music needs to be good for its application, instantly feeling right for the job it's trying to do. There's no point being obscure, or creating something that's a little bit awkward. The better it is, the more it will get used, and the more money you and the library company will make. It's useful for library music composers to check, from PRS statements, which pieces of their own material have been successful, particularly when they have been used abroad. For example, a fast harmonica/blues piece which my partner Bella and I wrote for our first library CD was picked up in Japan by NHK TV and used for a networked weekly fashion show. It seems remarkable to me that someone found it, liked it and used it! Having said that, it's rare for an editor or producer to have the time to sit and listen to a hundred CDs before they come across that perfect track. Often they build up a relationship with one particular library company over the years, and it's not uncommon for that library to be called and asked if they have anything along the lines, say, of a particular Dire Straits track, or something like the Gypsy Kings.

These people could just use the originals, but often, they cost a lot more to clear for worldwide use. Some artists, such as Elvis Costello and Peter Gabriel, feel particularly strongly that their music should not be used in commercials. Irving Berlin's catalogue of songs (including 'White Christmas') can never be used for advertisements, because he stipulated that in his estate. This protects the original material from the grimy associations of being used to sell loo paper or financial products.
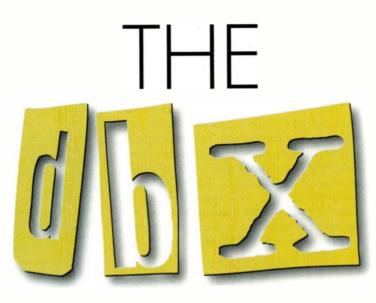
AND FINALLY...

Bella and I are currently working on a library project with a 48-piece orchestra, which is extremely interesting — if time-consuming! It's rare for media writers to have access to such a big orchestral palette, as even the better-financed TV series don't have the money for so many players. When we started writing music for the media (after being in a band together), we were just working with synths and our fingers. Now we're discussing the most effective pitch for the piccolo flutes against the first violins. No other occupation would have given me the same sort of daily satisfaction as I feel with our progression over the years through many styles of music. Writing library music has been a big part of that. **SOS**

THE dbx FILES

ROB URRY OF dbx

Now in their 26th year, American company dbx rose to fame on the back of their compressors and noise reduction systems. PAUL WHITE talks to President Rob Urry about dbx's ongoing plans to serve both the high-end and project studio analogue processor market, while also embracing the digital age.

Despite the fact that dbx celebrated their 25th anniversary last year, it's probably reasonable to say that the company has had a fairly low profile in the UK, other than for their noise reduction systems and near-legendary compressors. AKG, the Austrian microphone company, purchased dbx several years ago, and then in 1992, AKG was itself purchased by the Harman Group, and installed in the Harman Music Group centre alongside the company then known as DOD Electronics.

When I visited the dbx headquarters in Salt Lake City, Utah, I was shown a whole range of new products which were about to make their debut at the AES show in Los Angeles. Amongst these were some very high-end analogue products, but dbx are also committed to building more affordable analogue equipment, as well as some fairly esoteric digital gear. I asked the company's President, Rob Urry, to fill in the broader picture, and give us some insight into where the company is going.

ONLY dbx GET THE BLUES

Rob Urry: "This company has traditionally meant compression and noise reduction, so we're trying hard to play to those strengths, with products like our new high-end Blue Series. Lots of US touring companies have racks of dbx 160 and 165 compressors, and the idea behind the Blue Series is that it embodies a state-of-the-art implementation of what dbx stands for.

"The 160S Blue Series compressor is derived from the sound of the original 160 and 165, which were designed in the mid-'70s. The 165 was upgraded to the 165A in the early '80s, to include Peakstop, a peak blocking system for controlling transients without the harshness of conventional peak limiters. These included some very nice detector patch processing features, although the 166 and our recent 1066 [see review in *SOS* December '96] also include some new dynamic processes. The 160S is really the marriage of all those traditional and modern dbx technologies in a high-end box. You can make it sound like a 160, a 165, a 166 or a 1066. In a typical compressor, the dynamic circuitry doesn't take up that much board space, but in the 160S, there are copies of all those control circuits which get switched in and out, and the new VCA is a state-of-the-art implementation of the original Blackmer VCA. David Blackmer started the company back in 1971, and his deci-linear VCA and RMS level detector circuit are still used in current dbx products.

"The original VCA was a four-transistor log amp multiplier, but the new version is a 32-transistor version of that same topology. The idea is that as you double the number of devices, your headroom goes up by 6dB, but any uncorrelated noise goes up by only 3dB, so every time you double the number of devices, you get a 3dB increase in dynamic range. This VCA gives us 127dB dynamic range. Because the transistors have been optimised for minimum THD, they run at a slightly higher current than before, so a special aluminium-zinc package has been designed, filled ▶

▶ with a thermally-conductive resin to keep the devices at an even temperature. If this isn't done, the devices would drift in and out of spec as the temperature changed."

On top of that, you have huge output transformers for driving long lines, meticulous engineering and a very expensive-looking, blue anodised front panel. From the styling, it's obvious that you're going up against the likes of Focusrite.

"Yes, but our unit has the classic dbx

dbx's new high-end Blue Series 160S compressor.

compression sound, which has been heard by virtually everybody who's listened to the radio for any length of time. In the broader sense, the whole upper echelon of the world of compressors is our competition."

GOING DIGITAL

The Blue Series also includes a highly-specified mic amp built to the same kind of standards, but what can the customer expect to see next in the series?

"I think there are some really obvious next boxes, one of which would be a digital dynamics processor. I was going through a box of dbx patents the other day, and the circuitry that is used in our 242 Project One [see review in *SOS* January

'97] is actually very cool. The circuit was designed by Bob Orban, and is very sophisticated, but the implementation is aimed at the project studio market. It might be nice to use that design with high-end components and really good input and output stages. The great thing about our new transformer output stages is that it doesn't matter what you hook up to them, whether it's five feet of co-ax or a 1000ft snake."

Was it a conscious decision to mix digital and analogue products within the Blue range, where both have similar cosmetic styling?

"Yes, it was. My background is mainly in digital engineering, and I see some large advantages with digital, especially in the permanence of storage, complex editing, random access and so on. But I don't feel that the gap has been bridged from proper-sounding analogue to proper-sounding digital. The technology is moving very quickly, and we're getting very close. Our Type IV conversion system, which has the equivalent of 27-bit performance, has been employed in a box called the 704 (nicknamed Bad Boy), which is intended to make digital behave and sound more like analogue. Everybody in digital has been focusing on expanding the linear operating region of the devices — that's what classical engineering says you should do — but if you look carefully at the overload region and the sub-noise region, you can get a much wider perception of dynamic range for a given bit width. The Type IV starts out with a 27-bit conversion process, and then it gets reduced to either 16, 18, 20 or 24 bits."

I understand that you've introduced a non-linear region at high levels to approximate the effects of tape compression.

"The range over which non-linearity occurs is user-adjustable, and there's a very smooth transition from the linear domain to the log domain. If you were to overdrive the thing by 15dB, the output would rise by around only 5dB, so at that point, you're really getting a non- ▶

NOW WE ARE (TYPE) III

You've now announced the Type III noise reduction system, which is designed for use in systems requiring simultaneous encode and decode — for example, to reduce the circuit noise of equalisers. How does this work?

"Traditionally, dbx had Type I and II encode/decode processors used for tape noise reduction where decoding takes place at a different time to encoding. Type III was a concept that hit us sideways, and we wondered what to do with it for a little while. The principle is that the encode and decode units are physically linked, so that there are no real tracking errors — the side chain of the decoder is driven directly from the encoder block. In effect, you end up modulating noise floor below the signal, but even when the signal is very close in level to the noise floor, the perception of the noise floor is that it's very low. In fact, it looks as though we'll get a couple of patents on specifics of the process. Since the tracking error is virtually non-existent, the process can operate very quickly.

"Unlike earlier tape noise reduction systems that use a 2:1 compansion ratio, Type III uses only 1.5:1, and we've employed it in a graphic equaliser design to get around the circuit noise that the 31 filters in an EQ, say, could introduce. When you move the faders up, you're basically adding the outputs of those filters, and the noise floor can easily rise to -70dB instead of -90dB. And that's with a good EQ! As far as the user is concerned, it's just a button that is in or out. It was important to make the Type III bypassable, so people could check that this was really a very good EQ without noise reduction. We didn't want anyone to think we'd forced Type III onto them to cover up a cheap EQ. Type III is used in the 2231, 2215 and 2031 models, which also include the dbx PeakPlus limiter."

What else can you apply this process to?

"Since the encode and decode happen at the same time, you can only use the process in systems where there are no delays. EQ is a great application, but you couldn't use Type III with a delay unit or reverb device, for example."

dynamic compression of around 3:1 — although it's not like regular compression, because the individual cycles are being processed. It's actually a good approximation of what happens in magnetic recording media, but it isn't designed to exactly replicate tape saturation."

dbx-STYLE DITHERING

Down at the bottom end of the scale, analogue signals may disappear into noise, but you can hear them through it. With digital systems, low-level sounds become progressively more distorted as there are fewer bits to represent the waveform. Have you addressed this by introducing some form of dither system?

"Yes. If you employ plain truncation to reduce the bit length, it sounds pretty horrible at low levels. An extensive amount of work has been done on triangular probability density function dither — that's the one where you can add the minimum amount of noise to remove the distortion and be left with just noise. Then comes the concept of noise-shaping, where you have a certain amount of noise energy which can be pushed around in the spectrum. If you look at an A-weighted curve, you can see where the human ear is least sensitive, and push the noise there. You're most sensitive at 2-3kHz, so we put the noise where you're least sensitive. Most systems push all the noise up to around 15kHz, but our system also places some of the noise at low frequencies. This provides a significant performance advantage, but there are still problems — we don't yet have an accurate way of measuring digital systems in a way that relates what you're hearing to your experimental data. Getting the best numbers out of an audio precision test set won't necessarily mean you have something that people will judge as being the best-sounding.

"Type IV employs techniques based on random number theory and psychoacoustics to give you a white noise floor with something like 15-20dB better noise performance. Also, since we start with a wider word, you can still hear the signal as it's decaying when it's maybe 25dB below the noise floor. With a lot of digital systems, you get 5 or 6dB below the noise floor, and that's it. Our system has a lot of applications in mastering, or for when critical signals are being recorded to digital multitrack."

If you were to look at a lower-cost implementation of this concept for project studio owners, what is the smallest A/D bit-length that would make sense?

"To get pretty close to get a theoretically-perfect 16-bit signal, you probably need an additional six or seven bits. To make the best of a 16-bit CD, you really need to start your tracking using at least 20 bits. Depending on how you manage everything at mixdown, you may or may not get there. A 20-

bit system would still yield a significant improvement, and even an 18-bit system is an improvement over 16 bits. I think converter technology is progressing at the rate of a dB or two a year. Off-the-shelf 24-bit converters aren't far away, though their actual resolution is more like 18 or 19 bits in terms of signal-to-noise ratio.

"The Type IV has a very flat white noise performance, and the flagship component is the dbx A/D converter type 704. This allows you, in a graphical fashion, to change the shape of the noise floor, so you can make the noise-shaping match the instrument or sound you're tracking. It's all to do with the type of psychoacoustic masking that's going on — purists don't like to hear that, but in reality, the perceived benefits are very high."

Could you arrange the process such that the noise-shaping automatically changes to suit the spectrum of the material being processed?

> **"I don't feel the gap has been bridged from proper-sounding analogue to proper-sounding digital. The technology is moving very quickly, and we're getting very close."**

"Well, I think that could be a really cool feature — we'll think about that one! You could take an FFT and keep analysing what's going on — that's a good concept."

FUTURE PLANS

Finally, what else are you launching in the immediate future?

"Not everything is aimed at the high-end professional user. For example, we've updated our Project 1 286 microphone processor with a new, tougher chassis and an internal power supply. Now called the 286A, the unit has phantom power, an additional frequency pot in the de-esser section, and the front-panel markings have been improved. Similarly, we've upgraded the 266 dual compressor/gate to become the 266A. This now has hard/over-easy switching, +4 or -10 operating levels, and the same revised chassis and internal power supply as the 286A. We have the capacity to build some very serious equipment at affordable prices, which should be of interest to the project and home studio markets."

Rob Urry pictured with a rack containing some of the units in dbx's new Blue series.

dbx — PLUGGING INTO TDM

Rob Urry: "I should mention the *DC66* TDM plug-in, which puts dbx compression into the Pro Tools environment. Essentially, the *DC66* is based around our 1066 hardware, although the screen layout is considerably different, with very comprehensive metering for level and gain reduction. The plug-in has selectable hard- or soft-knee operation, and includes PeakStopPlus [*the updated version of PeakStop* — *Ed*]."

To the Nth Degree

KORG NS5R SYNTH MODULE

First impressions of the Korg NS5R? I'm ashamed to admit that I thought that Korg had accidentally packed a car radio when I opened its box. The size, display, soft keys and 'tuning' dial all suggested that the NS5R might be the latest in in-car entertainment. Of course 'car hi-finess' is in the eye of the beholder, but what is undeniable is that with the NS5R, Korg have made a dramatic departure from their normal styling conventions. A two-tone grey replaces the standard black livery, and a large, backlit orange liquid crystal matrix display supersedes the fiddly green effort seen on earlier modules. Inside the NS5R throbs the heart of the

to what is almost its ultimate expression in this half-rack module.

SCREEN ICON

What principally makes the NS5R more than just a sequencer-less repackaging of the latest Korg workstation (the N364) is its user interface. Nifty little icons point you in the right direction, while the 'home-page' displays a valuable amount of user-tweakable parameters, giving the unit a very straightforward and approachable feel. A refined user interface has at last become a priority for manufacturers, which is a most welcome development. As far as sound modules go, Yamaha impressed with their VL70m user interface, and Korg are working along much the same sort of lines, keeping the most oft-changed parameters never far from the surface, in an environment awash with friendly icons.

Perhaps 'more sounds for less money' is becoming a tired selling point, but CHRISTOPHER HOLDER finds some features that set Korg's latest GM sound module apart.

latest in Korg's AI^2 synthesis, packing 12Mb of PCM samples, 64-note polyphony and 32-part multitimbrality. Serious specs indeed.

By way of a precursor to this review, I admit that in the grand scheme of things I'm a comparative sapling in the high-tech recording scene. I only mention this because, fittingly, my first introduction to synthesis came via the Korg M1. I was as uninformed as any one man could be, and the M1 was a revelation, with its great sounds, great effects and on-board sequencer, all in one unit. I soon discovered its limitations — I ran out of sequencer space halfway through my third tune, and had my first encounter with note cancellation due to limited polyphony around the same time — but it still was an awesome beast. I can tell you that the spirit of the M1 (and subsequent AI synthesizers) lives on in the NS5R, but it's been refined, revised, expanded and taken

On powering up, you're greeted by four slider icons, a pan pot icon, a patch name, bank number, channel and part number, and the name of the effect program in use. Also running along the bottom of the screen is space for bargraph information on the 32 parts, nominally displaying MIDI note velocity messages. Parameter buttons allow you to move about the screen, sometimes triggering an info 'balloon' giving you the numerical status of the information provided by the slider or pot icon. The parameters shown are Volume, Expression, Pan and two effects sends. Expression is a bit of a mystery here (expanding and contracting the dynamic range of the part, I'm reliably informed) and I'd like to see it replaced with something a little more useful, perhaps a mono/poly switch. The pan pot gives the usual hard left through to hard right control, as well as providing a great 'random' option, which, as you might expect, gives you something you won't expect. The Effects sends are somewhat misleadingly called RevSend and ChoSend — misleading because they can be customised to control your own effects combination (a flanger and a delay, for instance).

In all, the screen has a reassuring newsiness, and it allows you to set up the Part and voice structure of your tune with the absolute minimum of fuss.

YOU CAN CALL ME AI

Korg have been pursuing the AI PCM-based method of synthesis for nigh-on nine years now,

and we've seen it packaged in a good number of synths, from the seminal M1 to the *Top of the Pops*-friendly Trinity. Practice makes perfect, and the sounds on the NS5R are brighter, snappier and better than ever.

The NS5R is a General MIDI module, so naturally Korg have ensured that it will be compatible with basic GM and its variants (notably GS and XG), as well as with the earlier 05R/W module. Aside from a particularly grim violin patch, the GM sounds are beyond reproach. Korg's traditional strength lies in the 'keyboard' sounds, and I can't detect any decline in this department. Where there's scope for some flair (the pads, for instance), sounds have been programmed with great width and movement.

A quick spin of the data wheel takes us into Programs and Combinations territory. There are 1177 Programs on offer, 128 of which are user-definable. I refuse to put on the 'I've seen it all' poker-face of the reviewer in this department, because this is a staggering number of sounds, and practically every one is worthy of its place.

One criticism often levelled at digital synthesizers is that they lean too heavily on built-in effects to create the impact for their presets. You know the usual routine — an overcooked patch bristling with full wraparound multi-tap delays and fierce flanging blows you away in the shop, while at home in multitimbral mode it turns out to be a cowering and pathetic bit of shortwave interference. Not so with the NS5R: all the Programs are sweetened with a sprinkling of reverb, and that's it. The presets are almost uniformly spotless, with only the odd looping surprise in some of the pads. Otherwise it's a classic Korg showcase, packed with the fascinating and useless, the classic and potentially over-used, and the unglamorous but useful. If there is any craftiness on the part of the Korg programmers, it's in the use of the double oscillator. If you use two identical or very similar samples, pan them hard left and right, and add a slight phase and tuning difference, you immediately have a sound that's intrinsically more interesting than the more mundane Single sample building block. This is more a credit to the programmers than any deception, but it has to be remembered that the 64-note polyphony is halved if you're exclusively using double-voiced Programs.

If over 1000 Programs aren't enough, you can have 512 Combinations to get your creative juices flowing (128 of these are in RAM). As many as eight Programs can make up a Combination, so naturally there's plenty of scope for interest amongst the presets. For my taste, most of the Combinations are a little too busy or dense to be particularly useful, but fortunately there are lots of exceptions.

The 32 drum sets are perfectly serviceable, with some nice orchestral samples and percussion sounds, as well as some completely bizarre special effects. Each sound in a drum set is individually editable (including individual effects levels) for added flexibility. There are 286 drum sounds to choose from, and enough memory to allow you to create two of your own drum kits.

SHIPPED TO NAVIGATE

Programs are composed of either a single waveform or two waveforms, while a Combination is comprised of as many as eight individual waveforms. No change there for Korg M-, T-, O-, and X-series owners, then. This may all be familiar territory for those people, but with the new interface they'll be flabbergasted by how easy it is to come to terms with the manner in which sounds are constructed, and how straightforward it is to manipulate and edit those sounds — it's like having an on-board user's manual. I'd love to have learned to sort out my EGs from my LFOs on the NS5R.

If there's anything calculated to cause confusion in the editing process, it's the fact that you can make rudimentary changes to a Part, as well as to a Program or Combination itself. For instance, if you have a piano sound in Part 1, you may edit that Part itself, lowering the cutoff frequency, or shortening the release time, say, but if and when you replace the piano with, say, a wind chime, the frequency content and ADSR alteration will be carried over — it's the Part that's altered, not the patch. Obviously, you have to remember what you've altered, but this facility is a valuable slant on non-destructive editing. All too often your favourite patch has its outstanding features frittered away as hundreds of minute edits are made to get it sitting correctly in the mix. No more — make the changes while keeping the original sound intact.

Program edit mode is of course altogether more sophisticated than Part edit mode, but no less easy to navigate. The initial page offers you, through self-explanatory icons, the choice of exploring the delights of either the oscillator, filter or amplifier section. One or two of the 527 PCM samples may be selected to undergo the Korg treatment. As is customary, the NS5R isn't packing a resonant filter. What it does have (which will be familiar to other Korg owners) is a parameter oddly called Color Intensity, which, far from being a new line of Dulux all-weather gloss, purports to offer something akin to resonance, but actually does precious little. You'll find that with most of the filter settings, it's a very restrained affair, refining the sample to your taste rather than completely reshaping it. It's almost like Korg saying "we've supplied you with all these painstakingly crafted samples: why do you want to go wrecking them?".

▶

HALLELUJAH CHORUS: THE EFFECTS

Two separate effects engines and 47 effects are on offer here, and by getting your head around the weird and wonderful Korg routing system and hooking up a couple of dual effects modules, you should be able to haul four different effects into action simultaneously. Korg have supplied 256 effects patches, which combine two each of the 47 effects in various permutations and at various settings, all of which you can save, name and edit yourself. This is great for creating and saving your favourite effects combinations — almost like a real effects unit! The Korg routing system utilises four separate busses (ABCD) which the signal can follow before being sent to the L and R outputs. It's well worth inspecting the manual to investigate getting the most from the effects section. One point that should be mentioned is that the NS5R comes supplied with a 'Hall reverb and Chorus effect' combination applied across all parts. This effects patch uses a parallel routing mode that has the Hall reverb set on the C buss, and the Chorus set on the D buss, so the RevSend

and the ChoSend sliders on the home page work by sending signal to the C and D buss respectively. Depending on which routing option and effects configuration you choose, these controls will not necessarily work as sliders to 'turn up the effects'. What they will do, regardless of your editing, is continue to send signal to buss C and buss D respectively, an incredibly useful facility nonetheless — but only if you're sparing a thought for which routing option you're using.

The effects themselves are very likeable. I have a real soft spot for Korg phasers and choruses. The choruses are particularly strong, with four types on offer, all adding extra width and depth without sacrificing the integrity of the original sound — particularly useful on the brass patches. Delays are well represented and the combination patches well conceived. I confess to not being a big fan of Korg reverbs — they're a little too metallic and resonant for my liking — but if Korg released an effects unit *sans* reverbs, I'd be their first customer.

Korg NS5R

▶

THE RIGHT IDEA

What with computer memory becoming cheaper and computers bursting at the seams with previously-unthinkable amounts of RAM and ROM, it should come as no surprise that synths and sound modules are loaded with thousands of sounds, heaps of raw waveforms, and enough polyphony and multitimbrality to sink an orchestra. Nevertheless, it's a credit to Korg that the quality of programming is consistently as good as it is, with as

little dead wood as possible. Having said that, there a good number of manufacturers in this market who can boast high-quality sounds, and it would be fruitless to compete on quantity alone. Here Korg have the right idea with their superlative user interface: its a joy to work with and, for me, probably the NS5R's biggest selling point.

Superficially, there are a number of things about the NS5R that suggest it might be aimed at the computer hobbyist-cum-multimedia boffin (all fancy names for computer games players). I'm all for new colours for gear, but I suspect that the choice of 'multimedia grey' came as no coincidence. Similarly, the phono inputs on the back panel are great for input-starved home studios, but even more useful for mixerless computer hobbyists — and patently not a high priority for those serious about making music. The NS5R is a half-rack module (which, again, has certain multimedia connotations) and you have to fight your way through a stack of General MIDI banks before you can even sample the delights of the 'real' instrument. These are all distractions, and shouldn't be construed as an intention by Korg to snub the semi-pro market — be in no doubt that this is a serious synthesizer. One of my few irritations is the lack of outputs (just one set of L and R jacks). How Korg can expect their customers to get any decent use out of 32-part multitimbrality and 64-note polyphony through a single L and R output astounds me, and also seems a tragic waste of a flexible and powerful effects section. But it would be ridiculous to end on a sour note, because my opinion of the NS5R is overwhelmingly positive. A powerful and versatile synth module at a wholly reasonable price. **SOS**

BOX OF TRICKS

The NS5R has a few tricks up its sleeve. As standard, Korg supply a Mac or PC format *SoundEditor* disk that allows you do your editing with a mouse, keyboard and monitor. Your computer communicates to the module via a cable connected to the serial port. It's simple enough to install and worth experimenting with. Something else of interest can be found on the back panel, in the form of stereo phono inputs. These allow you to input the signal from another keyboard, drum machine or the like, which will then be output, together with the NS5R sound, via the L and R output jacks.

Also around at the back, you'll notice a volume knob for an optional tone generator expansion board. An AG-WB Wave Blaster GM sound board will increase maximum polyphony to 96 notes, and Korg will be happy to install it for you.

Lastly, the NS5R comes equipped with an integral MIDI interface, which makes for one less worry if this module is you first foray into MIDI music-making.

info

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PAUL D LEHRMAN
discovers things he
never knew about his
recordings...

Computers make wonderful tools for music production. As recorders, editors, MIDI and studio controllers, signal processors, and sound designing systems, they have made themselves invaluable in every aspect of audio. One area, however, in which they have lagged behind has been analysis of recorded sound. Many digital audio programs allow visualisations of waveforms and some, like *Alchemy* and *Sound Designer*, have spectrum and/or three-dimensional Fast Fourier Transform displays, but these are static displays, which can only look at an instantaneous slice of sound. True real-time audio analysis, usually handled by a rack of pricey dedicated hardware, involves looking at audio signals as they are played or produced, through multi-mode level metering, instantaneous displays of the relative levels of different frequency bands within the audio spectrum, and two-dimensional displays of the relative phase of stereo signals.

This is what Intelligent Devices' *Pro Audio Analyzer* (PAA) software does, but instead of dedicated hardware, it uses a Macintosh computer, working either with the built-in audio hardware of a Power Mac, or one of Digidesign's audio cards.
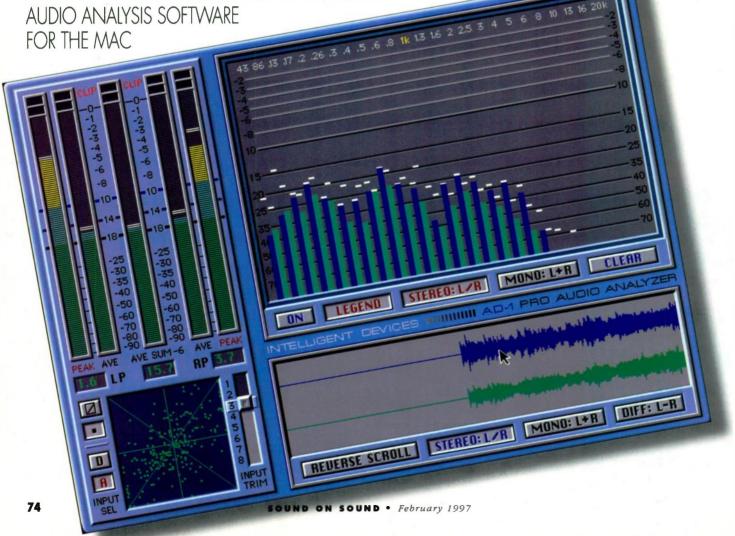
The software comes in two versions. The stand-alone version is for use with external analogue or digital audio sources, which can be brought into a Power Mac through the computer's own 16-bit 44.1kHz audio inputs (or an internal CD player), or into any NuBus Mac equipped with an Audiomedia II card or a Pro Tools I or II system. The TDM version is a plug-in for Pro Tools II or III systems (both NuBus and PCI) with at least one TDM card. It shows up as an insert, either in mono on a channel, or in stereo on an auxiliary input. Even though the processing that the plug-in version does on the signal is not as apparent as, say, a compressor, it uses up one DSP.

THE FEATURES

The program's display is in four sections, which can be viewed all together, or in various subsets.

• The leftmost part of the display has five vertical level meters: vertical peak and average (VU) indicators for each stereo channel, in green and yellow, with a pair of red clip 'lights' at the top of each one; and a central meter which can be switched to show the average or peak levels of either the sum of the two channels (with a 6dB pad, which keeps it more or less at the same level as the individual channel meters), or the difference between them. The meters can be made to look like LEDs (large segments), plasma displays (small segments), or continuous displays. They cover a range of 96dB, which means they're almost always showing something — input noise, convertor noise, hum — unless you're playing a signal that consists ▶

Test Case

INTELLIGENT DEVICES *PRO AUDIO ANALYZER* 2.0
AUDIO ANALYSIS SOFTWARE
FOR THE MAC

"This description of *PAA*'s features really doesn't do justice to the great usefulness of the product."

of mathematical silence. Above the coloured segments are small white bars for peak hold indication. The release rate for the meters and the decay rate for the peak hold indicators are separately adjustable over a range of four speeds. An input trim control adjusts the input signal in eight steps over about a 20dB range. The steps are uncalibrated, but the 'zero' position is clearly marked.

The two clip lights atop each meter — one goes on at the first sign of clipping and stays on until you reset it, the other resets itself after a while — can be set to turn on anywhere from 1dB to 0.05dB below maximum level, so if you want to be daring, you can try to push your levels really, really close to the max. Below the meters are numeric peak indicators, which show, to the nearest 0.1dB, the highest level encountered since they were last reset.
• Over to the right of the level meters is the spectrum display, which divides the audio range from 43Hz to 20kHz into 24 bands, each approximately 1/3-octave. Individual vertical bars show the instantaneous level in each band — either the two halves of a stereo signal side by side, or the summed mono signal. A switchable reference 'legend' shows heights from -70 to -2dB, and the gain of the display can be raised 6 or 12dB. Peak hold indicators are here too. Their decay time, the display's update time, and the main release time are all adjustable. Using a slow update and decay rate can give you a good picture of the overall spectral balance of a recording over time — very useful when mixing.
• Below the spectrum is the waveform monitor, which will be familiar to anyone who's used a digital audio editor. It scrolls the waveform as the audio plays, and can show two channels in parallel, summed mono, or the difference signal (L-R), at any of three speeds, or reverse. In addition, any points at which the signal clips are displayed in red and stay that way, showing the clipping history of a piece over a fairly long period of time.
• Finally, there's the phase scope. This generates dots representing the instantaneous stereo position of the signal, against either a vertical or diagonal axis. The dots can be either of two sizes, and there are three magnification scales. A 'phosphor emulation' can be switched on that causes the dots to die away slowly, and a 'length' feature controls the overall time a dot stays on screen — at the longest length, they turn into little snake-like things. An overlaying grid can be switched on and off.

The update speed is not directly controllable, and unfortunately it's not very fast. I found this feature to be the least useful of the bunch: it can give you an idea of the overall stereo spread of a mix, and also show you when things are seriously out of whack, but if you're looking for the source of a balance or phase problem, the slow update speed makes it less than ideal. Apparently, this display is given lowest priority of the four, and although there's a slight speed improvement when all of the others are switched off, it's still no substitute for a real oscilloscope set up to monitor L vs R.

You can hear the audio that is passing through *PAA*, and control it via a dialogue box accessed with option-clicks or, in the stand-alone version, on a menu. You can select stereo, mono, or the difference signal, or put the sum on the left and the difference on the right. There's also a 20dB pad and a mute switch.

THE GOOD, THE BAD & THE UGLY

This description of *PAA*'s features really doesn't do justice to the great usefulness of the product. Simply by allowing you to look at music, while you're hearing it, in such a wide variety of ways and from such different angles, it can teach you a tremendous amount about how to mix music as you play favourite (or unfavourite) CDs through it, and watch the levels and spectral balances over time. When it comes to mixing your own music, by dint of being faster and more critical than just about anything else out there (certainly anything that costs less than several thousand dollars), *PAA* makes it easy for you to maximise your levels, and thus get the best dynamic range out of digital recording without encountering the unconscionable horrors of clipping. If you paired *PAA* with a good digital dynamics processor, like Waves' *L1 Ultramaximizer*, you'd have an unbeatable combination for making mixes that scream without screeching.

Although *PAA* is highly recommended for those who are serious about knowing what they're putting on tape or CD (and who have a fast Macintosh), there are a few ways in which it could be improved. The owner's manual, written by Intelligent Devices president (and audio columnist) Stephen St Croix, has a tutorial that reads like a magazine piece: breezy, clever, with plenty of amusing asides, and also informative and thought-provoking. Unfortunately, that's all there is: there are no instructions on what you need to run the software, or how to set up your hardware or the Mac's treacherous System Folder. For seasoned Pro Tools users, this is not too much of a problem, but for new Power Mac owners, who want to see what's going on with their CD collections or their own mixes, it's pretty off-putting. Neither is there a reference section, or even a picture describing what everything does. There is a list of keyboard commands, but it leaves out some of the most important ones, which you have to find by digging through the dense text. And the number of spelling mistakes is pretty embarrassing too.

Some less-visible features are clumsy and don't work too well. There are command keys for setting many of the functions, which can be reconfigured by the user. Unfortunately, you can't just change one command key: if you want to change any of them, you have to change all of them, and the reconfiguration window doesn't show the default assignments, so you have to remember what they are (not all of them are in the manual) and enter them all in by hand. To make matters much worse, when it is in reconfiguration mode, the TDM version responds incorrectly to the Mac keyboard: when you tell it to assign a function to 'c', it thinks you're typing 'option-shift-6'. The choices of

command-key defaults also leave something to be desired: the space bar turns the spectral display on and off, but in Pro Tools, it happens to be the main transport control. When *PAA* is open, the key's transport function is disabled!

The manual tells you that if you put *PAA*'s window in the background on a Pro Tools screen, it won't work well, but it doesn't mention that you can't. This is actually a problem with all TDM plug-ins, but it's particularly annoying because *PAA*'s window is so big. (There's no re-sizing of the window *per se*, although you can choose not to display some sections.) If you have a small monitor and want to do things like operate Pro Tools' transport controls, or bring *PAA* in and out of an insert point in the Mix window, you have to do a lot of sliding things around.

In a strange violation of Macintosh interface guidelines, there's no title bar in the stand-alone version — to move the window, you have to hold down the command key while you click and drag it. Another anomaly of the stand-alone version is that you cannot shut down your Mac when it is running; you have to specifically quit the application first.

THE VERDICT

Rough edges aside — and at least some of these will undoubtedly be cleaned up in future revisions

— *Pro Audio Analyzer* is a marvellous tool for anyone serious about audio. If you work with sound, you'll find a use for it: studio designers will use it to tune rooms, live sound engineers will look for hot spots and feedback points in performance spaces, mastering engineers will spot anomalies and imaging problems, broadcasters will make sure their signals are providing maximum punch; the list goes on. You may be tempted, as company president St Croix says he was, to spend hours looking at other people's recordings to see why they sound good — or not so good. It would be time well spent: with any luck, you'd learn to apply the principles thus discovered (the good ones, that is) to your own recordings, which should stand you in very good stead in the long run. **SOS**

NEATER METERS

The manual makes a strong point of saying that *Pro Audio Analyzer* is far more accurate and reliable than any other digital metering system because it will show a clip as short as a single sample, whereas other systems (like the meters on a DAT deck) might pass several clipped samples in a row before they acknowledge the problem. I was able to fool *PAA* sometimes, sending it signals just below clipping that caused the red lights to fire, and sending it single-sample clips that didn't, but it got it right nearly all of the time. The usefulness of this feature was borne out when I fed the system a sine wave that just clipped at the very top: *PAA*'s red lights blinked furiously, while a Tascam DA30 MkII DAT deck was placidly telling me I had 3dB of headroom left.

Reality Bytes

MAKING MIDI SEQUENCING MORE REALISTIC

Is it real or is it synthesized? PAUL FARRER passes on some tips to help you inject some real feel into your MIDI programming.

Morphing a kazoo into a male voice choir, or mapping the pysically-modelled shape of an orchestral hit onto the parameters of a sample of your granny knitting, may well be the kind of tricks that equipment manufacturers like to dazzle us with. But one thing that we continually demand from sequencers, sound modules and samplers is their ability to sound more 'real'. For the kind of work I do, the majority of my time is taken up with trying to encourage my 19-inch, rack-mounted box of circuitry, SIMMs and SCSI boards (otherwise known as a sampler) to talk to my over-stuffed source of many compatibility headaches (otherwise known as a PC) in such a way that the result will sound just like a 40-piece symphony orchestra, or a guitar trio, or maybe even just a rock 'n' roll band. A fruitless waste of time, or an exciting programmer's challenge? Read on...

DIFFERENT STROKES

One of the first things to bear in mind when thinking about MIDI instruments is that they are absolutely nothing like the real musical instruments they're pretending to be. The player/instrument relationship between a violinist and their violin, and the keyboardist playing a violin sample via MIDI from a keyboard, is so wildly different that it would seem silly to pretend otherwise. Both require discipline and musicality to coax the best sound possible from each instrument, but there the similarity ends. Once we acknowledge and respect this fundamental difference, I believe the process of trying to emulate real instruments via MIDI will become much easier.

PIANO FORTE

Preset number 1 in nearly every synth and sound module since the year dot has been the piano. This must be the most duplicated, sampled and synthesized sound in MIDI history, and yet many of us still find it hard to make what we programme into our sequencers sound like a real pianist. The main trick is to try to utilise all the MIDI controllers that are relevant to the particular sound you are trying to re-create:

• **PEDAL POWER:** The number of people who still try to play a MIDI keyboard without a sustain pedal is astonishing. This is the main expression controller that a piano has, so to try and program piano parts without one is a sure-fire way to make your performances seem flat and one-dimensional. As MIDI controllers go, sustain pedals are both versatile and extremely cheap (I picked mine up for a fiver from a bargain-bin at a music store!), but it's worth remembering that pianos are not the only thing to benefit from the sustain pedal. Long, sustained synth pads work very well if you play a chord, push the sustain pedal, then move your fingers to the next chord (whilst the notes are still playing); then, just before you play the second chord, release the pedal. Repeat this process as many times as you need to and this produces an almost seamlessly legato chord sequence. Don't worry if you've never used a pedal before — you'll be surprised at how easy it is to pick up, and once you've used one, you'll wonder how you ever managed without it.

• **PUMP UP THE VOLUME:** The volume pedal is another valuable piece of MIDI gear that is often overlooked. Swelling brass or string parts is great fun with a pedal, and this saves hours of fiddling around in a sequencer's real-time MIDI Generator page getting the crescendo you want. Bear in mind that, particularly within an orchestra, the ability to increase and decrease volume (often very quickly) is one of the keys to a good arrangement. Although pop music generally has less dynamic variations, the volume pedal can be an invaluable tool, particularly for live work, where a high degree of control is needed but you don't have time to fiddle with countless edit screens.

IMPERFECT PERFECTION

No instrumentalist (no matter how good they are) plays exactly in time or in tune. There will always be tiny imperfections and inaccuracies in every performance, and that is what gives a piece its emotional content. One clever way of adding these imperfections artificially is to try and picture in your head not only the kind of instrument you are emulating, but also the player and all the relevant playing styles. General MIDI has given us a couple of pointers in this respect, adding 'Fret Noise' and 'Breath Noise' to the list of GM sounds. But why stop there? If you're recording an acoustic guitar through a conventional microphone, there are loads of unwanted scrapes, string buzzes and fingerboard thumps that, whilst being the bane of the recording engineer, can be the salvation of the MIDI programmer. I recently sampled a selection of these thumps and scrapes and dropped them at random places throughout a sequenced nylon guitar part. Providing they aren't too loud, the effect can be quite dramatic. I even managed to fool a guitarist friend of mine into thinking I'd ▶

Realistic MIDI Sequencing

got a real player in to record the part (the ultimate benchmark of a programmer's success!).

The same approach could be adopted for all sorts of instrument sounds. Breath noises are an obvious way of giving a touch of authenticity to wind instrument parts, but it's worth looking at the other factors involved in a realistic saxophone, clarinet or flute solo line. Physical modelling has

Figure 1.

given us a big boost in this respect, and anyone who has used Yamaha's VL series of synths or a Korg Prophecy will tell you just how realistic many of their woodwind sounds are. But for those of us without access to the latest generation of hardware, pretty good results can be obtained from a standard synth module, as long as you keep in mind a few simple rules of thumb.

• **A CASE OF THE BENDS**: One of the most common forms of expression in any instrument is its ability to 'bend' the pitch of the notes it plays. With wind instruments, the most common form of pitch bending is a form of portamento, such as

when a saxophone or trumpet bends up to reach its note. Players usually start between a tone and a semitone below the note they're going for and, depending on how jazzy they're feeling, often start a fraction ahead of the beat as well. This is very often coupled with an increase in volume as the 'true' note arrives. Working with MIDI, this can be duplicated with a combination of pitch-bending (plus or minus two semitones), and use of the expression pedal and/or MIDI volume level adjustments (see Figure 1). A good way to program these is to record the part using only the pitch wheel, then listen back to it and decide what other expressive elements could be added. Move to a separate record track (on the same MIDI channel) then add other features, such as modulation or volume changes. Once you're happy with these, simply merge the two tracks together. Of course, if you can get away without having to quantise a solo of this kind, all the better, but if not, there's always the option of quantising more difficult sections and leaving the rest untouched.

• **ALL MOD CONS**: Moving onto modulation, I've noticed that a good number of manufacturers' factory preset sounds seem to come with the modulation wheel set firmly on the task of modulating the pitch of a sound, rather than modulating its amplitude. If you think about it, and certainly with wind instruments, vibrato is achieved through a subtle mixture of both slight changes in pitch and variations in volume. If you're programming a flute sound, for example, it's worth listening to a real flautist to see just how they use vibrato. Very often the vibrato doesn't appear until quite a way into the played note, and therefore modulation that you programme from the start of the note will always sound unnatural. Also it's important to decide just how much vibrato you need, and of what kind (amplitude or pitch). This is where a volume pedal could come in handy for programming the ▶

QUANTISE QUESTIONS

The actual notes you play are surely the most important element to any performance, so what if your playing isn't up to it? This is where quantising can be a great help. We're always being told that quantisation kills expression, but there are many situations in which it can be both time-saving and creative. Quantisation is a tool, and like any other musical tool, if used correctly it can produce great results. The basic quantise levels of early sequence packages have now been replaced by much more 'intelligent' quantising systems. Emagic's *Logic* is a good example of this, but later versions of *Creator* and *Notator* were also well-suited to the less robotic 'groove' style of timing correction.

It's worth noting that within any given piece, certain sounds will always quantise better than others. For instance, bass lines and bass drums seem the obvious things to quantise first. They are the foundation upon which the rest of the track rests, and timing errors here will almost certainly give the whole song an unstable or sloppy feel.

Sounds with longer attack times (such as synth pads) seem to quantise less well. You'll probably find that when you play a string line with a slow attack time, you automatically play a little ahead of the beat to compensate for the lateness of the note. This is fine, but if you then quantise this line, the notes start exactly on the beat and therefore sound a little *behind* the beat. You can cure this to some extent by quantising the whole track, then either using a negative delay or, on the edit pages of your sequencer, selecting all the relevant notes and dragging them forward slightly until they sound right. Be warned, though: if the first note of a line edited in

this way starts at the very beginning of the song or pattern (bar 1, beat 1) it often 'drops off' the front of a song and can disappear altogether, so in this case you would have to start editing from the second note in the line.

Some software packages offer the happy compromise of first quantising a MIDI performance, then 'humanising' it by scanning through the inputted data and adding small timing inaccuracies. Although this sounds good in practice, I've never really managed to get a huge degree of success using it, and I can't help feeling that this feature is an ironic and somewhat 'soulless' approach to the whole business. If something isn't sounding right, don't immediately leap for the quantise icon: why not slow the track down a little or try to record it in easy-to-manage sections? Like many things in life, the quickest and easiest answer is not always the best.

Realistic MIDI Sequencing

diminuendos and crescendos so crucial to the expressive element of any wind instrument performance.

BRASSED OFF

Of all the sounds that your average synth module can produce, surely the most shockingly artificial of them all has to be the 'Brass Section' preset. They all stink, and nearly every module has within it the innate capability to give any sequenced track that 'Children's BBC' feel, simply by playing a few choice lines on what the manual describes as "...a rich multisample featuring all the instruments of a contemporary brass section...". One of the main reasons for this inherent cheesiness could stem from the fact that a brass 'section' is something of a contradiction in terms. There are as many different configurations for a brass section as there are records made using one, but the traditional line-up usually consists of at least one of each of the following: trumpet, trombone, and saxophone. Depending on the kind of music you're recording, the line-up will change accordingly, and it's not unusual for a trumpet player to also double on flugelhorn, and the saxophonist to also play all four types of sax (soprano, alto, tenor and baritone) — not forgetting the all-important French horn, which, although mainly a 'classical' instrument, is often seen lurking within the ranks of the brass section even for pop songs these days. When you consider this collection of instruments (and all their amazingly versatile playing styles) you can begin to see just how surprising it is that any synth patch can claim to even try to re-create their sonic range in a single preset.

The trick when programming any brass section is to view each instrument separately. Most sound modules give you a good number of the individual sounds as solo patches (Sax section, Trumpet, Single Trombone, and so on) and by assigning one to each of the parts in your multitimbral set, you not only have a greater degree of control over the individual sounds (particularly with reference to pan positions, volume levels and tunings), you actually have a much more accurate representation of what a brass section actually consists of (see 'Brass Tack' box for a suggested brass section setup). Programming the parts, of course, takes a little more time, in that it

requires you to think a little more about what each individual part is doing, but you'll probably find that working in this way encourages you to be a bit more creative with your brass parts. And should you need to get live players in to replace any of the lines for real — in a live situation, for example — the process of transforming what has been programmed into notes that the players can work with is speeded up greatly.

STRUNG OUT

When programming any orchestral string parts, as when programming brass parts, you should always try to think of the orchestra as a collection of individual instrumentalists, as opposed to simply a collection of different instrument sections.

We've all got our favourite string preset or multisample, and when we play a big G minor chord they sound great, but this is quite different from the workings of a real string orchestra. Violins, cellos and violas normally only play monophonically, and the arranger's job is to allocate each note of a chord to a particular section — for example, cellos could play the root (A), violas could play the third (C#), and the violins might play the fifth (E). The resulting sound is a complete chord of A major, and the richness and depth of any arrangement is in the way in which these elements interact and combine throughout a piece. Again, as with the brass parts, it can take a bit more time to program string parts in this way, and it also requires a greater knowledge of the chord structures you're using, but the end result is nearly always richer, more interesting and undoubtedly more realistic than if you had simply 'strummed' the chords polyphonically using your keyboard.

If, however, you feel your playing really isn't up to the job, or you don't have the time, you can always cheat a little by taking a chord track and using the command within your sequencer software that extracts the top and bottom lines of any given chord track. You can then assign these individual monophonic lines to different tracks, and therefore different sounds. For example, drop the lowest note of the chord by an octave (12 semitones) and assign it to a cello sound, then perhaps take the top line, make a copy of it, raised by one octave, and assign both tracks to the same violin sound. This certainly 'widens' the sound of a chord and is much more in keeping with a real string ensemble.

When you're programming these individual single string lines, you may find that there are slight gaps between the notes you've played (caused by releasing the keys a bit early, in order to move to the next note). This can be cured by using your sequencer's 'Force Legato' command. This 'stretches' the programmed notes within a given section and triggers MIDI note off information only fractionally before the start of the next note. The effect is to 'glue' the notes together without any gaps, and can often be very effective.

If the string part is still feeling a touch hesitant, you can always try going into your edit pages and creating a slight overlap between the individual ►

BRASS TACK: BUILDING A BRASS SECTION IN A MULTITIMBRAL SETUP

PART	INSTRUMENT	VOLUME	PAN (-64 to +64)
1	Solo Trumpet	100	-10
2	Solo Trombone	90	-40
3	Tenor Sax	115	+30
4	Alto Sax	90	+15
5	Baritone Sax	75	+50
6	French Horn	100	+00
7	Flugel Horn	95	-20
8	Trumpet 2	90	-15

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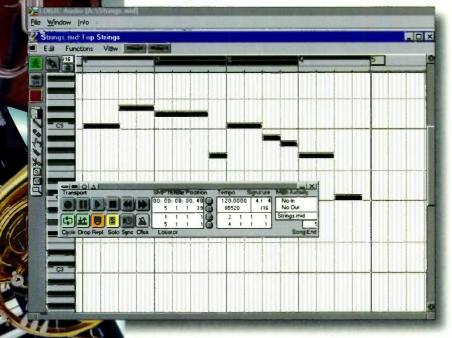
Realistic MIDI Sequencing

▶ notes, so that each one plays slightly into the next (see Figure 2).

Another thing you might like to try, particularly if you're limited by polyphony or 'timbrality', is using a string preset with a fairly fast attack time. This helps to ensure timing accuracy. With most samplers and sound modules, it's a relatively simple procedure to programme the attack time to be proportional to

Figure 2.

key velocity. This gives you a good amount of flexibility for playing both soft 'legato' phrases (at a lower velocity) and sharper or 'spiccato' lines when playing the keyboard a bit harder.

STAR GUITAR

The success of the guitar (arguably the world's most-played instrument) lies in its sheer versatility, range of sounds, and expressive potential. But it's these very qualities that make it so difficult to recreate realistically via MIDI.

If you'd like to try anyway, the first thing to think about is what kind of guitar sound you'll be working with. Whether it's a nylon, steel-strung acoustic, or electric will have as much bearing on the playing style as it does on the final sound. Most sound modules have a brave stab at nylon and steel-strung acoustic sounds, with varying degrees of success, but despite the fact that a distorted guitar patch is on the General MIDI list, I've yet to hear one that sounds like anything other than a pair of wasps in an aluminium jam-jar. The key to guitar realism mainly rests in the choice of sounds, so sampled guitars seem to be the only way to approach the problem seriously. Sample CDs such as the *Steve Stevens' Guitar Collection* offer us some interesting options, namely splitting the guitar into a number of playing styles (lead sounds, power chords, strums, etc) and letting us work with those. Certainly, a distorted set of guitar power chords in fifths work

very well to lend body to the average rock track, and (just like a real guitar) expression, through pitch-bending up to the note, modulation and volume changes, all help to add to the realism.

Picked acoustic guitars can be a little trickier, as they generally sit slightly louder in a track and are usually more exposed. The key here is to try to get away from the idea of programming the sound in the same way that you would play a keyboard. The tunings of the guitar strings are such that a standard G major chord on a piano is very different to a standard G major chord played on a guitar. For example, a simple G major guitar chord in MIDI-speak looks something like this:

G2 B3 D3 G3 B4 G4

Not many people would play a keyboard chord in this way, but if you're to emulate a guitar successfully, these are the kind of things to think about. Timings are also crucial; generally the plectrum hits the lower notes first, so really any guitar chord is nothing more than a very quick arpeggio, starting from the lowest note up. The faster the track, the closer together these notes will be, obviously.

Another favourite form of expression amongst guitarists is to bend certain notes within a passage. Using traditional MIDI pitch-bending techniques, this would mean bending every single note being played on that particular sound at the time, which gives the impression of a 'whammy-bar' as opposed to a fingered bend. A good way around this problem is to assign more than one of the MIDI channels in your multitimbral setup to the same sound. In this case, if MIDI channels 1 and 2 are both hosting the same acoustic guitar preset, the majority of the work can be done on channel 1, and any detailed picking involving bending single notes can be recorded using channel 2. As the two channels are independent of each other, bends for the solo parts on channel 2 do not affect the pitch of notes on channel 1. The added benefit of working in this way is that you can also easily control volume and any other performance data (such as modulation, detuning, expression, and so on) of a solo part using the same sound. As I've already mentioned, any amount of 'sampled guitar ambience', such as string squeaks and buzzes, that you can throw in will all help with the overall sound.

External processing can also often be a godsend for giving that all-important bite to even the tackiest and thinnest of synth sounds. Many of today's samplers and modules come with tasty built-in effects processors, but these can generally only deal with one or possibly two effects at a time, so if your sound source has extra outputs, it may well be time to brush off that old guitar pre-amp or fuzz box and let rip with some unconventional MIDI guitar sounds of your own. Try starting with a clean electric guitar sound, assign it to a separate output, and route it through an external effects processor, in much the same way as you would if you were working with a real guitar. The results may surprise you. **SOS**

MIDI + AUDIO
on the PC

THREE TOP PC PROGRAMS COMPARED

- STEINBERG *CUBASE*
- EMAGIC *LOGIC AUDIO*
- CAKEWALK MUSIC SOFTWARE *CAKEWALK PRO AUDIO*

Last year saw the proliferation of reliable, fully-featured MIDI + Audio sequencers for the IBM-compatible PC — packages which were previously the preserve of the Apple Macintosh platform. JANET HARNIMAN COOK takes a look at the three top PC programs in the field to see how their facilities compare.

In retrospect, 1996 should be remembered as the year when PC MIDI + Audio sequencing came of age — a dream come true for the PC-based studio musician. Now, with the aid of these high-quality, low-cost applications, PC musicians can record and edit multiport MIDI and multitrack digital audio side by side on a single screen. The pace of development shows no sign of slowing down, either: faster Pentium processors and the new generation of Windows 95 MIDI + Audio applications should continue to revolutionise modern music production, both in the home and in the professional studio. In this article — designed to mirror Mike Collins' overview of Macintosh MIDI + Audio sequencers, which appeared in *SOS* December '95 — I will be giving a brief overview of the capabilities of three PC MIDI + Audio sequencers — Steinberg's *Cubase*, Cakewalk Music Software's *Cakewalk Pro Audio* and Emagic's *Logic Audio*. These three applications are the current cream of the crop, and their advanced features will no doubt define those we will see in the PC music systems of the near future.

IN BRIEF

• STEINBERG *CUBASE*

Cubase is the best-known and most widely-used sequencer in Europe, and has been the powerhouse behind innumerable chart, soundtrack and dance successes throughout the '90s. Originally written for the Atari ST, the leading edge of *Cubase* development is now on the Apple Macintosh. *Cubase Audio* for PC initially only supported the Yamaha CBX HDR system until early 1996, when version 1.6 for Windows soundcards was released. Unlike *Cakewalk*, which exists in just one form, *Cubase* is marketed as a trio of specialised packages. All the *Cubase v3* range offer the same MIDI + Audio recording and editing facilities as the 'vanilla' *Cubase* package (£329). Enhanced, advanced DTP manuscript origination facilities are available in *Cubase Score* (£499) and these are complemented by support for Digidesign Session 8 and Yamaha CBX in *Cubase Audio XT* (£699).

• EMAGIC *LOGIC AUDIO*

Logic Audio is the best-selling Macintosh MIDI + Audio sequencer throughout Europe, and is a favourite of many of the recording elite (for example programmer *extraordinaire* Pete Gleadall, interviewed in *SOS* December '96). *Logic Audio v2.5* for Windows 95 entered the market in summer 1996, and the latest version (2.5.3) costs £399. The recently-released *Logic Audio Discovery* (£269) is essentially the same as version 2.5.3, but has fewer score and time/pitch compression-expansion facilities.

• CAKEWALK MUSIC SOFTWARE *CAKEWALK PRO AUDIO*

The first MIDI + Audio sequencer to use PC soundcards was *Cakewalk Pro Audio v.4*. *Cakewalk* originated on the PC in the dark days of DOS, and its successor, *Cakewalk for Windows v3.1*, was the first program to be capable of true commercial-quality MIDI recording on the PC. The current version, *Cakewalk Pro Audio v5* (£329), is a 32-bit native Windows 95 application. In the USA, *Cakewalk Pro Audio* is the market leader in MIDI + Audio sequencing on the PC.

CHOOSING A PACKAGE

All three of these applications are excellent modern music recording tools. Hardware mismatches aside, such strengths and failings as they exhibit lie in the detail rather than in overall function. Which one you choose will depend to a great extent on your individual preferences and needs. If you have external hard disk recording hardware, such as Soundscape, Digidesign Session 8 and Yamaha CBX systems, you will be limited by the current support provided by each application. Of the three, only *Cakewalk Pro Audio v5* supports all three systems; *Cubase Audio XT* supports Session 8 and CBX. *Logic Audio* does not yet support any of them, but I am assured

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REVIEW SETUP & SYSTEM REQUIREMENTS

The reference PC is an Intel Pentium 100 with 256K pipeline burst cache, 32Mb of RAM, 2.2Gb EIDE hard drive space, Turtle Beach Multisound Classic & Pinnacle digital soundcards, and a 2Mb Trio+ PCI graphics card.

A note of caution: quoted minimum PC system requirements may be misleading. There is a world of difference between using a minimum-spec PC that stutters and crawls along, and a large, fast, well-configured machine. For the sake of your sanity, I recommend using at least a Pentium 100 with 24Mb of RAM, and a fast 1Gb EIDE or SCSI hard drive for MIDI +

Audio applications. This said, to add a couple of tracks of audio to 10 of MIDI, you should get by with a 486 DX266 with 16Mb and PCI buss. A 17-inch or larger monitor should ideally be used to accommodate the detailed multi-screen environment favoured by MIDI + Audio applications. To optimise the PC for audio, it is essential to observe good working practices, and the most important of these is to keep your hard disk in pristine condition by defragmenting the drive after each session. For more on this, plus details of crash prevention and recovery regimes, see my article on surviving PC crashes in *SOS* December '96.
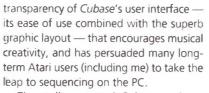
that Emagic are in the process of developing extensions for CBX, Soundscape and the Akai DR8. All three MIDI + Audio applications support the Digidesign Audiomedia III card. Multiple Windows soundcards are not (yet) supported, and solo soundcard support is surprisingly uneven, with *Cakewalk Pro Audio* again proving the most accommodating. A further consideration is that if you have used a specific MIDI sequencer, getting to know the MIDI + Audio version will require far less time than learning an unfamiliar package from scratch.

This article will take a look at how well this complex and innovative technology works in practice. The points of comparison under consideration are as follows:

• Hardware Compatibility
• Ease of Learning (manuals, help, tutorials)
• Quality of User Interface (look, feel and ease of use)
• Global Control (song structure editing, tempo control, mixing, automation)
• MIDI Editing
• Audio (including recording, editing and DSP functions)

STEINBERG *CUBASE* v3.03
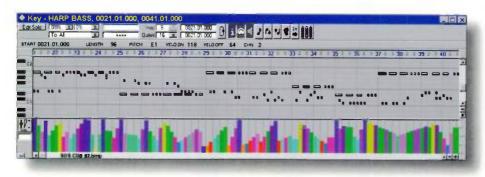
• FEATURES & COMPATIBILITY
Cubase is not yet available in true 32-bit format, and version 3.03 is a 16-bit application for Windows 3.1 and Windows 95. *Cubase v3.03* provides up to eight stereo tracks of digital audio and 64 tracks of MIDI. Audio multitrack capability is determined by the specification of the PC used to run the application. *Cubase* played

transparency of *Cubase*'s user interface — its ease of use combined with the superb graphic layout — that encourages musical creativity, and has persuaded many long-term Atari users (including me) to take the leap to sequencing on the PC.
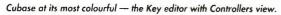
The well-presented *Cubase* package contains a CD-ROM, the copy protection dongle and four manuals. The manuals (entitled *Getting Started*, *Getting into the Details*, *Modules* and *Audio*) are clearly written, with good indexes, and contain around 500 pages. An additional booklet

The layout is analogous to that of a multitrack tape recorder: tracks are arranged vertically and contain MIDI data displayed as parts across the horizontal time axis.

If you are gazing at a screen for hours, a well thought-out, colourful graphic environment can help reduce fatigue, and *Cubase* is the most graphically attractive of all three packages featured in this article. Colour is generally used well, but some screens are still very grey, and have changed little since the old Atari versions. Notepad

Cubase v3.03 in action, playing back eight stereo audio and 11 MIDI tracks.

facilities are rudimentary, with basic cut and paste editing, and text import from the Windows clipboard.

• GLOBAL CONTROL
Cubase is object- and menu-driven. Maximum resolution is 384 pulses per quarter note, or ppqn (1/1536 whole note) with options for 96 and 192 ppqn. Most edit and file activities can be done on the fly, and

Cubase at its most colourful — the Key editor with Controllers view.

eight stereo tracks on both the Multisound Classic and Pinnacle cards, and recording proved straightforward for both cards. When changing between analogue and digital on the Pinnacle, it is best to configure the card before altering *Cubase* audio hardware settings.

• EASE OF LEARNING
Cubase is very user-friendly, and beginners should achieve results quickly. This is a powerful piece of software, and is fast and enjoyable to use. Most frequently-used facilities are readily accessible, yet beneath the surface is extraordinary depth. It is the

with advanced score techniques is included in the *Cubase Score* and *Cubase Audio XT* packages. The CD-ROM features an extensive range of high-quality multimedia tutorials and the installation software for *Cubase* and *WaveLab Lite*, its companion audio editor.

• USER INTERFACE
The first version of *Cubase* appeared in 1989, and defined the page layout adopted by *Logic*, *Cakewalk* and most modern sequencers. The Arrange window is the main *Cubase* work space, and displays the overall structure of your piece.

Drag and Drop editing is common to all graphic views. In practice, I have found a combination of keyboard commands and mouse to be the fastest means of getting around. In *Logic Audio* and *Cakewalk*, global parameter definition is common to the whole track. But in *Cubase*, the constituent sequences of each track have their own Part Inspector. This Inspector enables basic instrument parameters to be quickly set up, and includes velocity, pan, offsets and dynamics for each part. This is fabulous, but it would be even better if effects and user-defined controllers were ▶

MIDI + AUDIO
on the PC

▶ included. Installing the Studio module adds instrument patch names to the Inspector, and provides simple voice editing for some common synths in the way that Steinberg *Satellite* did on the Atari ST.

When building a piece, you often need to change the order of the verses, chorus, breaks and bridges until the structure feels right. *Cubase* provides a good range of cut, paste and copy functions to make song structure editing painless. And as each song may contain as many Arrange windows as you need, you have the freedom to quickly switch between multiple versions. Oddly, *Cubase* does not have markers. The locator sets are easy to program, but they only recall position, and have no text association. So to distinguish song sections in the Arrange page, there is little choice but to create a dummy track and use its part names as marker labels. I would like to see a Markers area that displays section names or cues, as is found in both *Logic* and *Cakewalk*.

The MIDI Mixer is a user-configurable virtual multitrack mixer, and provides the means to automate your mix and to record MIDI controller changes in real time (although audio ▶

TABLE OF COMPARATIVE FEATURES

APPLICATION	CUBASE	LOGIC AUDIO v2.5.3	CAKEWALK PRO AUDIO
Price	From £329	£399	£339/£379
Audio Tracks	8 stereo	8 or 16	Unlimited
MIDI tracks	64	Unlimited	256
Score Editor?	Yes	Yes	Yes
Piano Roll Editor?	Yes (Key editor)	Yes (Matrix editor)	Yes
Event List Editor?	Yes	Yes	Yes
Drum Editor?	Yes	Yes (Hyper editor)	Yes
MIDI Controller Editor?	In Key editor	Yes (Hyper editor)	Yes
Chord recognition?	In Key editor	No	No
Lyrics allowed?	Only in Score editor	Only in Score editor	Yes
SysEx Editor?	in Event editor	In Event List editor	Yes
Tempo Editor?	Graphic & List	Graphic & List	Graphic
Audio Editor?	Yes	Yes	Yes
Wave Pool	Yes	Yes (Audio editor)	No
Sample Editor?	Off-line	Yes	Yes (Audio editor)
Audio recorded as WAV files?	Yes	Yes	No
On-line Help?	Yes	No	Yes
Extensive tutorials?	Yes — multimedia	No	Yes — multimedia
Notepad?	Yes	No	Yes
Copy Protection	Dongle	Dongle	No
AVI support	Yes	No	No
Windows 3.1 support	Yes	No	Yes
Native Windows 95	No	Yes	Yes
MMC Control	Yes	Yes	Yes
Ships as	CD-ROM	Four floppy disks	CD-ROM & floppies
DSP FUNCTIONS:			
Normalise	Yes	Yes	Yes
DC Offset correction	Yes	Yes	No
Time-stretch/Pitch-shift	Yes	Yes	No
EQ	No	No	Yes
Audio Quantise	Yes	Yes	No
Audio to MIDI	No	Yes	Yes
Limiter	No	Yes	No
Dynamics	External	No	No
Sample rate conversion	External	Yes	No
Auto Region creation	No	Yes	No
Noise Reduction	No	Yes	No
Fades/Crossfades	Yes	Yes	Yes
Define MIDI Tempo by Audio	No	Yes	No
HARDWARE SUPPORT:			
Yamaha CBX?	Cubase Audio XT only	No	Yes
Digidesign Session 8?	Cubase Audio XT only	No	Yes
Digidesign Audiomedia III?	Yes	Yes	Yes
Soundscape?	No	No	Yes
Akai DR8?	No	No	No

The 1066
Our Next Industry Standard

dbx 1066 Compressor/Limiter/Gate

Look at the equipment list of any large studio or live concert system and chances are you'll find **dbx** products. Studio and live sound engineers rely on dbx to deliver silky smooth vocals and crisp, tight drums and bass which are the **dbx** trademark.

In building a successor to our early classics we talked to hundreds of working engineers and producers about the way they work and what's important to them in a dynamics processor.

The result is our new 1066 Compressor/Limiter/Gate. Classic **dbx** sound, an almost fanatical level of build-quality and the most comprehensive specification of any processor in its class.

A Solid Foundation

The heart of any dynamics processor is its VCA and **dbx** really scores in this department. Our proprietary VCA's are good enough to be used in famous name mixing consoles as well as our own products, and the 1066 features our best VCA yet. The new V2™ boasts superb dynamic range while maintaining very low distortion and almost immeasurable noise.

Classic DBX features at your fingertips

Our original *OverEasy®* soft-knee compression is switchable to hard-knee for that almost special-effects sound.

The Smartest Sidechain

The 1066 makes using external devices in the sidechain simplicity itself. *Monitor* lets you instantly flip between the sidechain and the compressed signal for rapid set-up whilst the *External* switch allows you to A/B processed and unprocessed signal without the need for complex patchbay normalisation.

PeakStopPlus™:

is **dbx**'s latest limiting algorithm and the best yet. It suppresses those unwanted transients while minimising the distortion common to many other hard limiters.

Add to this numerous other switching options, a ratio control on the gate and metering of input level, output level and gain reduction and you have the most comprehensive set off controls available.

Hand built in the USA.

All **dbx** effects are hand built in our factories and scrupulously tested before shipping. Components are meticulously selected and the 1066 is no exception. It features illuminated switches, detented pots and new super bright lightpipe technology which replaces standard LEDs found in other units.

We think the 1066 is a worthy successor to our original compressors. See and hear one for yourself and see if it sets your standards.

MIDI+AUDIO
on the PC

▶ events cannot be entered from the mixer). Configuring the mixer is rather long-winded, even though Steinberg have included a selection of mix templates.

The *Cubase* tempo and time signature editor is the Master Track. Tempo changes can be viewed as a list or graphically, and there are sophisticated tempo scaling features. Tracks can also be time-locked: with this feature, tracks may run at different tempi even whilst crossfading. Events on locked tracks remain at the same time position and stay in sync regardless of the tempo changes affecting MIDI — useful for voice-over and sound effect placement.

• MIDI EDITING

Cubase has excellent MIDI control, and its editors are rich in information. MIDI can be edited as a table of events in the List editor, or as standard notation in the Score editor. *Cubase Score* has the best scoring and

manuscript editing features found in any sequencer, and will produce professional, DTP-ready results.

The Key editor displays notes on a piano roll grid with MIDI controller information graphically displayed beneath. Only a single type of controller information can be displayed, although it is possible to open several Key edit screens of the same part, with each showing a different controller. In practice, this is not a great hindrance, but an option showing multiple controllers (like *Logic*'s Hyper editor) would be a more elegant solution. However, colour is used well in *Cubase*'s Key editor; colour sets can be assigned to channel, velocity, and pitch events.

Audio and drum parts have specialised editors in *Cubase*, and changes based on mathematical criteria can be made in the Logical editor. The *Cubase* Drum editor is probably the best on the planet, and it is easy to build your own super kit using mapped drum voices from many instruments. It is a pity that a maximum of nine bars can be displayed without losing the drum voice list. There is also a useful General MIDI editor, which caters for Roland GS and Yamaha XG extended formats.
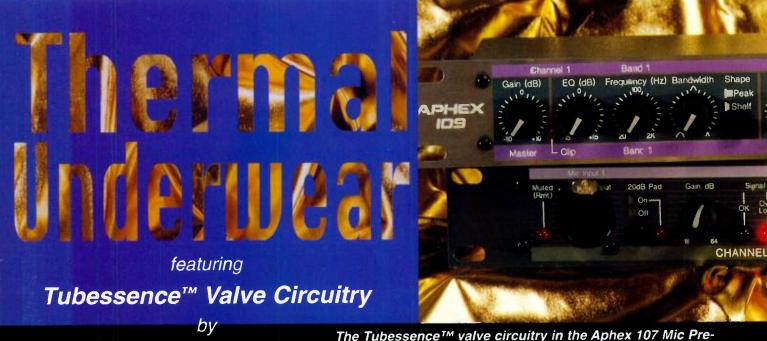
The IPS (or Interactive Phrase Synthesizer) is an interactive super arpeggiator and a useful composition aid. The Styles module is more conventional, and provides auto-accompaniment type features. On those days when the ideas aren't flowing, Styles may do the trick.

• AUDIO

Audio recording is straightforward. Tracks can include mono and stereo parts but all parts in a song share a common sample

rate. When recording multiple takes, *Cubase* automatically creates and names a new soundfile for each take. Parts are edited on the Arrange page in much the same way as MIDI, and double-clicking on an audio part opens the Audio Editor. If the Track is set to 'Any', all audio channels can be viewed together. Audio is displayed as a waveform and each event has Start and End Inset handles that can be used to quickly trim audio — a quick fix which is handy for simple cut and paste edits. Volume and pan curves can be drawn in the Audio editor. Channels are monophonic, and events with overlapping times cause channel-defeating if set to play back on the same channel. This means that the first audio event will play through to its end point, and until playback stops, any events starting later will be muted.

Cubase does not have an onboard sample-accurate graphical audio editor, but includes a link to a designated external wave editor. *Cubase* is bundled with *WaveLab Lite*, but can be used with any other editor, such as the full version of *WaveLab* or *Sound Forge*. A serious complaint is the lack of track sequence EDL (Edit Decision List): there is no simple method of viewing the sequence of audio events in each Track, and audio events are curiously not included in the List editor.

The Audio Pool is the sound file manager, and contains a list of the root sound files and the derived regions or segments that form the play sequences for each audio Track. Sound files can be imported into the Audio Pool in PC WAV or Mac AIFF formats, and segments are added to the Arrange window by drag and drop.

The Audio Pool also contains basic wave editing facilities, including Normalise, Reverse, Varispeed, Erase Unused (condense sound file) and resample/time domain functions.

EMAGIC *LOGIC AUDIO* v2.5.3

• FEATURES & COMPATIBILITY

Logic Audio v2.5.3 is a Windows 95 application, and will not run on Windows 3.1n. *Logic Audio* can be very, very fussy about Windows soundcards, so it's advisable to check with both Emagic and your soundcard importer to confirm compatibility. Certain mice, including those by Logitec, are incompatible when used in-line with the *Logic* copy protection dongle and should be moved to a free PC Comm port. I'm also informed that certain PC motherboards and/or chipsets may not be compatible with *Logic Audio* systems that rely on Digidesign's Audiomedia III card.

Logic Audio offers unlimited MIDI tracks, and eight or 16 audio tracks, depending on your recording hardware. Only a single sample rate can be used for each song, and stereo recordings occupy two tracks; stereo files imported from CD-ROM or hard drive are split and re-recorded to two discrete new soundfiles, which stay

Logic Audio's Arrange page.

together automatically.

The Multisound Classic soundcard in the reference PC played and recorded eight audio tracks, and *Logic Audio* recognised the analogue and digital audio ports on the Pinnacle card. Both soundcards' MIDI ports and the MQX32-m MIDI interface used in the reference PC appeared automatically in the *Logic* ports list, and I experienced none of the problems mentioned by Paul White in his recent review (see *SOS* November 1996). The only major problem was that the pickup and release of audio tracks in the Arrange page was delayed, sometimes by up to four seconds! This was at its worst in cycle mode, and made accurate positioning of audio very difficult indeed; I think the cause

must be a buffers problem. Emagic technical support was unable to help, but offered the consolation that no similar problems have been reported with Audiomedia III. Similarly, in Hyper Draw, there was a 2.4-second offset between audio volume controller entries and their implementation.

• EASE OF LEARNING

Newcomers may initially find *Logic Audio* rather daunting, but it is a wonderfully-equipped and very classy application with most of the commonly-used functions very close to hand. For example, transpose, quantise, looping, copying and moving can all be done in the Arrange page without having to sift through menus. The bulky ring-bound 600-page manual includes an ►
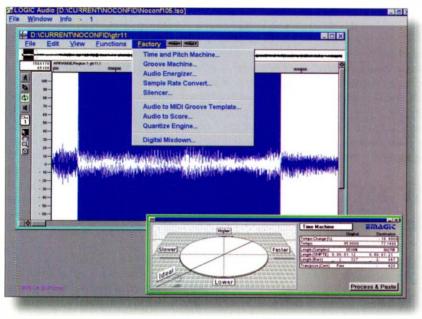
Logic Audio's Sample editor, with the Time Machine and Factory menu.

tempo editing, including custom tempo curve template creation and tempo scaling.

• MIDI EDITING

MIDI handling in *Logic Audio* is good, and editing is fast and enjoyable — if *Logic*'s Edit windows seem sparsely featured, they still work well. *Logic* has a range of editors, offering a number of event perspectives similar to those in *Cubase* and *Cakewalk*.

Logic's graphical editors make very little use of colour, and it is sometimes difficult to read event status information, especially in the Hyper editor window. The Hyper editor displays multiple controller types simultaneously, and is preferable to the single-controller displays found in *Cubase* and *Cakewalk*.

Controller data can be drawn directly onto sequences, as a series of envelope points, in the Arrange window using the Hyper Draw function. Hyper Edit also acts as the main *Logic Audio* drum editor, which performs well enough, but does not possess the versatility of its *Cubase* counterpart. Notation editing and printing is also not quite as well-featured as *Cubase Score*, but has the edge on *Cakewalk*.

• AUDIO

Audio data can easily be cut, pasted, copied, deleted and even mixed from *Logic*'s Arrange page, and most of the MIDI editing tools also apply to audio. For example, a region may be divided using the Scissors tool. Audio regions may also be looped, and there's an automatic procedure for making the tempo of your MIDI data precisely match the length of an audio region. Volume and Pan information for each audio track may be recorded as a Hyper Draw sequence, though on an adjacent track rather than on the same one, but strangely, you can only switch the record status on and off by visiting the audio section in the Environment — this is ▶

▶ audio section, and it is for the most part comprehensive, well indexed and clearly laid out, but the absence of Windows on-line Help, the bareness of the factory default setup, and the apparent complexity of *Logic*'s Environment create a less than friendly welcome. If ever a program needed a CD-ROM multimedia demo, this is it!

• USER INTERFACE

Logic has a pleasant, slightly quirky feel to it, but is very stable, and inspires confidence because of its low crash rate. Screens are generally clear and well laid out, although more colour would be welcome, as would better fonts. Screen redraws are usually fast, but sometimes proved sluggish in the Sample editor.

Getting around *Logic* is quick and easy, with excellent markers that carry through to the editors (*Cakewalk* markers are only visible on the Arrange page, and *Cubase*, as we have seen, does not implement markers at all). To help you get up to speed, a huge range of user-definable keyboard and MIDI remotes are included. *Logic Audio* also wins plenty of brownie points for its innovative screensets, which allow you to store and recall up to 90 different screen layouts. This is essential if you have a small screen, and because *Logic* allows you to open the same window more than once, you could create a screenset showing the Arrange overview of the whole song, with another Arrange window zoomed-in on a 4-bar section. Clever linking means that open windows can all stay in sync with each other if required.

• GLOBAL CONTROL

Common activities, like saving to disk and editing, can be carried out in real time,

with the MIDI Resolution fixed at 960ppqn. *Logic Audio* does not address the ports of your MIDI interface directly from the Arrange page, but does so through virtual instruments in the Environment. These act like junction boxes, and define the MIDI signal path between the MIDI Out data in track sequences and the PC's physical MIDI Out ports. Multi-instruments also store patch names which may be imported from text files, or typed in. Ready-made Multi-instruments, complete with patch names, are supplied on the support disk for the more common instruments.

Virtual Environment objects also include *Logic*'s internal MIDI effects modules (such as delay lines, arpeggiators and chord memorisers). Virtual faders may be used to control individual devices, or can be grouped to form an automated virtual mixing console. The Environment also provides the only access to instrument patch names; you can't automatically load in patch names from your synths. Like many other users, I found *Logic*'s Environment rather badly explained in the manual, and although the concept is simple, its apparent complexity could cause confusion for the novice.

Most activity in *Logic* is focused on the Arrange screen, where each track has its own MIDI activity meter. The Track List contains track names, MIDI activity indicators and user-selectable instrument icons. Instrument and patch names are available from the Arrange page, providing you have created the appropriate Multi-instruments in the Environment.

Song structure editing in *Logic Audio* is fast, and uses the Windows cut and paste conventions. *Logic Audio* also has superb

MIDI + AUDIO
on the PC

▶ one function that really should be accessible from the Arrange page.

The Audio editor displays all the wave files used in the song, complete with constituent regions, length, anchor points and sample rate information. The Strip Silence function may be used to scan audio material, extract pauses and automatically create new regions, which is useful when separating different takes from a recording. *Logic Audio* also contains its own sample editor, where audio may be edited to single-sample accuracy. Unfortunately, it is not possible to scrub through digital audio material and locate exact cue points using a Windows soundcard — if you require this, you must invest in a Digidesign Audiomedia III PCI card.

The Sample editor also contains the Factory, where a selection of very tasty DSP audio processing tools reside. The Audio Energiser uses a smart limiter algorithm to raise the perceived overall level of the audio without introducing distortion — I used it on a complete mix and the results were excellent, with the processed audio sounding fuller and more punchy. The Silencer noise reduction module consists of two processors which may be used independently or together: Spike reduction reduces impulse disturbances such as clicks, while Noise reduction reduces hiss and other unwanted broad-bandwidth noise. Time-stretch and Pitch-shifting are processed by the Time Machine, and are capable of very good results if used with discretion. Other snazzy goodies include the Groove Machine, which lets you alter
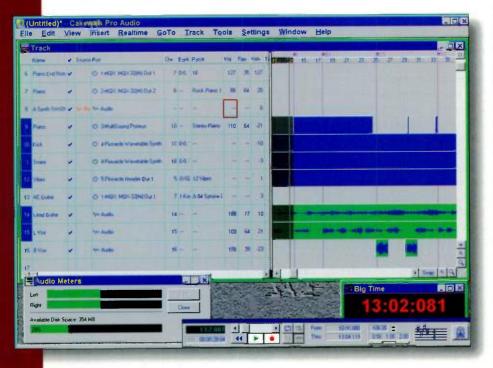
the swing or groove of the audio material, and the Audio-to-MIDI groove template function, which creates a MIDI groove template from rhythmic audio material. The Audio-to-Score Streamer will take a monophonic audio recording and derive a MIDI sequence from it, while the Quantise Engine allows you to apply a groove template to an audio sequence. This latter feature is very impressive, and isn't restricted to simple drum parts.

CAKEWALK MUSIC SOFTWARE
CAKEWALK PRO AUDIO v5

• FEATURES & COMPATIBILITY

Cakewalk Pro Audio ships on CD-ROM with version 5 for Windows 95 and version 4.5 for 16-bit Windows 3.1. There are 256 MIDI tracks and the number of audio tracks is limited only by the speed of the PC. A common sample rate must be used throughout a song. *Cakewalk Pro Audio* is also the least expensive of the three MIDI + Audio applications under scrutiny here, but offers the widest support for internal and audio hardware, including support for the Soundscape hard disk recording system.

• EASE OF LEARNING

Top marks! Cakewalk Music Software have made great efforts in *Cakewalk Pro Audio v5* to help the new user get started. The *Cakewalk* CD-ROM includes a useful upbeat 7-part *LotusCam* multimedia tutorial that demonstrates the basic recording and editing techniques for MIDI and audio. The printed manual is conveniently sized and contains 402 pages: the information within is well presented, comprehensively indexed, and easy to read. This aid to getting started is further supplemented by very good on-line Help and Windows Tooltips. *Cakewalk Pro Audio* is easy to approach and would be an ideal introduction to MIDI + Audio sequencing.

• USER INTERFACE
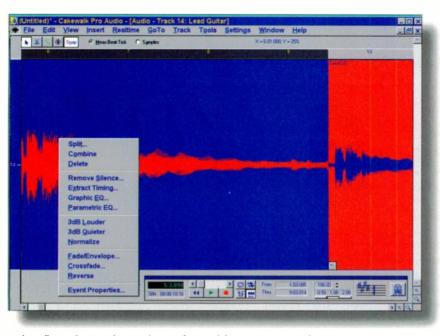
Cakewalk has a well laid-out graphic interface and a good range of customisation options. The main Tracks page would not win prizes for its aesthetic merits, but it is clear and highly informative. *Cakewalk Pro Audio* does not hide basic instrument parameters in Inspector boxes, as do *Logic Audio* and *Cubase*. Both of these applications only permit track and patch names, MIDI parameters, port assignments, offsets and so on to be viewed one track at a time. In *Cakewalk* all of these parameters are visible at a glance from the Tracks page. Patch names derive from Instrument Definitions — an onboard database which lets you select any voice, from any MIDI instrument in your rig, by name or program number. To help you get set up quickly, *Cakewalk Pro Audio* contains a large list of patch names for most common instruments. You can also easily create custom banks with the names of your own patches from within *Cakewalk* or by editing the instrument's INI file from your word processor. Setting up the record and playback parameters for each track can also be carried out from here.

• GLOBAL CONTROL

Cakewalk offers a choice of timing resolution, from 48 to 480ppqn, and features a 128-level Undo/Redo facility with an Edit History list. Both *Logic* and *Cubase* feature list and graphical editors for tempo editing, but *Cakewalk* only has a graphical tempo editor. This is great for drawing in curves, but a list-based tempo editor is often preferable, as it enables greater editing precision and can be quicker. Nor does *Cakewalk* have an auto-quantise option to quantise on the fly while recording MIDI — a shame, as such an option can save time when recording drum parts from a MIDI keyboard, for example.

Mix automation is the domain of the Faders view, and this is where you find the virtual faders, knobs and switches that make up the *Cakewalk* virtual mixer.

Audio recording in Cakewalk.

Cakewalk's Audio view showing the waveform and the Inspector menu edit options.

Whereas creating a Mixer page in *Cubase* usually means building mixer objects from scratch, in *Cakewalk*'s Fader view they are automatically generated. You can edit the Fader objects, but it is hard to prevent the Faders view from appearing cluttered.

• MIDI EDITING

Most of *Cakewalk*'s graphical editing screens are attractively designed, and generally its MIDI editing options are good, but the program does not have the depth and sophistication of features that both *Cubase* and *Logic* enjoy: the Drum editor, for example, lacks the facility to map drum voices from different sources. Controller editing also has one important shortcoming: events are displayed according to the channel on which they were originally recorded. Of course, parts created by cutting and pasting between tracks may have controllers from many different channels, so the lack of a View All Channels function can make editing difficult. There is also no way to remap MIDI controllers in real time, to enable you to enter MIDI volume messages from a keyboard modulation wheel, for example.

Staff view is a good basic score editor, and will display and print up to 24 staves with song lyrics and traditional expression and dynamics markings. Curiously, the Event List editor does not include an event filter to aid selection or exclude defined types of MIDI event from the edit move. The System Exclusive view is well specified, with a 256-bank SysEx librarian, and provides a good way of saving and loading your MIDI equipment configurations.

• AUDIO

Audio recording is simple and quick, and this makes Cakewalk a wonderful, easy-to-use songwriters' sketch pad — you don't even have to name your audio files, unless you want to export them. Unfortunately, *Cakewalk* does not save its audio as ordinary WAV files, but uses a proprietary format instead. So if you want to process *Cakewalk* audio files in a dedicated editor like *Sound Forge* or *WaveLab*, you must first convert back to the Windows WAV format and then re-import the file back into your *Cakewalk* song after you have finished processing it. More cohesive audio editing is hinted at, as the *Cakewalk* Tools menu contains an as-yet-unimplemented link to *Sound Forge*. The Audio editor can display multiple tracks, and permits audio scrubbing. Supplementing the usual cut and paste options, there are simulated DSP tools, including graphic and parametric EQ, a noise gate, and normalisation, crossfade and fade envelope editing options. During playback, MIDI volume and pan controller messages can be used to control audio events, but real-time MIDI control needs a fast PC processor. The forthcoming Waves *Native Audio Pack* for Windows 95 includes *Cakewalk* extensions, and a joint project is underway between Microsoft and Cakewalk Music Software to create a Windows audio plug-in specification. 〔SOS〕
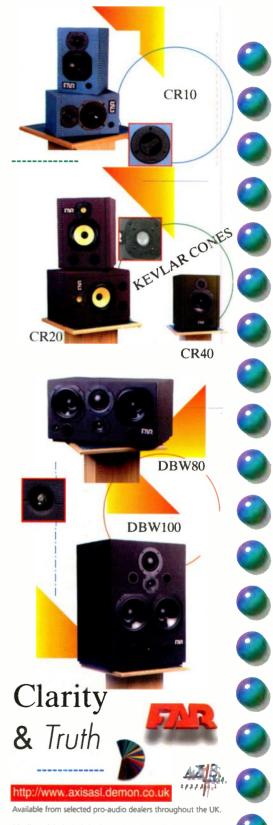
Many thanks to Mark 'Pixie' Ward, my original Cubase *guru.*

Sound à la carte

GUILLEMOT MAXI SOUND 64 PC SOUNDCARD

The Maxi Sound 64 is a new Plug and Play full-duplex soundcard with some pretty tasty features, cooked up by French company Guillemot International. **MARTIN WALKER** *tucks in.*

It must be hard for any new manufacturer of soundcards to get a look in when the mainstream market is already dominated by the likes of Creative Labs' Soundblaster range, which firmly occupies the middle ground, with low-cost, anonymous clones beneath and expensive high-spec cards with high-quality D/A converters and digital I/O at the dizzy heights. In the middle and low-cost areas, compatibility with games is still one of the most important features, and this normally means SoundBlaster. This area of compatibility does tend to prevent mainstream acceptance of more revolutionary products. Gravis found it extremely hard with their Ultrasound card, which, despite having excellent features, initially had a poor, software-only Soundblaster emulation. Despite sterling efforts to encourage the computer games industry to support the unique features of their card, it seemed to be the single factor that slowed initial sales.

Guillemot International have taken a different approach. Rather than trying to market a revolutionary product, they've made a card which falls firmly into the 'evolutionary' camp. There's nothing startlingly new on offer, but nearly every feature has had its specification carried one stage further than the territory mapped out by the SoundBlaster AWE32 card, which seems to be the yardstick until the new AWE64 range appears. Particular emphasis has been placed on sound quality, and for the musician, one of the most interesting features will be the ability for playback of up to eight simultaneous WAV files whilst recording (with full duplex operation). Even the forthcoming AWE64 cards from Creative Labs don't have hardware duplex capability (see Paul White's interview with Creative Labs in the December 1996 issue of SOS).

Guillemot International are a French company that I have not come across before, but they've already done extremely well across the rest of ▶
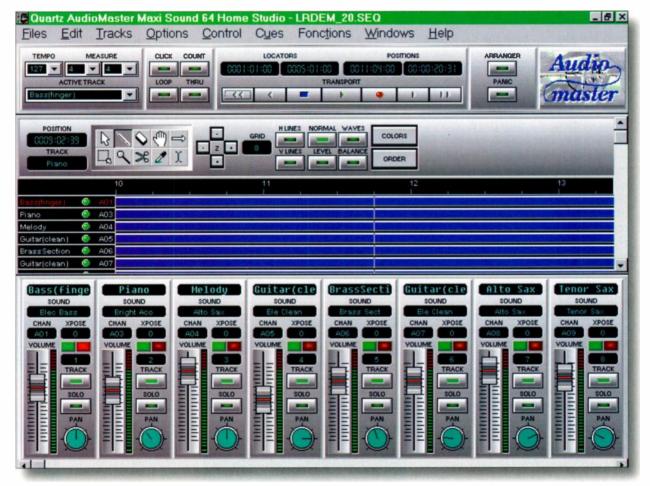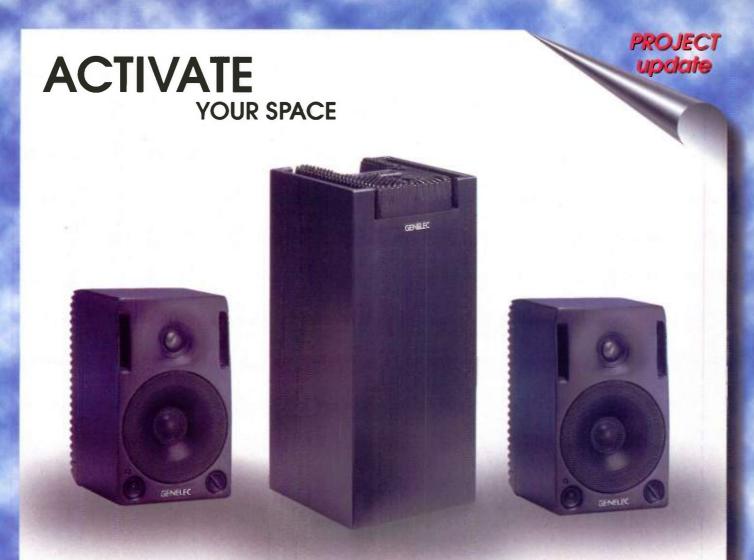
Figure 1: Quartz Audiomaster — all you need for 4 (or 8) track audio and 32-track MIDI recording.

Maxi Sound 64

▶ Europe with their Maxi Sound range of PC soundcards, which includes a daughterboard featuring Korg-based wavetable synthesis. The UK operation is handled by Ubi Soft, who are well known from their many published titles in the computer game world, as well as more recently for various music software and hardware releases. Et Cetera also look set to include Guillemot products in their range. The Maxi Sound 64 card is quoted as being the first sound card available with 64-voice polyphony, true 4-channel surround sound, 4-band equaliser, superior-quality wavetable sounds in 4Mb ROM, and "real-time special effects never before achievable on a PC". The Home Studio version also offers a genuine four to eight audio tracks of direct-to-disk record and playback, with real-time effects.

CRYSTAL CLEAR

The core of the Maxi 64 card is a CS4236 single-chip Multimedia Audio Subsystem from Crystal Semiconductor, which has been available for about a year. This offers complete Plug and Play compatibility, full-duplex operation, 16-bit, CD-quality sound and FM music synthesis for PC motherboard and adaptor card applications. For a soundcard manufacturer, the beauty of using a single-chip solution like this is that their overall design gets onto the market faster, costs are lower, so savings can be passed on to the customer. Another advantage is that Crystal supply a complete suite of drivers supporting all Windows operating systems, so any driver updates are also taken care of at source.

This very chip can now be found incorporated into the motherboards of Toshiba's latest range of notebook computers, as well as a new range of Dell desktop PCs. More recent chips from Crystal Semiconductor include versions with licensed QSound 3D audio enhancement and Dolby Digital surround technology, so let's hope that these other devices find their way into more soundcards soon.

WAVE AFTER WAVE OF SOUNDS

Guillemot have provided a full complement of extras on the Maxi. The wavetable synth fills a generous 4Mb of ROM, and provides a highly-satisfactory 64-voice polyphony with a total of 425 sounds, comprising a GM/GS set, variations, and 200 percussion sounds, divided up into 16 drum kits. For comparison, both the AWE32 and 64 cards have 1Mb ROM. The Maxi sounds are pretty good, although the current yardstick which most musicians will use to judge them by is the Yamaha DB50XG, and they're not quite up to this standard —though they're certainly not far behind. The individual sounds are clean and crisp, although some of the multisamples are too short to allow much character to develop in certain voices. Many parameters can be edited via NRPN (Non-Registered Parameter Numbers), for sound customisation. A bit of chorus fattens up the sounds nicely, and for general multimedia use the overall effect is impressive. The synth is 16-part multitimbral, which is standard these days, but the 64-voice polyphony should make note-stealing a thing of the past for most people. A single socket is provided for a SIMM module — either a 4Mb or 16Mb 32-bit type, which should cost you about £20 or £70 respectively. This memory allows you to download your own sounds to the card. Although the fact that the Maxi doesn't provide any memory as standard may seem limiting, it does help to keep the initial cost of the card down, and this feature is basically in line with the Soundfonts feature of the SoundBlaster range.

The WAV file portion of the card will probably interest musicians particularly. The A/D and D/A converters were chosen for their sound quality, and the card caters for replay rates of between 4kHz and 44.1kHz, with full-duplex capability. This is vital for modern hard disk recording. (Half-duplex cards can be used, but only if you run one for recording and a second for playback.) Hardware full-duplex operation should result in less processor overhead than by using a special software driver, and this may well provide more audio tracks for the same speed of computer. The technical spec of the card (frequency response, signal-to-noise ratio, and so on) is described as "exceptional", but since no figures are quoted anywhere in the manual, I can only say that everything sounded very good to my ears. I can understand some manufacturers' reluctance to quote figures when the interference from every PC will influence the results to some extent, and some people's figures can be extremely optimistic because of this, but it would have been nice to see something for reference.

There's a socket for a daughterboard, such as the well-loved Yamaha DB50XG, and my DB50XG sounds emerged from the other side of the soundcard remarkably unscathed compared with my hot-rodded direct outputs (see my article elsewhere in this issue on making the most of daughterboards), so circuit quality is certainly high, and noise lower than some other soundcards at similar prices. ▶

Maxi Sound 64

▶ There are mic and line inputs for the WAV section, as well as three different internal sockets for CD audio connection, and a standard IDE CD-ROM interface. A 4W stereo amplifier feeds the speaker outputs, which also double as line
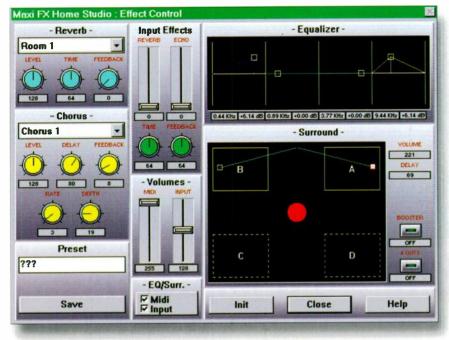
Figure 2: We've got you surrounded! The Maxi FX utility provides full control for every output.

outputs once a couple of jumpers are changed — using this option and feeding the outputs at line level into an external mixer will give much better audio quality. A second pair of outputs is provided for quadraphonic surround use, and this will no doubt sound wonderful with games. To comp ete the array of inputs and outputs, there is a fully-implemented MPU401-compatible MIDI interface with one input and one output, but,

unlike many cards, the MIDI output used to drive the daughterboard socket is entirely separate from this, allowing 16 channels of MIDI to be used independently on each. A joystick socket is also attached to the MIDI flying lead with its two DIN plugs, and it's nice to see this lead included in the package, rather than being an optional extra.

Using part of the on-board DSP (Digital Signal Processor) capacity, up to four WAV files (each in stereo) can be replayed simultaneously (eight after a minimum of 4Mb has been added to the sample RAM). This is an excellent idea, especially as this feature can be used by any standard MIDI + Audio sequencer, since each track features as a separate WAV output as far as Windows is concerned. Playing back multiple files in this way can already be done in software using any of the MIDI + Audio sequencers and a Pentium machine, but Guillemot's hardware method should give a much lower processor overhead, and this saving could be used with something like *Wavelab* 1.5 to give more real-time effects, or enable more tracks.

DSP EFFECTS

An intriguing set of effect features are also provided by the onboard DSP, including surround sound, reverb, chorus, and 4-band EQ. The DSP allows a selection of effects to be applied to both WAV and MIDI sounds, including those from any attached daughterboard. All operate in real time, and parameter changes are heard immediately the mouse button is released. Reverb and Chorus types are set up globally, and then their levels can be set individually for each MIDI channel. What's more, each WAV output can also have its reverb and chorus levels set independently, and can have a top-cut filter applied to it (obviously using the dynamic noise filter that helps to achieve the low-noise output). The chorus works quite well, and ▶

PLUGGING AWAY: INSTALLING THE MAXI SOUND 64

Although I know the theory of Plug and Play well, this was my first hands-on experience of a Plug and Play peripheral, so I was interested to see how my Windows 95-equipped, but non-PnP motherboard, computer would cope. The only initial problem was of my own making — I decided to remove an existing soundcard to make way for the Maxi, but neglected to also remove the associated drivers using the Control Panel. When I re-booted the computer, the screen stayed black, and I had to remove the card, re-boot again to enter Windows, then remove the drivers and start again. This time I was pleasantly surprised. Despite being basically a halfway measure, the PnP component of Windows 95 correctly recognised that a piece of PnP hardware had been added, and automatically put up a message that it was "Building driver information database." It then prompted me to insert the supplied floppy disk containing the Maxi Sound 64 Windows 95 drivers, and then even noticed that I'd already attached my Yamaha DB50XG daughterboard and so installed a driver for that as well. The only small point that would have made the installation perfect would have been a reminder to remove the floppy before prompting me

to restart the machine. I left the floppy in as a test and sure enough, on restarting, the computer tried to boot from this, and a French error message appeared, advising me to remove it and press any key. Apart from that, it was plain sailing, and all seemed uncomfortably too easy.

After installing the soundcard and restarting Windows 95, I found that installation had indeed been too easy to be true. Although nothing to do with the Maxi, my still-installed Gravis Ultrasound card was obviously trying to use one of the same resources, so its driver was summarily disabled, leaving only the Maxi to carry on. Also the MIDI port of the Maxi conflicted with my Roland MPU401 interface, again pre-PnP. This is the sort of problem that can arise when PnP and any legacy (pre-PnP) hardware is mixed within the same machine.

To remedy the situation, I first of all got out my printed list of currently-used resources (every home should have one!) to see where the conflicts lay. The tricky part was that the Maxi has such a comprehensive set of functions that it requires three IRQs, two DMAs, and a staggering seven I/O addresses. However, since there are 12 possible

choices for most of these addresses, and they are automatically allocated by the Windows 95 PnP, this is hardly a problem in itself. The conflict lay with the 16-bit Windows 3.1 drivers of the Ultrasound card, which are not automatically recognised by Windows 95. One of the three IRQs selected for the Maxi for its own use was IRQ11, and this was in my list as already being used by the Ultrasound card. By checking in the Advanced section of the Multimedia part of the Control Panel, where the selections for the Ultrasound driver lay, I found that IRQ15 (still available on my machine) could be used for the Ultrasound, so I selected this and then restarted Windows, to let this new setting take effect. This resolved the problem with the Ultrasound. Similarly, the Roland MPU401 interface was moved out of the way of the Maxi, and then everything seemed to work happily together.

Once you've tried PnP, having a peripheral that can self-adjust seems wonderful. Unfortunately, it's nearly always the legacy devices that gum up the works, and in some cases it's easier to disable the 'Automatic Settings' box for a particular PnP driver, so that you can force it to use the only IRQ you know you have left — even though it would prefer to use one already occupied by legacy hardware which Windows 95 refuses to acknowledge.

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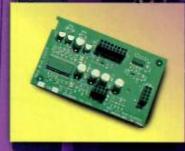
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▶ offers control over five parameters. The reverb comprises three room settings, two halls, one plate, and two delays. These algorithms are extremely basic, and despite adding some ambience to the sounds, have a lot of metallic ring, giving a most unnatural sound. Musicians may be happier switching them off and using an external multi-effects unit. The 4-band EQ is

a luxury card, and with its reverb, chorus and surround effects it will have no problems in making a dent in this part of the market, as the price seems to be pitched at the right level. For the musician, sound quality is paramount if a card is to be used for hard disk recording, and its MIDI synth used alongside external synths. WAV recording and playback on the Maxi 64 is of good quality for a card of this price, and the facility to play back up to eight simultaneous channels of WAV files is excellent. Running a daughterboard through it will also add very little noise to the proceedings. The MIDI synth has a good range of usable sounds, although you probably wouldn't want to master an album with it. The DSP effects are many and varied, and the only slight downside is that the reverb sounds are nowhere near the quality of even the cheapest multi-effects units now available (or the

BUNDLED SOFTWARE

Like most soundcard packages, this one comes with a selection of software.

• *Sound Impression* is one of the standard 'rackmount hi-fi' simulations, which lets you control each part of the package. It works well, and has an integral Wave editor which is basic but quite usable.

• *Quartz Audio Master SE* is a 32-track 'direct to disk studio' (only supplied with the Home Studio version of the Maxi Sound), which allows combined MIDI and audio tracks to be recorded. Up to eight stereo audio playback tracks can be used, along with one stereo or two mono recording tracks, and 32 tracks of MIDI data. Both Grid and Score editors are provided, and extensive control of all the DSP effects is available. The whole package is comprehensive, even including options such as SMPTE sync.

• *Cakewalk Express* now seems to be included with every musical package, but it is a good basic sequencer at the bottom of an impressive range of packages — start here and work your way upwards to the dizzy heights of

Cakewalk Pro Audio!

• *Internet Phone* uses the card's full-duplex capability to offer two-way on-line conversations. This theoretically allows you to have international telephone conversations for the cost of a local call, but both parties will need a full-duplex card, and unless you have a high-capacity ISDN line rather than a BT one you're unlikely to find it much more than a novelty. Still it's free, so who's complaining!

Figure 3: Sound Impression rack control — looks like a hi-fi but doesn't need wiring up!

> "Even the forthcoming AWE64 cards from Creative Labs don't have hardware duplex capability."

described as "graphical and parametric", but is in fact a 4-band sweep equaliser with fixed Q setting. This works well.

Many people will be intrigued by the Surround window. The surround effect can be applied to a stereo signal, or, if you activate the '4 outputs' button, you can attach the second amplified stereo output socket on the back of the card to a pair of rear speakers, giving the classic quadraphonic effect. All of these DSP settings can be adjusted via MIDI NRPN, so that it should be possible to, for instance, set up a mixer in *Cubase* to control everything in real time. The surround settings are very effective, and I can see this facility being used a lot.

CONCLUSIONS

The Guillemot Maxi 64 falls neatly between two stools, and so should easily find its own niche. Games players and general users will consider this

DB50XG's onboard effects). With the possibility of adding up to 16Mb of sample RAM — so you could use the Maxi as a multi-voice PC sampler — in addition to all its other facilities, the total package does look and sound impressive. Ultimately, this is a high-quality and well-priced soundcard, with a huge array of features. I wish Guillemot every success with it. [SOS]

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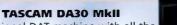
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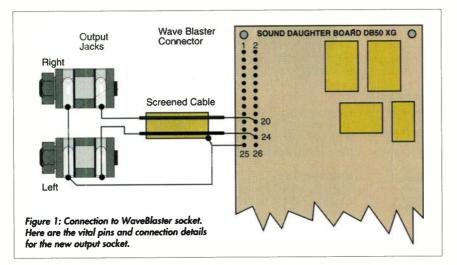
When you buy a daughterboard for your soundcard, you may be impressed by the sounds themselves, but disappointed by the sound quality. MARTIN WALKER shows how to add hi-fi quality outputs to a daughterboard, and passes on some editing tips to help you get the best from your purchase.

MAKING THE MOST OF SOUNDCARD DAUGHTERBOARDS

When I first bought one of the highly-regarded Yamaha DB50XG daughterboards (see full *SOS* review in the May 1996 issue), I was extremely pleased with the sound quality from the board itself. Unfortunately, this high-quality signal normally has to pass through the generally lo-fi circuitry of the average soundcard before emerging into the outside world. The design of typical soundcards is obviously driven by market forces, and because prices need to be kept down for the typical user playing games with a couple of 3-inch satellite speakers, performance is never likely to compete with the sort of audio equipment that musicians normally use outside the PC. However, since the output of all daughterboards is a stereo audio signal, rather than a digital signal which is then converted back into analogue by the soundcard (such as the wavetable portion of a Soundblaster AWE32), it is possible to tap into this to provide a direct output. This can be connected to a line input on your external mixing desk, bypassing the soundcard completely. In addition to providing a much better quality signal, this direct output has the advantage of separating the daughterboard audio from any other soundcard sources, such as .WAV playback, allowing external processors to be used and EQ to be independently tweaked if desired. It should be pointed out that making such modifications will render your warranty null and void, for obvious reasons. Anybody who can use a soldering iron carefully should be able to carry out this mod in a couple of hours, and the difference in performance can be remarkable, but naturally you do it at your own risk and *SOS* cannot be held responsible for any damage caused!

THE MOD SQUAD

This modification will work with any soundcard that sports the WaveBlaster 26-pin socket, including the Soundblaster 16 and AWE32, Sound Galaxy NX Pro 16 and Basic 16, Turtle Beach Tahiti and the new MultiSound Pinnacle, MIDI Edge interface, and others. With the number of awards being given to the Yamaha DB50XG, all manufacturers will be sure to let you know if your card can support it! If you use a Tahiti or Pinnacle, the soundcard sound quality is already particularly good, but you will still gain a separate output. Other daughterboards, such as the WaveBlaster II and Wave System, will also benefit from the mod, although of course the sounds still won't be in the same league as those of the Yamaha.

Thankfully, for our purposes, the DB50XG has all of its pins neatly numbered on the board itself, so it's easy to point you to the appropriate places. Essentially, all daughterboards are MIDI synthesizers, so the WaveBlaster socket has a MIDI In and Out (and no, you can't just solder a wire onto these pins to give yourself another MIDI interface free of charge; firstly, manufacturers normally tap into the existing MIDI interface on the soundcard rather than add an additional one for the daughterboard alone, and secondly, you would need extra circuitry and opto-isolators to connect it to the outside world). The other connections on the WaveBlaster socket are for powering the card, and the two we're mainly interested in are the Left (pin 24) and Right (pin 20) outputs, along with a suitable earth point (any of pins 15, 17, 19, 21, 23, or 25). These are the three pins that allow us to connect to the outside world (see Figure 1). ▶

Figure 1: Connection to WaveBlaster socket. Here are the vital pins and connection details for the new output socket.

Making the most of Soundcard Daughterboards

► Any twin-core screened cable can be used to connect to these pins (it must be screened, by the way, to minimise the possibility of interference from the rest of the PC getting into the audio signal — see the 'Picking Up Strays' box), and possibly the best cable to use for this task is one of the slimline balanced mic cables. These are generally about 3mm in diameter, and flexible enough to be threaded amongst the many other items inside your PC before you can get it to the outside world. Be generous with the length of the cable. After all, it's easy to cut some off if it turns out to be too long, but not so easy to add some back if it's too short! If, like me, you're concerned about voiding the warranty on your DB50XG, and are connecting to an old soundcard, you can, if you prefer, solder the cable onto the soundcard side of the WaveBlaster socket (ie. onto the non-component side of the soundcard). Make sure that the joints are sound, and use the minimum amount of solder. Check mechanical integrity by gently wiggling each of the connections (the last thing you want is for one of the wires to fall off and touch something else!).

At the other end of the cable, you need to connect to a suitable socket. This is far easier than having a few metres of cable hanging out of the back of your PC with a couple of jack plugs on the end! There are several suitable socket types — most soundcards use stereo 3.5mm jack sockets, but these can be flimsy and unreliable. The next step up would be a quarter-inch stereo version, as used by most headphones, but I must admit that I chose to use two standard quarter-inch mono sockets. This means that you can use standard jack-to-jack leads to connect to your mixer.

There are several possible places for these sockets. On the back of my PC there were several unused blanking plates, normally used for additional serial or parallel ports, that could be fairly easily pushed out to leave a handy-sized hole ideal for the fitting of a pair of quarter-inch jack sockets. If there are no suitable blanks like this on your PC, you could simply remove a spare blanking plate where there is an empty card slot, feed a few inches of your cable through the gap, and solder this to an in-line socket. It will then look similar to many of the multi-output MIDI interfaces which use 5-pin DIN sockets on the end of a short set of cables. Try to route the cable well away from the graphics card and disk drives to minimise interference (see 'Picking Up

Strays' box). Before reconnecting the power, check that no stray blobs of solder are bridging any of the pins on the WaveBlaster socket, and before replacing the cover of the PC, double check that the cable is not going to be trapped and squashed when the lid is screwed down. And there you are — the perfect way to hear what Yamaha intended. In practice, sound quality will vary slightly depending on your PC, but with half a dozen MIDI channels going at once, the background noise level should be on a par with many synth keyboards and modules, and well up to being used on CD tracks!

CARD TRICKS

Apart from its high sound quality, the other main feature that sets the DB50XG apart from its contemporaries is the amount of control that it provides over many sound parameters. Many of these can be controlled in real time using a variety of MIDI controllers. But probably the single must-have feature that distinguishes this card is the provision of three completely separate effects processors. Two of these are set globally and default to reverb and chorus (although each can be selected from a wide range of other types). The effect sends are individual for each MIDI channel, so that the amount of reverb or chorus can be set independently for every sound — a vast improvement over many stand-alone keyboards and rackmount modules. However, there's more to come! The third processor type can also be inserted on any channel, such that it acts only on one instrument. This allows, for example, screaming distortion to be set up on a lead guitar sound, fully controllable in real time, entirely independently of the other two processors. This is really what sets the DB50XG apart. Even though the set of sounds is still basically of General MIDI origin, albeit with many extra banks of variations, a track recorded in XG format will be in a different league from one using GM or GS. Some people may think that to access all this power they really need a dedicated synth module. But, as many have found, the limited number of buttons on the average MIDI module often results in a none-too-intuitive user interface.

BEING IN CONTROL

If you have more than one or two synths, you'll probably be wondering whether to opt for a universal synth editor/librarian, or to buy a separate ►

PICKING UP STRAYS

The PC is host to a myriad signals. High-end soundcards tend to have much better shielding than their low-cost counterparts, which helps to reduce the amount of stray pickup which finds its way into the audio chain. Spurious computer-related interference normally exhibits itself as high-frequency whistles, often related to processor activity, as well as sundry noises such as tickling when the hard drive or CD-ROM kicks in. Some of these arrive at the soundcard via the power supply, since no filtering system is perfect, and the remainder are picked up as stray interference by the card itself. This latter type can be minimised by choosing a card slot as far away from the source of interference as possible. Unfortunately, this is not always easy.

The main culprits are normally the graphics card, disk drive and power supply, and it's often the case that moving your soundcard further away from the graphics card brings it closer to the disk drive or power supply. If you're prepared to spend some time tweaking this (and it only has to be done once), the best way to optimise is to install your soundcard in several positions, and in each case record a few seconds of its line output

background noise level to DAT. Make sure that the output level is set to maximum and any gain controls on inputs are turned right down. You'll soon see whether any positions are better or worse with your particular PC. Don't bother using the microphone inputs at all if you can help it, as this will exaggerate any problems, since the signal levels are so small. Turn any mic gain controls right down to minimise their contribution.

Making the most of Soundcard Daughterboards

dedicated editor for each one. I have followed both routes at various times, but for in-depth editing there's nothing to beat a well-designed dedicated editor, simply because it *is* dedicated — every parameter can be laid out in a logical and useable fashion, rather than having to conform to the restrictions of a universal design template. With the DB50XG, Yamaha provide a demo version of an excellent dedicated shareware editing program called *XGedit*, which was written

Figure 2: The XGedit main screen, showing the new altered parameter readouts in yellow.

specifically for use with the XG range — DB50XG, SW60XG, MU50 and MU80 — and the general XG level 1 standard. It really is easy and enjoyable to use, and it was only a few days before I had sent off my registration fee to the author, Gary Gregson, to unlock the save features and do some serious editing. The most recent version is 2.0, and this has some sparkling new features, as well as addressing some niggles from previous versions. Thankfully, Gary is a software author who responds to user feedback. For me, one of the few frustrations with version 1.2 was that all saves of complete banks defaulted to filenames with .ALL extenders. This is what *Cubase* uses for its song files, and meant that it was tricky to place

song files and their respective synth edit files in the same folder without manually editing the appropriate filenames. Version 2 addresses this problem by altering default file extenders to .XGE — a small point, but one that will put a smile on the faces of many existing users.

New features include the ability to load standard MIDI files, and any edited parameter now changes colour. This is ideal if you download XG MIDI files from one of the XG web sites (see 'XG Web Site' info box), as it allows you to load them and see exactly how other musicians have created their sound — every edited parameter, from sounds selected and mix levels to reverb settings, can be seen clearly.

Another welcome addition is an enlarged help file which includes details of how to use *XGedit* with any of the multi-client MIDI drivers that are now available (more details on this later).

STRESS-FREE SYSEX

Many people are confused about using System Exclusive commands to access the many parameters that can be changed in real time with cards such as the DB50XG. Thankfully, no-one needs to type in strings of unintelligible hexadecimal codes any more — this is exactly why programs such as *XGedit* have been written. It gives easy access to every one of these parameters, allowing full real-time control in NRPN (Non-registered Parameter Number) and SysEx form to be recorded into a sequencer. Moving one of the on-screen controls causes SysEx data to be automatically sent to the synth to alter the parameter setting. In the context of one-off changes, such as selecting sounds, setting levels, effects routing and so on, once you have tweaked everything to perfection, you simply save the final set of edits as a PC file. Then, the next time you're working on this track, you simply run *XGedit*, load up the edited file, and this will then be automatically downloaded to the daughterboard.

Real-time edits are slightly more complex, but only so far as the setup is concerned. Once your sequencer, MIDI interface and *XGedit* are correctly connected, you can play a track using your sequencer, change settings such as filter frequency/resonance or reverb levels in real time, by altering any of the on-screen *XGedit* controls, ▶

CD-ROM AUDIO NIRVANA

In the same way as the daughterboard mod, the audio signal from typical CD-ROM players can also be intercepted and re-routed to external sockets on the rear of the PC casing. All CD-ROM players, in addition to the digital cable that connects the data from CD-ROM disks to the computer busses, will also have an audio cable. which is normally connected to the soundcard for replay of music CDs or audio tracks on CD-ROMs. If you cut this cable,

it's possible to route it to a couple of standard quarter-inch jack sockets. Simply connect both cut ends to the jack sockets, so that you are effectively just tapping in on the original cable, and leave the original plug connected to the soundcard. This will allow you not only to replay CD audio through your soundcard mixer as before, but also to have access to a much higher-quality signal directly connected to the outside world. For several years

I've used one of the old double-speed Panasonic CR562B CD-ROM drives. The audio output from this is of extremely high quality, and indeed, about 18 months ago, after buying a new, £250, highly-respected stand-alone CD player, recommended as a 'Best Buy' by many of the hi-fi magazines, I ended up getting a full refund when I discovered that despite very careful listening, the hi-fi stand-alone player could not be reliably identified in a blind test with my hot-rodded CD-ROM audio player!

Making the most of Soundcard Daughterboards

▶ or set external MIDI controllers such as mod wheel, pitch bend, or aftertouch to control selected parameters from your normal keyboard. All of these changes will be recorded by the sequencer (as long as your sequencer will record SysEx data), and again, no numbers need to be typed in. The only fly in the ointment in the past has been the lack of multi-client MIDI drivers.

NOT NOW — I'M BUSY

The main reason why most people may have given up trying to alter synth parameters in real time in the past is because as soon as they tried to run both a

from your keyboard, you can also (and in real time) alter controls using a program like *XGedit*. The SysEx data output by *XGedit* will then be recorded by the sequencer, along with the normal MIDI note data.

Thankfully, although most MIDI drivers are still single client, there are several small utility programs available that can add multi-client capabilities to both inputs and outputs of most MIDI drivers, allowing all of the facilities mentioned above. Herman Seibs' *MultiMid* driver, and the Hubi *LoopBack* and Hubi *MIDI Cable* drivers are both freely available via the Internet (URLs for these are given in the 'XG Web Site Info' box), and although

Figure 3: Setup using MultiMid to hear edits whilst playing back a sequence.

Figure 4: Setup using MultiMid to record edits whilst playing back a sequence.

the concept takes a bit of getting your head around at first, in practice, once you've set up your system, you can simply get on with making music. To use an editor and sequencer simultaneously, and hear sound edits while the sequencer carries on playing, you need the setup shown in Figure 3. This overlays the MIDI output so that several programs can send data to it. Figure 4

sequencer and a synth editor, one or other would complain that the MIDI interface was already being used by the other program, and either refuse to load, or carry on regardless but refuse to accept any MIDI data at all. Even if both worked together, it was often in a very flaky way, with random crashes being quite a common occurrence (I know, because it used to happen to me!). The problem is that most of the Windows drivers provided with MIDI interfaces are of the single-client type — they will only work with one program at a time. This prevents an editor having access to the MIDI output to change any sound while the sequencer is also sending MIDI data to the same output. Although there are now several drivers that allow this (notably the Twelve Tone MPU401 driver shipped with *Cakewalk*) what is ideally needed is a driver which allows not only multiple clients (programs) to output to the MIDI device, but also allows multiple clients for the MIDI input. Then, not only can you generate MIDI data

shows a more advanced setup that allows both your keyboard and software editor to send data to the input of the sequencer so that, when set to record, it will capture not only notes from the keyboard, but also any SysEx or controller data that emerges from the editor when you twiddle an on-screen control. Do bear in mind that controller movements, and in particular SysEx, can result in large amounts of data being generated. Although most sequencers will allow you to give note information a higher priority, you may find that thinning out the extra information (for example, using the 'Reduce Controller Data' command in the Functions menu in *Cubase*) will produce no audible changes, but may overcome any timing problems that you get if your MIDI interfaces are starting to sag a bit with the extra load. Once you've got everything properly connected, you'll not only have much greater expression in your music, but far less noise from your outputs as well! **SOS**

There's never been a better time to burn your own CDs at home, as the price of CD writers and the necessary mastering software continues to fall. Now the Red Roaster bundle is bringing the homemade CD dream within the reach of Pentium PC owners. JANET HARNIMAN COOK experiences burning desires...

Hot Stuff!

HOHNER MIDIA *RED ROASTER 24-BIT* CD PRE-MASTERING SOFTWARE FOR PENTIUM PC

The cost of CD Recorders and their distinctive gold blank disks has steadily fallen during the past year, and the software necessary for audio CD production on the PC has now started to appear; Hohner Midia's *Red Roaster 24-bit* is the first application to bring desktop CD creation to the Pentium PC, and is not only affordable, but uses Windows soundcards rather than expensive external audio hardware. The software gets its name because it 'burns' or writes an audio CD that conforms to the Red Book standard used worldwide by the recording industry. The CDs *Red Roaster* creates can be played on ordinary CD players like the one in your car or hi-fi. And there is no doubt about it: a CD of your work will make a better impression on your clients or on A&R departments than would a cassette. In addition, your CD can be taken to a duplication plant and used to manufacture a CD run. By authoring the CD master yourself, you retain control of the final creative stage of the audio production process (strictly speaking, the process of creating a CD master is more accurately called pre-mastering, as the glass master made by the duplication plant from your CD is the actual master). For although a good mastering engineer may provide the final touch of magic that can make the difference between commercial success or failure, a poor mastering job can ruin the whole project. Also, if you are producing short runs of demo CDs on a tight budget, you have the option of avoiding expensive mastering services.
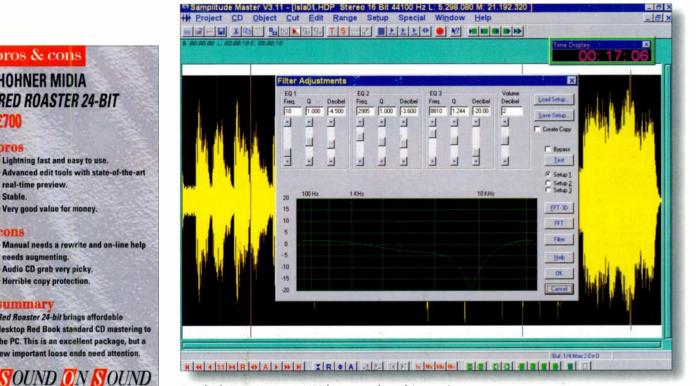
SETTLING DOWN TO A GOOD *ROAST*

Hohner Midia's *Red Roaster 24-bit* is a 32-bit application for Windows 95 and Windows NT v4. The version reviewed here is for standard 16-bit Windows MMS soundcards. The package consists of two floppy disks containing the installation files and a spiral-bound 84-page A5 manual.

The *Red Roaster 24-bit* package actually consists of two applications — *Samplitude Master v3.11* and *PoINT CD Audio v1.2*. *Samplitude Master* is used to assemble the audio tracks and perform the final edits. When this is done, a Table of Contents (TOC) is prepared which contains the track list, the track indices and the associated timing information. The TOC is loaded into *PoINT CD Audio* for PQ subcode editing and Red Book CD creation.

SAMPLITUDE MASTER

The first step in creating a CD with this package is to assemble the audio tracks in *Samplitude Master*. It is a good idea at this stage to make a track sheet on paper or a word processor containing relevant details such as names, times, source, running order and edits required.

Samplitude Master's parametric EQ has spectacular real-time previews.
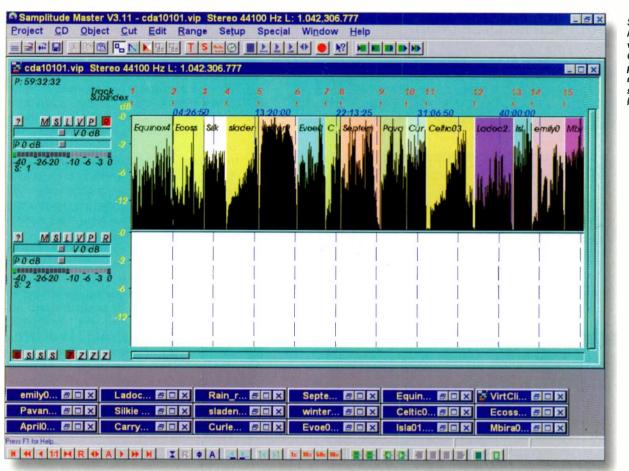
Samplitude Master VIP with Tracks, CD Index points and minimised source audio files.

Audio can be recorded in 16-bit resolution or in 24-bit floating point format. Higher bit rates reduce data quantisation losses; when 16-bit audio is resampled, as for example during sample rate conversion, the reductive nature of sampling can lead to a perceptible loss of audio quality. Compared with analogue processing, these losses are slight, but they become increasingly more obvious as edits accumulate. If editing is carried out in 24-bit floating point format, the audio remains essentially distortion-free. Audio in 24-bit float is saved to disk in 32-bit format and requires about 20Mb per stereo minute. On playback or before creating a CD, the 24-bit audio is converted back to 16-bit.

If your source audio is on 48kHz or 32kHz DAT, you will need to resample the material to the CD-standard 44.1kHz rate. This usually entails recording the material into 48kHz WAV format and then converting to a new file at 44.1kHz. *Samplititude Master* cunningly combines these tasks and performs them in real-time as you record from DAT. This saves disk space, cuts production time, and minimises quality loss. Incidentally, when naming files, you need to use 8-character names for your projects; although *Samplitude Master* accepts long filenames, *PoINT CD Audio* does not!

If your audio tracks are stored on your hard drive as Windows WAV files, you can import them directly into *Samplitude Master*, by just hitting the 'W' key and double-clicking on the file entry. The audio file is quickly loaded and is placed in the VIP list (*Samplitude*-speak for a Virtual Project, or the collection of audio tracks and associated data that will make up the CD). You then repeat the procedure until all your audio tracks are entered, and then save the VIP file. In a VIP, editing is non-destructive ie. no permanent changes are made to the wave files themselves. Incidentally, it took *Samplitude Master* a staggering five seconds to load a VIP file containing over 57 minutes of audio from the hard drive — impressive.

When loaded into the VIP track list, an audio track is called an Object. It appears as a pastel-shaded wave display block and is arranged horizontally on the track bar, in playback sequence. The lower toolbar is used for navigation and zooming. The upper tool bar contains file, track, sub-index, TOC, transport, fades and cursor commands. One small glitch I discovered was the failure of the keyboard left and right arrows to provide horizontal scrolling in VIP mode. *Samplitude Master* has multiple levels of Undo, but does not include an Undo History describing the various edit stages.

INDEXING

Each track is automatically loaded in sequence with the two-second gap required by the Red Book spec in place (if you wish, you can edit the default pause time in the CD menu). To index your tracks, you simply click the red Auto Indices button on the ▶
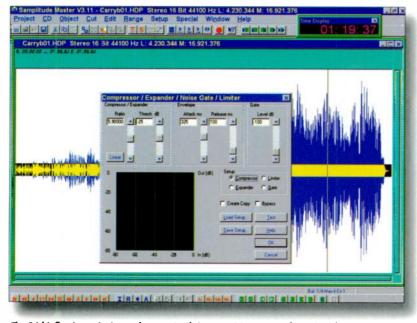
ROAST THOSE WINDOWS — WINDOWS 95 SYSTEM ADJUSTMENTS

Two small adjustments have to be made your PC before it can be used with *Red Roaster*. First, you need to double-click on the CD-ROM icon in your PC's Device Manager to call up the CD-ROM list, and go into Properties\Settings. In the Options box, tick Disconnect and leave Sync Data transfer and Auto Insert notification unticked. If there is a CD-ROM reader also installed, similarly tick Disconnect and Auto Insert notification, but leave Sync Data transfer unticked. CD Recording is very demanding, and requires an uninterrupted data transfer stream for the duration of the recording process. Furthermore, before writing any CDs, ensure best results by defragmenting the hard drives on the PC that contain the operating systems (DOS & Windows 95), program applications and audio data.

The 24-bit floating point internal processing brings precision to your dynamics editing.

▶ toolbar next to the CD icon. Track index numbers appear above the time ruler, and Time can be displayed in seconds, samples, beats, CD MSF (75 frames per second) or as SMPTE. You can also add track indices and sub-indices during playback by clicking on the T and S icons in the toolbar. Indices may be removed via the CD menu. To rearrange the track order, you hold down the Control key and then drag and drop the Object to the new position; all track indices are then updated automatically. In this way, you may completely restructure the track sequence, and even re-edit the source. There can be a maximum of 99 audio tracks, each with 100 sub-index points (these are used mainly in sample and sound effects CDs, and in classical recordings).
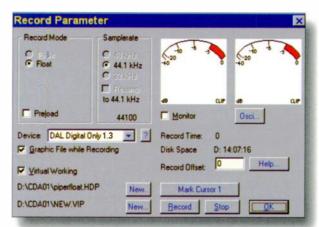
When you are satisfied that the tracks are correctly indexed, a click on the CD icon produces the Make CD dialogue box, which contains three modes of TOC generation. The 'Make No Calculations' option is the most basic, and simply creates the TOC from the original wavefiles; no audio editing is performed. 'Use Original Files' mode applies all

24-BIT RECORDING

Recently, we have seen the arrival of PC soundcards that are capable of greater than 16-bit digital-to-analogue conversion — and of course, the bigger the bit resolution, the better the audio quality. The Arian Darc 2 audio card is supported by *Red Roaster 24-bit*. Darc 2 is a well-featured professional PC soundcard with true 24-bit stereo DACs, and has optical and co-axial connectors for digital I/O. The Darc 2 is £2049.99 including VAT, and is also available from Hohner Midia (see address at the end of this article).

fades, crossfades or volume curves to the WAV files. It is a good idea to back up the audio files in case something goes wrong, as this is a destructive edit process! The 'Complete File' option makes one large wave file image of all the audio files together with fades, crossfades or volume edits. This requires over 11Mb of disk space for every minute of audio (60mins = 660Mb) so you need to make sure there is sufficient free disk space if you choose this option. The process is non-destructive, so the new 'mega file' will require disk space in addition to that occupied by your original wavefiles. The TOC is stored in the VIP file, and after creating the TOC, the VIP file should again be saved.

Do not use the 'Call up *PoINT CD Audio*' link in 'Make CD' dialogue. If you do, you will get an information box stating 'an unexpected error has occurred whilst reading the serialisation information'. You are advised to check the 'consistency' of your hard drive and if necessary re-install *PoINT CD Audio*. If you do get this far, do not follow this advice! The information box is wrong — the link between the software is just not yet properly implemented. Instead, quit *Samplitude Master* and manually open *Point CD Audio*.

EDITING

In addition to the non-destructive edit tools in VIP mode, the *Samplitude Master* Edit menu also has some very powerful Dynamics, EQ and Noise

Audio recorded into Samplitude Master is converted from 16-bit to 24-bit.

Reduction tools for directly editing the wave data on your hard drive. Once again, before embarking on destructive editing, it is best to make a backup of your source material, so that if the edit subsequently goes wrong, you can restore your original audio file and start again. *Samplitude Master* uses 32-bit assembler language to achieve ultra-fast processing times — typically half that of 16-bit applications. With speeds like this, studio quality real-time preview of EQ and Dynamics processing is possible. When you're happy, you can save your custom processing settings.

The EQ in *Samplitude Master* is excellent; it is a 3-band parametric device covering the audio frequencies 10-25kHz, with +/- 20dB of cut/boost. The Q point is variable between 0.1 (10 octaves) ▶

24-BIT INTERNAL PROCESSING & DITHERING

Samplitude Master v3.11 can perform many of its internal audio processing calculations in 24-bit float precision mode. The operations are track bouncing, fades and crossfades, volume and pan gain, and curves. Internal precision may be changed from the Setup menu — the default resolution for Pentium PCs is 24-bit. When converting audio from 24-bit to 16-bit, audible aliasing and distortion (particularly audible in quiet passages) may occur due to quantisation errors. Although very low in actual volume,

artifacts possess a metallic edge that sounds unpleasant and intrusive to the human ear. Dithering introduces low-level randomised noise to mask these artifacts, and can improve the subjective perception of sound quality. In *Samplitude Master*, dithering happens in real time and requires no additional editing. The Setup menu includes a choice of dithering options, and finding the most suitable for a given piece of audio is a matter of experimenting with the alternatives.

PoINT CD Audio's main page, with Table of Contents and sub-code editing.

interpolation algorithm to restore damaged audio.

The noise reduction algorithm works best on constant noise such as air conditioning units, analogue tape hiss, fan noise from your PC, and electrical noise caused by transformers and mains hum. A sample noiseprint is taken from an otherwise quiet part of the audio file, and is then used as a template by the noise reduction algorithm. The area of audio to be processed is selected in the usual way and the processing may be fine-tuned using the Resolution, Precision and Absorption options in the Noise Reduction dialogue box. The Preview function is used for finding the best settings. Some care is needed, as artifact noise may be introduced by the noise reduction algorithm if the levels of the frequencies constituting the noise are as loud or louder than those in the audio material. If this happens, you can try repeating the process using less extreme settings.

One other potentially useful function is Load CD Tracks (accessed from the Project menu), which purports to lift audio tracks as WAVs onto the PC hard drive, and would be very useful for loading samples to hard drive from audio CDs. Sadly, the function did not work with my Plasmon CDR 4240. In fact, it would not even work on the audio CD I had created using *Red Roaster*! A colleague also reported similar results with his Yamaha CD writer, so for the moment, the jury is out on this one.

POINT CD AUDIO

Once you have processed and edited your audio, *PoINT CD Audio* extracts the embedded Table of Contents from the VIP file created in *Samplitude Master*. You open the TOC file corresponding to your project from the File menu, and can then edit the PQ subcode data for each track by double-clicking on it. At this point, you can make provision for tracks that have been recorded using pre-emphasis (a treble boost applied by some DAT machines, such as those made by Casio, when recording); a flag is entered into the subcode on the CD that instructs the CD player to use its de-emphasis circuitry. You can also implement Copy Prohibit (the precursor of the dreaded SCMS system used on some DATs) to prevent a track being digitally re-recorded.

Once the subcode editing is completed, you're ready to put a blank CD into your CD-writer and click the red Write button! The final dialogue box allows you to select the recording speed of your CD-R, and the Simulate Recording option can be used to determine the optimum write speed. Simulate Recording is also used to perform a test run of the recording process. No data is recorded, and any faults or discrepancies in the TOC will be revealed. When you are satisfied that all is well, you just click Write to start the recording process proper. The Plasmon CDR 4240 used for the purposes of this review (see the 'Mind Your Ps & Qs' box elsewhere in this article) writes at 2x speed, and my 57-minute project took about 30 minutes to write to disk.

▶ and 10 (0.1 octave). This makes it perfect for precise attenuation of any band in the audio range, and allowed me to get rid of a weird 22-23Hz peak that appears mysteriously on many of my recordings. The frequency curve is displayed graphically, and changes are previewed in real time — spectacular! The Dynamics functions (compression, expansion, limiting and normalisation) work in smart mode — in other words, the algorithms are anticipatory — and do not produce peak distortion or other artifacts. There is also a graphical display of the dynamics curve, and ratio, threshold, release and gate level parameters may be adjusted and previewed in real-time. Finally, *Samplitude Master* claims to be the only application that can remove digital and analogue clipping; it uses a high-quality

MIND YOUR Ps & Qs — THE CD-R MINEFIELD

Great care must be taken when buying a CD recorder. Not only do CD Recorders vary in price considerably, but many older models, while useful for data backup, are not suitable for audio CD mastering, as they do not read or write Red Book PQ subcodes and indices. Every CD contains eight channels of subcode data (channels P to W). Red Book audio CDs use two of these — channels P and Q. The P channel data sends playback status information to the CD player; the Q channel data provides track order, running times, copy protection and other information.

The Plasmon CD Recorder CDR 4240 used for this review is based on a Matsushita (Panasonic) CW7501 drive unit with Plasmon software and drive electronics (firmware revision 1.13), and writes at 2x speed and reads at 4x speed. Disk-at-once recording is supported, and the Plasmon 4240 writes CD sub-indices and ISRC, making it suitable for Red Book audio CD mastering and CD-ROM creation. Some CD recorders are very fussy about the brands of blank CD-R they will

recognise, but the CDR 4240 writes to all eight manufacturers' blank disks. It does this by reading the manufacturer's code on the blank disk and configuring its laser accordingly. The Plasmon CDR 4240 is good value for money, and has worked flawlessly on CD creation and data transfer tasks. In my opinion, it is one of the best drives on the market.

Prices for the CDR 4240 start at £482 for the internal drive supplied on its own and £546 for the external drive, and go up to £599 for the internal drive and £664 for the external drive when supplied in a bundle including Adaptec's *EasyCD Pro95* and *EasyCD Backup* data recording software, a SCSI card and cable. Contact Plasmon for further price details. All prices given here include VAT.

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📞 01763 262963.
📠 01763 262980.
🌐 http://www.plasmon.com

HOHNER MIDIA *RED ROASTER 24-BIT*

PoINT CD Audio's Recording Parameters dialogue box is the final stage in the CD writing process — note the Simulate option.

CONCLUSION

Samplitude Master is dazzlingly fast, and works well with *PoINT CD Audio*. The processes of audio and subcode editing are easy to understand, and the real-time effects preview is wonderful. However, the copy protection needs to be improved (see the 'Copy Protection' panel). The manual also needs a rewrite; it starts off well, giving an instant overview of the procedures used by *Red Roaster* to write a CD, but quickly becomes fragmented and incomplete — and occasionally, poor translations from the German result in misleading ambiguities, which are unforgivable in what is, after all, a technical manual. The Windows on-line help is a little better, but many items are not implemented, and there's no comprehensive index. The non-functioning link between *Samplitude Master* and *PoINT CD Audio* and the lack of warning concerning this in the documentation also caused problems.

Overall, it is hard to escape the impression that these parts of *Red Roaster* were left unfinished in the rush for commercial release. This is a pity, because it is obvious that a lot of great care, flair and effort has gone into writing the program routines. This said, *Red Roaster* is a very fine bundle and its developers deserve to be congratulated. It lives up to its promise of delivering affordable Red Book audio CD mastering, and will appeal to budget mastering facilities, project studio owners, sample CD producers, and anyone creating music demo CDs. **SOS**

COPY PROTECTION

Installation of *Red Roaster 24-bit* on my PC went smoothly, which was fortunate, as the copy protection employed on the disks makes it impossible to create backups. This form of copy protection is particularly unsatisfactory; it does not discourage the determined software pirate, and unfairly penalises the legitimate user. It is ludicrous that an application costing hundreds of pounds ultimately depends on the dubious reliability of a 20p floppy disk! If the applications need to be reinstalled and either program disk has become corrupted, the user is unable to use their Red Book CD Recording facility. This is unacceptable for professional software, and the prospect of losing business while waiting for replacement disks sadly undermines the credibility of *Red Roaster*.

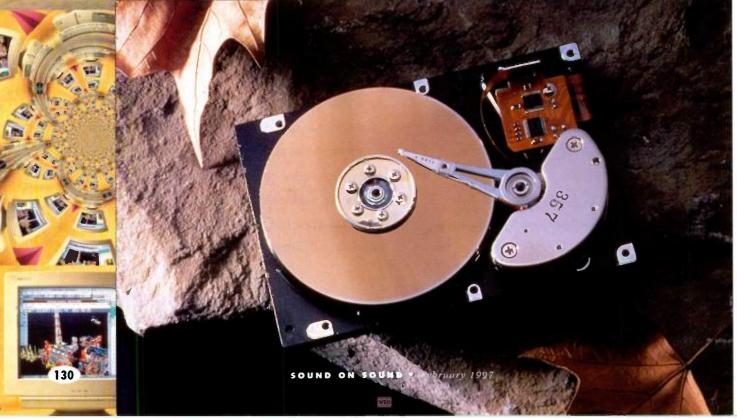*Many thanks to Stash Huchrak at GOSH in Bath.*

Drive

CHOOSING PC DISK DRIVES AND CONTROLLERS

time

SCSI (or Small Computer System Interface) hard drives have always been a popular choice for hard disk recording, since they have consistently seemed to be slightly ahead in the speed stakes compared with IDE and EIDE drives (see the first part of this article in last month's *SOS*). The biggest advantage of introducing SCSI into your PC system is that it allows you to run devices outside the PC casing as well as internal devices, and this is especially useful if you are running short of internal drive space or also wish to use the SCSI device with a Macintosh or sampler (see my previous feature on integrating samplers into your PC setup via SCSI in *SOS* October '96). For high-performance applications, SCSI still leads over EIDE devices, as it offers true multitasking, allowing multiple drives to be handled simultaneously.

There are two main disadvantages to SCSI. Firstly, you need to purchase and have space inside your PC for a separate controller card (the so-called Host Adaptor). This contains the necessary circuitry to control the maximum of seven SCSI devices that can be used simultaneously. Secondly, the price of SCSI drives compared with IDE drives of the same capacity is significantly higher. Having taken the decision to 'go SCSI', the choice of a controller card is not without its problems. Adaptec cards are widely recommended as providing good performance and maximum compatibility with a wide range of hardware and software, but if you

PART 2: If your PC's hard drive is full to overflowing, then perhaps you may be tempted to expand sideways. In the last of this short series, MARTIN WALKER examines the implications of SCSI, and its application to the world of music.

look in an Adaptec catalogue, there are literally dozens of different cards suitable for various machines and applications. In the search for maximum data throughput for the modern hard disk recording system, there are many factors that determine the maximum achievable speed. Most blanket recommendations mention the Adaptec AHA 1542 card as being an industry standard, since after selling millions of units it is the card that every manufacturer checks compatibility with, ▶

Choosing PC disk drives & controllers

▶ but even this has appeared in at least four flavours to date, and with a typical VAT-inclusive price not much under £200, it is surprising that so many people buy this card without considering the other options. At the other end of the scale, the Adaptec AVA 1505 card is less than £50, and may be perfectly adequate for basic applications.

MEET THINE HOST

The easiest way to narrow down the list of options is to exclude all but those suitable for the buss standard in your machine. Pentium machines have both ISA (16-bit) and PCI (32-bit) slots, so for optimum disk performance, always buy a PCI buss SCSI card if you have a Pentium. There are also VL-buss and EISA devices available, but these standards have largely been superseded by PCI. For those with a portable PC, SCSI controllers are available that use PCMCIA slots (Adaptec SlimSCSI) or which plug into the parallel port, but most of these have seriously restricted top speeds compared with normal cards, and whilst useful for connecting a Zip drive or CD-ROM, they are not really up to the rigours of hard disk recording. If you are considering buying a new computer, the other option is SCSI built into the motherboard. The chips often used in this case are made by Adaptec, and of course one more card slot will be available for other things. Overall, I think that SCSI motherboards may well be far more common in the future.

There are two more points to consider before narrowing down the field still further. If you propose to only use a SCSI drive with your PC, you will need booting capability on the SCSI controller so that it is possible to boot up your PC from the SCSI drive.

The new 1542CP is Plug and Play-compatible, plugs into an ISA slot, and uses buss Master DMA for best performance.

The Adaptec PCI 2940 SCSI Master card is ideal for a PCI system.

If you already have a main drive conforming to the IDE or EIDE standards, you can continue to boot from this as before, but use the additional storage of your SCSI drive in tandem. For many people, this is an ideal way of working, as EIDE drives are still less expensive than an equivalent-sized SCSI one of similar performance, and by keeping your word processor, sequencer and other programs on this separate IDE drive, the SCSI drive will always be used solely for audio data, and can be regularly defragmented for optimum performance. This approach also makes it easier to use one of the new removable SCSI drives such as the Iomega 1Gb Jaz drive, or the SyQuest 1.3Gb SyJet drive, since individual projects of up to about an hour's ▶

SCSI STANDARDS

• SCSI1
This is the original version, with a clock speed of 5MHz, from which all subsequent standards have developed. It allows asynchronous transfers (see the 'Jargon Buster' box) at up to 1.5Mb/second, and synchronous transfers at up to 5.0Mb/second.

• SCSI2
A tweaked version of SCSI1, with slightly faster handshaking and a doubled clock speed of 10MHz. The result is that asynchronous transfers can run at up to 3.0Mb/second and synchronous transfers at up to 10.0Mb/second.

• FAST SCSI
Any device which can do synchronous transfers at speeds in excess of 5.0Mb/second. SCSI1 devices cannot manage this. Fast SCSI can achieve 10Mb/second. At still higher speeds, Differential SCSI (see the 'Jargon Buster' box) needs to be used for reliability.

• WIDE SCSI
This standard extends the buss width from its original 8-bit width to 16 or even 32 bits, with a corresponding increase in maximum speed to as much as 20Mb/second with 16-bit, and 40Mb/second with 32-bit busses. By its very nature, any device using Wide SCSI is also Fast, so the standard is sometimes also referred to as Fast & Wide SCSI. To take advantage of this, you need special Fast Wide SCSI2 drives.

• SCSI3
This specification is still being finalised, but adds UltraSCSI and support for fibre-optic and serial cables, as well as more devices on the buss.

• ULTRASCSI
A method that enables very fast data transfer rate on the SCSI buss, with maximum rates of 20Mb/second and 40Mb/second for Wide devices. Few devices currently support this standard.

POTENTIAL DMA CONFLICTS WITH SOUNDCARDS

Using buss Master DMA can sometimes cause problems in audio-visual applications, since most full-duplex soundcards (ie. those that offer recording and playback simultaneously) will use two DMA channels, one each for recording and playback. With only three of the faster 16-bit DMA channels available on the PC (numbered 5, 6, and 7), conflicts can occur, and with a finite total time allocated for DMA, the SCSI card may hog too much DMA time, giving rise to audible dropouts with the soundcard. One solution to this problem is to use the lower priority DMA 7 for the SCSI card and select DMA 5 for playback and DMA 6 for recording. This will ensure that the soundcard gets all the time it needs for glitch-free operation.

Another option on some Adaptec controllers is to use the parameters 'buss On' and 'buss Off' to reduce the time that the SCSI card ties up the DMA buss. This can solve the problem, particularly if you are using the Digital Audio Card D Plus, but this may well reduce the maximum performance for SCSI. When DMA time starts to run short, it can also result in high-pitched whines appearing on the soundcard, or spurious faults. In some cases, you may even have to restrict your top SCSI speed to avoid audible problems.

However, rather than fighting for DMA time, it is well worth considering soundcards that don't use DMA. Turtle Beach soundcards, such as the well-respected Tahiti and Multisound, use a proprietary architecture named Hurricane which uses no DMA, and is claimed to

Choosing PC disk drives & controllers

▶ duration can each remain on their own cartridge, whilst your PC can carry on running regardless using the separate main drive. If you decide to opt for SCSI booting capability, it is also sensible to ensure that your SCSI controller supports the addition of floppy drives, so that you don't have to fill yet another slot with a floppy controller just to add this one vital feature.

The most recent cards, such as the Adaptec 1542CP, are Plug and Play-compatible, which in theory means that no conscious choices have to be made concerning system resources needed by the card. When the board is installed, resources such as BIOS and port addresses are automatically assigned to prevent conflicts. Used with Plug and Play SCSI peripherals and cables, the AHA 1542CP adaptor automatically locates all connected SCSI devices, and instantly resolves any SCSI resource conflicts. It also automatically terminates the SCSI buss, eliminating the need for jumpers and termination resistors. This all sounds wonderful, but back in the real world, even Adaptec recommend that if using their card with any legacy (non-Plug and Play) devices — such as virtually all soundcards — it is safer to disable Plug and Play support on the SCSI card and configure it manually.

At the upper end of the market, there are Wide SCSI drives that use 16-bit SCSI busses (this is the connection between the SCSI card and the SCSI device, and is not to be confused with the buss used by the computer itself, which is 16-bit in the case of ISA card slots, and 32-bit for PCI slots). If you decide to buy one of these drives, you will need a Wide SCSI controller card to partner it. This will provide bandwidth capability of up to 20Mb/second for 16-bit devices, and will still support standard 8-bit SCSI devices, which can work at up to 10Mb/second.

MODES OF OPERATION

SCSI controllers use various means to transfer data. Most of the simpler ones now use PIO mode (Programmed Input/Output), as do EIDE drives. This mode uses a significant amount of processor overhead, as the processor has to move data to and from the controller. Incidentally, this is the reason why many people were caught out buying early eight-speed CD-ROM drives — although data throughput is much higher at 1.2Mb/sec, the processor was tied up more than 50% of the time just reading data with some models, causing a bottleneck in the system. Simple SCSI cards using PIO will be fine for CD-ROM, CD-R, removable backup drives and a few tracks of hard disk recording.

The more advanced buss mastering mode uses our old friend DMA (Direct Memory Access) to carry out data transfer 'behind the scenes' without ▶

SCSI CONNECTOR TYPES

There are various plugs and sockets in use for SCSI, making mail order purchases a bit of a nightmare unless you can exactly specify which connectors you require.

- **50-PIN CENTRONICS**
The original SCSI1 used these connectors, which can be easily confused with parallel port leads. Don't be tempted to use a parallel port lead, though, however much cheaper it is. True SCSI cables use individually-shielded wires for each conductor, and are designed for high transfer rates (ie. high-frequency signals).

- **25-PIN 'D' TYPE**
Many more recent SCSI1 and SCSI2 devices use 25-pin 'D'-type connectors, as seen on many Apple Macs.

- **50-PIN 'D' TYPE**
Many SCSI2 devices use this type.

- **68-PIN 'D' TYPE**
Mostly used by Wide SCSI devices, to accomodate the larger number of pins needed for 16-bit use.

The many and varied connectors used by SCSI devices — study this before you try to buy a lead by mail order.

JARGON BUSTER

- **ASYNCHRONOUS TRANSFER**
Asynchronous transfer uses an interlocked handshake, where one device cannot send more data to another until it receives positive acknowledgement that the other device has received the last data transmitted. This ensures reliability, but is slower than synchronous transfer.

- **BUSS MASTERING**
A high-performance method of data transfer in which the host adaptor's onboard processor handles the transfer of data directly to and from a computer's memory using DMA (Direct Memory Access) without intervention from the computer's microprocessor. This is the fastest method of data transfer available for multitasking operating systems.

- **CABLE LENGTH**
At the sort of transfer rates we all aspire to, the length of the cable has a significant effect on overall speed. Asynchronous transfer is faster on short cables, while synchronous is faster on long cables. Calculations show that in ideal conditions, asynchronous transfers in SCSI2 can achieve 6Mb/second with a one-foot cable, 3.5Mb/second with a six-metre cable, and 1.5Mb/second with a 25-metre cable. With typical cable lengths between one and two metres, 5Mb/second can normally be achieved. The reason why real transfer rates often seem to bear little relation to manufacturer's specs — the latter are normally achieved with 'zero length' cables!

- **DIFFERENTIAL SCSI**
A balanced pair of wires is used for each signal, in the same way as a balanced mic cable. Susceptibility to noise and interference is greatly reduced, and longer cable lengths can be used.

- **HOST ADAPTOR**
The SCSI controller card itself.

- **SCSI (SMALL COMPUTER SYSTEMS INTERFACE)**
A buss interface standard that defines standard physical and electrical connections for devices. It enables many different kinds of devices (such as disk drives, CD-ROM drives, scanners, and tape drives) to interface with the host computer.

- **SINGLE-ENDED (NORMAL) SCSI**
A single wire in the cable for each signal that needs to be sent across the buss.

- **SYNCHRONOUS TRANSFER**
This type of transfer speeds up transfer by up to a factor of three, by sending multiple bytes before waiting for acknowledgement that the previous byte has been received. This mode was introduced to boost the performance on long cables. SCSI1 typically achieves rates of 5Mb/second with this mode, and SCSI2 has its maximum transfer rate limited to 10Mb/second.

- **TERMINATION**
A physical requirement of the SCSI buss. The first and last devices on the SCSI buss must have terminating resistors installed, and the devices in the middle of the buss must have terminating resistors removed. Failure to follow this will probably result in either unreliable data transfer or even SCSI devices that fail to work at all until termination is properly applied.

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SCSI Tutor is an excellent guide to all aspects of SCSI.

SCSI Bench is a useful way to test the speed of any connected SCSI device. By changing from sequential to random I/O, you will see how much slower a typical hard disk system will be when charging about on a defragmented disk.

▶ disturbing the main processor. Most higher-end SCSI controllers tend to use buss master DMA for highest performance. In general, choosing this type of card rather than PIO will give a far bigger increase in performance than upgrading from an ISA card using PIO to an equivalent PCI card. If you already use a Pentium PC and a soundcard with multiple DMA channels, do consider the higher-priced PCI cards, as

using 32-bit slots will result in less demand on DMA than using 16-bit ISA slots (see the box 'Potential DMA Conflicts with Soundcards' for further details).

For slower SCSI devices, such as CD-ROMs and disk drives that handle up to about 2Mb/second, there will be little difference in overall performance between the different PC buss standards. With faster SCSI drives for hard disk recording, throughput will be significantly better if you use SCSI controllers on the PCI buss (the performance more than doubled for one user when they upgraded from an Adaptec 1542 ISA card to a PCI version, but results will vary widely depending on your system).

OLD WIVES' TALES

Buying a faster SCSI card will not make your existing drive go faster unless the SCSI card it replaces has previously caused a bottleneck. Even plain vanilla SCSI1 is capable of achieving 5Mb/second data transfer under ideal conditions, using short cables, synchronous transfer (see the 'Jargon Buster' box), and a fast PC. So, even with the cheapest SCSI card, you could carry out CD-R recording, data and sample backups with a Zip drive, or even simple hard disk recording on up to about four tracks, with few performance limitations caused by the card itself. However, cards using PIO mode (and also EIDE drives, which use the same type of transfer) give a significant processor overhead which, whilst fine in many other applications, may cause a bottleneck with higher-end PC hard disk recording systems. This is also why the latest EIDE drives, although having comparable speeds to their SCSI counterparts, can still limit overall performance, by tying up the processor when it could be replaying an extra audio track or three. The maximum speed of any PIO mode device will also be affected by the processor speed. A faster machine will thus tend to speed up the disk transfer rate, and this is one reason why drive manufacturer's speed figures can only ever be used as a guide.

For hard disk recording, not only the sequencer software design affects the attainable number of tracks, but also the efficiency of the digital audio drivers written by the soundcard manufacturer. All of these factors explain why the only valid way to measure overall disk drive performance for musical applications is to run a utility like

ADAPTEC SCSI HOST ADAPTOR PERFORMANCE DETAILS

ADAPTEC MODEL NUMBER	SCSI BUSS TYPE	GREATER THAN 1GB?	MAXIMUM SYNC SPEED	MAXIMUM NO. OF DEVICES	TYPE OF TRANSFER MODE	BOOT DISK SUPPLIED?	FLOPPY DISK SUPPLIED?	TYPICAL STREET PRICE
AVA 1505	SCSI2	Yes	5	7	PIO (ISA)	No	No	£45
AVA 1515	SCSI2	Yes	5	7	PIO (ISA)	Yes	No	£65
AHA 1510B	Fast SCSI2	No	10	7	PIO (ISA)	No	No	£53
AHA 1522B PnP	Fast SCSI2	No	10	7	PIO (ISA)	Yes	Yes	£115
AHA 1542C PnP	Fast SCSI2	Yes	10	7	DMA (ISA)	Yes	Yes	£185
AHA 2920 PnP	Fast SCSI2	Yes	10	7	PIO (PCI)	Yes	No	£115
AHA 2940U PnP	Ultra SCSI2	Yes	20	7	DMA (PCI)	Yes	No	£185
AHA 2940UW	Ultra SCSI3	Yes	40	15	DMA (PCI)	Yes	No	£225

EZ SCSI 4.0 is supplied with many cards, and contains a wide variety of useful utilities, including a Tape Backup and CD copier.

HDSPEED.EXE (from *SAW* manufacturers IQS) or the *Cubase* performance utility (see my feature on PC hard disk recording requirements in *SOS* November 1996 for further details on these).

RECOMMENDATIONS

Although there are several other manufacturers of good SCSI cards, Adaptec is always the one that other people recommend. They have been in the marketplace a long time, have millions of units in the field with a huge range of types for every application, and are the first port of call for every SCSI device manufacturer when checking compatibility. Soundcard manufacturers also tend to use Adaptec chips for their onboard SCSI CD-ROM interfaces (for instance the Creative Labs SoundBlaster 16 SCSI). If you need a SCSI interface for a CD-ROM drive, do investigate this combined approach, but don't expect to add a high-speed SCSI drive to it. If you need a stand-alone SCSI card for CD-ROM, CD-R, or a lower-speed drive such as the Iomega Zip or SyQuest EZ135, then the AVA 1505, at a VAT-inclusive street price of about £45, will do the job adequately — or with booting capability, the AVA 1515 model comes in at about £65.

For people on a limited budget, you will get reasonable throughput using one of the AHA 1520 series (AHA 1510, 1520, 1522), but the widest compatibility for different applications is still offered by the AHA 1540/2 cards, which are even more attractive in their new Plug and Play incarnations, and will provide much lower overhead using the buss master DMA transfer mode. Anyone with a Pentium and PCI buss is well advised to opt for one of the AHA 2940 series, which will cater for high-spec hard disk recording. However, don't bother to pay more for the 2940W or 2940UW types unless you plan to buy a Wide SCSI drive, since you will only get the higher performance with these models.

CONCLUSIONS

Ultimately, when using a SCSI device, the transfer speed will be determined by the weakest link in the chain, and for this reason, there is no point skimping on the controller card and causing a bottleneck. On the other hand, there is no point buying a high-spec device to run an eight-speed CD-ROM drive which transfers data at 1.2Mb/second, unless you need a high-spec AV system, which has to process video files simultaneously and requires all the processor time it can get. I needed a SCSI card to work with a SyQuest

FURTHER INFORMATION

- **ADAPTEC**
  http://adaptec.com

- **DAL (CARD D PLUS)**
  http://www.digitalaudio.com

- **IOMEGA (ZIP AND JAZ DRIVES)**
  http://www.iomega.com

- **IQS (*SAW*)**
  http://www.iqsoft.com

- **SYQUEST (EZ AND SYJET DRIVES)**
  http://www.syquest.com

- **TURTLE BEACH (SOUNDCARDS)**
  http://www.tbeach.com

270Mb external drive, and now use the AHA 1510A. This is perfectly adequate for the job, and I don't yet have the luxury of a PCI buss in my PC. If I were using a program like *Cubase* or *Logic Audio*, it would be more sensible to choose from the AHA 1542 range (or AHA 2940 range with a Pentium), but only if I could justify something like the new Jaz 1Gb removable drive. Otherwise, it would be still cheaper to buy one of the latest large and fast EIDE drives, which have very similar performance to SCSI drives (albeit with a higher processor overhead), and add a lower-end SCSI card for use with external devices such as scanners and backup drives, which don't need such high throughput or have to talk to other SCSI devices such as samplers. However, one thing is for certain — as our PCs fill up with more cards and drives, external SCSI devices will always offer a tantalising way of expanding sideways. [SOS]

Soft Options

STEINBERG *WAVELAB* v1.5 AUDIO EDITING SOFTWARE & DSP PLUG-INS FOR PC

How do Steinberg's DSP software plug-ins for their **WaveLab** *software, now at version 1.5, compare with the industry heavyweights? PAUL WHITE unearths his nastiest recordings to find out...*

Steinberg's general-purpose PC-based digital audio editing package, *WaveLab*, has now reached version 1.5 (see the full review of version 1.01 in *SOS* August '96), and is capable of editing samples or audio files of up to 24-bit resolution, dialogue, music and sound effects (though there's no dedicated sampler support). The use of a floating point algorithm produces a huge internal dynamic range, and the program is also very fast, very stable and comes with a number of useful software plug-ins at no extra cost (details of which in a moment). Version 1.5 also allows plug-ins to be used in real time, but unless you have a fast PC, the more DSP-intensive stereo processing will probably have to be done off-line. However, it is possible to run several of the more routine plug-ins simultaneously, and when processing off-line, you can apply several processes simultaneously, or even batch-process files, courtesy of some very clever background processing.

Further plug-ins are available as options (ie. you need to pay to obtain these; they don't come with

WaveLab), and some of these are very serious tools indeed — hence this review. Optional plug-ins so far include the *DeNoiser*, the *DeClicker*, the *Spectralizer* harmonic enhancer, the *Loudness Maximizer*, and the brand new *Magneto*, which is designed to emulate tape saturation. One of the plug-ins supplied with *WaveLab* is the *Grungelizer*, which is a real bonus. As the name suggests, this module 'grunges up' your audio, and is perfect for recreating vintage vinyl effects, complete with crackles, hiss and hum. I don't want to appear cynical (no, honestly, I don't!), but it occurs to me that Steinberg might have collected up all the sonic rubbish that their other plug-ins take out, and then repackaged it so that we can put it back in again! Actually, I'm kidding — this plug-in creates stunningly authentic vinyl sounds, and according to Steinberg, the dance market has been crying out for something like this for a long time. If this is an effect you need, this plug-in is *the* way to achieve it!

WAVELAB v1.5

The main difference between *WaveLab v1.5* and its predecessors is the RealTime engine, comprising a Master section (see Figure 1) with six series-connected, virtual insert slots. The window also has nice meters, sophisticated dithering (see the 'Dithering About' box), clipping counters, a real-time level display, and an indicator for detecting dropouts of as short as a single sample. The manual states that users with a Pentium PC slower than 90MHz can turn off the Master section if necessary, but depending on the plug-ins used, I'm told you can even get by with using a 486. If you try to run the Master section with a 486, you get a warning telling you that you're underpowered, but you'll be allowed to try anyway, and depending on the plug-in you want to run, you may well get away with it.

Any internal or external plug-ins can be dropped into *WaveLab*'s six effect slots, and the number that may be used at once depends on the power of the PC running the system, and on whether mono or stereo processing is needed. A Pentium 166, or, better still, a Pentium Pro 200 is recommended for serious real-time use involving the more demanding plug-ins, but in the context of a professional environment, such an expense is not unreasonable. However, you can do a real-time preview in mono, and then process the file — which doesn't take nearly as long to do as you might think — and what's more, you can preview several plug-ins running at once, before processing them in one operation! This is in direct contrast to Digidesign's TDM plug-in system, where if you run out of DSP farm power, you have to process the file using some plug-ins, shut those down, then reprocess using whatever other plug-ins you need. I'm also told that

Figure 1: WaveLab's new Master section.

running under Windows NT instead of Windows 95 makes the processing some 35% more efficient!

Bundled with *WaveLab* on the master CD-ROM are nine plug-in effects: *AutoPanner*, *Chorus*, *Echo*, *EQ 1*, *ReSampler*, *Reverb*, *Leveller*, *Tools 1* and the *Grungelizer*. If the purpose of all of these is not self-evident from the titles, don't worry, as I'll cover their functions in just a moment. The optional plug-ins are protected by a hardware dongle, though Steinberg are thinking about changing to a disk-based system. As it is, if you want to run all the optional plug-ins, you either need a short ribbon cable to allow the dongles to hang down the back of your PC, or you need to knock a hole in the wall to allow them to poke through into the next room!

Once the plug-ins are dropped into their slots — actually, it's more a case of calling them up from a pull-down menu than physically dropping anything — they operate as if they were connected in series with the input, feeding module one and the output coming from module six. However, whether or not you can use all six slots in real-time or not depends on what plug-ins you want to use — for example, you can have several EQs and a sample rate convertor running at once with even a fairly basic machine. The *DeClicker*, *Loudness Maximizer* and *Magneto*, on the other hand, really need a Pentium 166, though I'm informed that the *Loudness Maximizer* and *DeClicker* manage OK on a Pentium 133. The two plug-ins that really stretch your system are the *DeNoiser* and the *Spectralizer*, but even if you go the whole hog and buy a Pentium 200 to run them in stereo and in real time, the total cost is still way below that for any comparable 'audio cleanup' system.

Other than the real-time window and the bundled plug-ins, there are few obvious mods to the basic program itself, other than the addition of a Windows menu bar for accessing the RealTime Engine functions and the ability to read AU (Sun and NeXT) formats, as well as RAW audio files. From my own point of view, *WaveLab* desperately needs a playlist feature, and if anything, the new plug-ins just make this more important. However, at this point, nobody's really sure what direction Steinberg plan to take with this product in the future.

BUNDLED PLUG-INS IN DETAIL

• AUTOPANNER
The *AutoPanner* is a simple device designed to pan audio from left to right and vice versa under the control of a low-frequency oscillator. The modulating waveform may be a sine wave or a pulse wave, and in addition to LFO frequency, you can also set the width of the pan, and control the two output levels independently. There are no fancy sweep sync options, and no MIDI control.

• CHORUS
This is the familiar studio effect, but with the addition of a Glimmer parameter that adds stereo panning to part of the treated signal. The user has control over depth, rate, feedback and delay, as well as independent control over the level of the feedback signal. Stereo spread, mix and output level may also be adjusted. The result is a very usable and versatile chorus, though I can't envisage

using *Chorus* at the editing stage very often.

• ECHO
Another familiar effect, this time with control over delay, feedback, balance and volume. Each channel may be set up separately to create a true stereo effect, and it's possible to link the channels so that the output of effect block one feeds the input of effect block two.

• EQ 1
Here you get variable-frequency high and low shelving filters, plus a fully parametric mid section, and in case you need more control, you can load the plug-in into two or more slots, provided your PC's processor can handle the strain. This way, you can set each one to work on a different part of the audio spectrum. Though digital EQ can lack character, this one is very positive, and does much as you'd want it to.

• LEVELLER
Think of this module as a digital volume control, able to increase as well as decrease the level of an audio signal. The range is from -48dB to +12dB, and the purpose of the module is to make up gain lost by other processing, such as EQ.

• RESAMPLER
On the face of it, this is just a sample-rate converter, but it has the advantage of being usable in real-time as well as off-line. The sample rate is adjustable from 11.025kHz to 48kHz, and depending on the application, you can choose from three quality modes to save processing time at the expense of quality. Mode is used to select the target sample rate if it is to coincide with one of the standard rates supported by your soundcard.

• REVERB
Lexicon aren't going to lose too much sleep over this one, but it's good enough to be useful nevertheless. Nine parameters are available for user adjustment, including room size, decay time and several early reflections-related functions. At some settings, this plug-in can sound a bit 'ringy', but it's useful for adding ambience to dialogue or music. Considering no additional DSP cards are needed to run it, it's amazing that it's as good as it is!

• TOOLS 1
This is an odd but useful inclusion, capable of independently inverting the phase of either channel, ▶

Figure 2: The Grungelizer.

Figure 3:
The Loudness
Maximizer.

STEINBERG *WAVELAB v1.5* & DSP PLUG-INS

▶ or working in one of two M&S (Mid & Side) modes. One mode deconstructs stereo signals into their middle and side components, exactly as you'd get from a middle and side mic, while the second option takes the output from a middle and side mic array and converts it to conventional stereo. Interesting experiments include splitting a signal into its M&S components, EQ'ing or adding reverb to the side components only, then using the M&S decode facility to put the signal back into stereo.

Figure 4: The Spectralizer.

• *GRUNGELIZER*

This plug-in is designed to make pristine recordings sound like vinyl by adding crackles (with a choice of 33, 45 or 78rpm settings!), hiss, distortion, vintage EQ coloration and mains hum at 50 or 60Hz. I wasn't entirely convinced by the harmonic content of the mains hum — it wasn't quite angry enough — but everything else is absolutely authentic. There's even a time control which can set the playback character to anything between 1900 and the present day! You must try this — absolutely any audio can be converted to sound like a badly-looked-after vinyl record played back with a chipped stylus! Uniquely, this plug-in has a brown front panel, presumably out of respect for its effect on the sound.

Aside from its obvious applications in dance music and sample creation, the effect is also useful for post-production 'futz', where it may be necessary to fake a vinyl sound for film or TV. This would be a great plug-in, even at a premium price, but to get it bundled for no extra cost is a nice surprise.

OPTIONAL PLUG-INS

• *LOUDNESS MAXIMIZER*

All of the optional plug-ins can be used to process files of up to 24-bit resolution. The *Loudness Maximizer* combines level change with automatic, intelligent dynamics control to make a signal sound as loud as possible (and to make use of all available headroom), but without introducing undesirable side effects, such as gain pumping or dulling of transients. As with most of these plug-ins, the controls are deceptively simple. One fader sets the desired signal gain, and alongside is a meter showing the possible gain increase that can be achieved without clipping. Controlling the dynamic side of the process is the Density slider, and next to that is a meter showing the 'Desired Gain Done'. In effect, Density controls the relationship between the compression and limiting aspects of the process, and at higher settings, more of the gain reduction is achieved by compression than limiting, resulting in a higher average signal level. As its rather clumsy name implies, the Desired Gain Done meter shows to what extent the process has been able to provide the desired increase in gain.

A Boost button adds a further 2dB of increased loudness, regardless of the other settings, and the Limiter section is switchable in 10 steps between soft and hard modes. Hard settings work well with aggressive, rhythmic pop music, while 'softer' values are better suited to classical and acoustic music. High-resolution output level meters are included, showing the output level to an accuracy of 0.1dB. User settings can also be stored and recalled for future use. As well as making mixes sound louder, the process is useful for treating 16-bit multimedia audio files before reduction to 8-bit.

This plug-in certainly makes signals seem a lot louder and punchier without increasing the peak ▶

STEINBERG *WAVELAB* v1.5 & DSP PLUG-INS

Figure 5: The DeClicker.

▶ level at all, and what's more, you can set the target peak level at any value you like. Most impressive is the lack of traditional compression or limiting artifacts — something to do with the cunning 'look-ahead' algorithm. To do this with analogue, you'd need a chip that could see into the future! I suppose the nearest approximation to this plug-in is the *L1 Level Maximizer* from Waves, and I have to say that I feel the *L1*'s controls are a little more logical — but that could be because I've been using it for quite a while now.

• SPECTRALIZER

Spectralizer is really a harmonic enhancer that works entirely in the digital domain, providing independent slider control over added second and third harmonic content. As with a conventional enhancer, a frequency control allows the user to set the frequency above which harmonics are generated. As the second harmonic is one octave above the fundamental, setting a filter frequency

Figure 6: The DeNoiser.

of 3kHz, for example, means that the lowest added harmonic will be 6kHz.

A Kick switch adds extra processing to transient sounds, and if you're not exactly sure what's going on, a neat block diagram of the process can be called up next to the control window. All the

variables in this module are controlled by sliders rather than by rotary controls, and once again, favourite settings may be stored.

Subjectively, the process is similar to an analogue enhancer, but it's quieter and more controllable. The sound remains warm and detailed, but there's no separate low frequency control, which might have been useful. This is a useful tool to have available during post-production audio sweetening, where you don't want to leave the digital domain. It's also useful for brightening up samples.

• DECLICKER

Noise removal software has been around for some time, but the results that can be achieved vary wildly depending on the type of process being used. No system is completely successful at tackling all types of noise at once, so the process tends to be split up into de-clicking, de-noising, and de-crackling. De-noising is used to remove broad-band noise such as tape hiss, while de-clicking tackles individual spikes, such as digital glitches or vinyl scratches. De-crackling is probably the most complex and specialised process of all, and is intended to clean up dense surface noise from vinyl. De-crackling is not yet an option for *WaveLab*, though dramatic improvements to vinyl recording can be made using the de-clicking and de-noising processes. Perhaps now I'll get around to putting my old vinyl records onto CD-R before they're completely unfit to play...

No specific details are given as to how de-clicking works, but the process really comprises two parts. The first task is to decide what is a click and what is a meaningful musical transient, and I'm told that a certain amount of modelling is used to identify genuine clicks. A user-adjustable threshold provides the user with control over the severity of the process, while Audition mode allows the scratches to be heard in isolation. This is immensely useful, as it's quite obvious whenever meaningful material is wrongly identified. Once you're sure that only scratches are being picked up, you can run the system in real time or apply the process to the file in question, and as if by magic, the clicks are removed, and their space filled with plausible material calculated from whatever is happening either side of the click.

To further fine-tune the process, the user can choose from four levels of processing, which give priority either to the amount of noise removed or to the quality of the finished audio, and there are three further settings for Old, Standard and modern material, roughly equating to 78s, vinyl at 33 or 45 rpm, and material with digital glitches. Of course, you can try any of these settings on your material to see what works best.

How well does it work? Better than you could possibly expect! Quite badly-scratched vinyl cleans up very nicely, digital spikes or clicks are dealt with, and even clipping distortion is made more tolerable. Setting up is extremely easy, thanks to the Audition mode, and the quality of restoration is quite simply amazing. If there are better systems

around, they can't be that much better, and yet most are a lot more expensive! As with the other plug-ins, you can run in real time or off-line, and settings may be saved.

• DENOISER

De-noising is invaluable for reducing the level of tape or equipment hiss, especially if you're remastering old recordings, but most simple systems suffer from noticeable side-effects. For years, I've been using Digidesign's *DINR*, but realistically, you can only expect a maximum of 4 or 5dB of noise reduction before you start to hear the noise floor being modulated by the noise removal filters.

Unlike *DINR*, which really needs a sample of 'noise only' recording before it can start work, *DeNoiser* uses a more advanced system based on an intelligent, adaptive algorithm to constantly reappraise the noise. When the file in question is run, a small window shows the dynamically varying noise spectrum, over which is superimposed a threshold line. This is the main user adjustment, and once the threshold is pushed just above the central portion of the noise spectrum level, processing begins. There are only three user controls — one for the threshold level, one for the amount of gain reduction and one for ambience. The Ambience control is designed to stop the process robbing ambient sounds of their character during the decay of the sound, and I have to say, it really works.

Once again, files of up to 24-bit resolution may be processed, and two different settings can be stored for direct comparison using the A and B buttons. Used carefully, the quality of processing is excellent, and in practice, you can get between 12 and 20dB of hiss reduction without messing up the sound, though the exact figure depends on the quality of the material being processed. Interestingly, when I tried the process on a sample of vinyl record recorded onto DAT, *DeNoiser* wiped away most of the steady-state surface noise, leaving only the scratches and the worst of the crackles. As you'd expect, the *DeNoiser* isn't designed to salvage horrendously noisy material, but even salvage jobs work out better than they have any right to. On the other hand, decently-recorded material with only moderate noise contamination cleans up beautifully.

• MAGNETO

This plug-in turned up in late beta form, just before the review, and with no manual. It was also the only one of the plug-ins that wouldn't run without glitching in stereo/real-time mode on a Pentium 166, but on contacting Steinberg, I was told that this was due to the beta version not being fully optimised, and that the production version would be significantly faster.

The controls are pretty much self-explanatory. Featuring two nicely vintage VU meters, complete with extra-large red overload areas, *Magneto* is designed to emulate the effects of tape saturation at both 15 and 30ips. The meters can monitor the input level, the output level, or the virtual to-tape level, and an HF control allows a little fine-tuning of the top end of the spectrum. Other than that, it's simply a matter of overdriving the

Figure 7:
The Magneto.

virtual tape by whatever amount produces the right subjective result, and though the process is fairly subtle, it's actually quite authentic.

CONCLUSIONS

WaveLab has gained a lot more credibility thanks to the high-powered optional plug-ins, developed for Steinberg by Spectral Designs. All the bundled plug-ins are clean and competent, (other than the *Grungelizer*, which is delightfully dirty and competent!), but for my money, the *DeClicker* and *DeNoiser* take the top prizes. These tools can produce genuinely professional restoration results, yet they remain simple to use and, above all, affordable. As I said earlier, I've been using Digidesign's *DINR* for several years now, but *DeNoiser* makes it look like a packed lunch! And, while I personally don't often have scratchy vinyl to deal with, the *DeClicker* is exactly what you need for sorting out those DAT transfers that have accumulated inexplicable clicks along the way.

The *Grungelizer* is worth a mention for its audacity, the *Leveller* is just what's needed for squeezing the last drop of subjective level out of an audio file without making it sound compressed to death, and the *Magneto* can add analogue warmth to a DDD recording. In all, this is a beautifully-presented, very slick suite of programs, and though the processor-intensive *DeNoiser* can only be run in real-time if you have a 166MHz Pentium computer or above, I can't blame the software — processing of this quality takes power, and if you have to wait a while for off-line processing, that's just the way it is.

If you need powerful, affordable audio processing, then you'll be doing yourself no favours by ignoring *WaveLab* and its plug-ins. [SOS]

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Finnish company Genelec have a long-established and fine reputation for building monitor loudspeakers of the highest quality. Their enormous range of mainly active systems includes designs to suit every conceivable application, from monster control room systems down to their latest and smallest monitors, the 1029As, with their dedicated sub-woofer the 1091A.

In my experience, the three key elements of any Genelec monitoring system are quality, accuracy and consistency, which pretty much says it all! The company pay great attention to every detail of the design, from the amplifiers and active crossover electronics built into most of their systems to the robust construction designed to withstand the worst possible abuse.

The 1029As are designed specifically for nearfield monitoring applications, and are particularly suitable for Outside Broadcast trucks, surround monitoring systems, meter bridge monitoring and budget or home studio use. They share the same sonic characteristics as the larger Genelec systems, such as the 1031As, and I have to say I was stunned at the overall performance of the complete 1029A and

for a standard Omnimount wall bracket. The base of the speaker also has a 3/8-inch UNC threaded hole, which allows the speaker to be fixed to the top of a mic stand.

The 1029A has two drivers, a 5-inch woofer and a 3/4-inch tweeter. The woofer is capable of handling signals down to a very respectable (for the cabinet size) 65Hz, thanks to the active equalisation and the ported 4.5-litre cabinet. The tweeter is mounted at the focus of the aforementioned DCW, which is used to give a uniform high-frequency dispersion, and also to improve the performance of the complete system at the crossover between the two units.

The active crossover uses a pair of complementary band-pass filters with slopes between 24 and 32dB/octave, the lower section passing frequencies between 68Hz and 3.3kHz while the upper section handles 3.3kHz to 25kHz. The outputs from the crossover are passed to a pair of 40W amplifiers, which drive the loudspeakers directly.

The amplifier design has very low distortion (0.08%) and fast slew rates, and is fully protected to avoid overloading the drive units. I was amazed to

Finnish company Genelec have long been admired for their high-end professional monitors, but have only recently started to fix the project studio market in their sights. HUGH ROBJOHNS checks out their smallest and most affordable active monitors yet to see if quality has given way to cost considerations...

Perfect Finnish?

GENELEC 1029A ACTIVE NEARFIELD MONITORS & 1091A SUB-WOOFER

1091A sub-woofer system. It might not be able to generate the last few dBA of volume of some of the largest systems, but other than that, this is a really powerful set of monitors.

THE 1029As

I don't know if it's a Scandinavian trait, but Genelec like to give all of their monitors very boring (and often confusing) numeric identifications. The 1000 series all share a similar look, with a narrow vertical ported vent on each side of the tweeter, which is at the focus of a subtle convexed surface, called the DCW (Directivity Control Waveguide).

The smaller the Genelec monitor number, the smaller the cabinet size, and the 1029A is the smallest of Genelec's current offerings, with a cabinet measuring 247mm (h) x 151mm (w) x141mm (d). The 'A' at the end of the number simply indicates the active nature of the monitor, with two 40W amplifiers and line-level crossover built in.
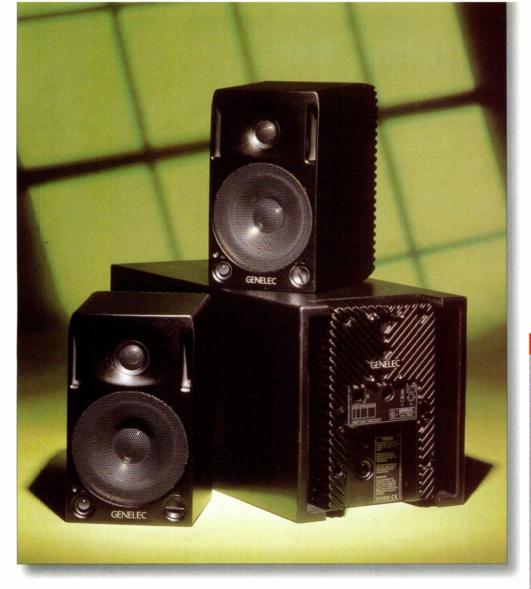
The speakers are astonishingly heavy, weighing in at 5.7kg, largely thanks to the unusual use of cast aluminium in the cabinet, but the large amplifier heatsink and the standard magnetic shielding probably contribute significantly to the weight as well.

Mounting the 1029As is a doddle, since three fixing options are provided as standard. The heatsink on the rear panel is equipped with three keyhole slots which accept 4mm large-headed screws for wall mounting, as well as fixing points

discover the sound-pressure figures for the 1029A, which suggest that they are capable of generating peak levels of 110dB SPL at 1 metre on typical musical material, with an even more amazing long-term RMS output of 98dB SPL! These are impressive figures for such a diminutive loudspeaker, and ones which confirm the ruggedness of the design.

AUDIO CONNECTIONS

The rear panel has three sockets, two for analogue audio and the third for the mains. A standard IEC mains socket is accompanied by a voltage selector switch, and a moulded mains cable is supplied (although in the case of the review model, these were of 2-pin European design). The input and mains sockets are all arranged so that the leads hang vertically downwards, which will allow the speaker to be mounted on the wall very easily. Although the XLR is latching, and the TRS is pretty solid, I'm not so sure that the IEC mains lead will remain in place for ever — perhaps a retaining clip would have been a good idea.

The two audio sockets are both balanced, one a 3-pin XLR and the other a quarter-inch TRS jack. Although balanced connections are preferred, unbalanced operation is possible, provided the plugs are wired to connect the cold side of the balanced audio circuit and the earth together. As with other Genelec products, the rear panel has plenty of silk-screened information about connector wiring and so forth, so even after you have lost the instruction

leaflet, the key information is still available.

Either of the two sockets on the rear of the 1029A may be used as an input in a stand-alone mode, although when it's partnered with the 1091A sub-woofer, the XLR is taken as the input, and the TRS jack provides the onward connection to the sub-woofer. This aspect of the design differs from that of the larger Genelec systems, which route the audio through the sub-woofer first (for low-frequency filtering), before passing it to the satellite speakers.

Aside from the dip switches (see the 'Dip Dip Dip' box elsewhere in this article), all the 1029As' controls are on the front panels, and consist of a circular rocker switch to turn the unit on, and a rotary level control to set the input sensitivity. There are no calibrations on the volume control, so matching a pair of 1029As is a matter of trial and error; but they will accept a wide range of input levels, which makes matching to professional or semi-pro equipment very straightforward.

1091A SUB-WOOFER

The 1091A sub-woofer is specifically designed to work with the 1029As. It is surprisingly compact at 505mm (h) x 251mm (w) x 230mm (d), and is finished in black with the amplifier heatsink on the top surface between a pair of large slotted port openings. The amplifier heatsink appears to be identical to that of the 1029A, and even has the same mounting options (none of which may be used, of course — the 1091A is strictly a floor-mounted device!). Most installations would probably have the sub-woofer standing upright with the vents facing upwards, but an acceptable alternative is to have the box lying down with the ports firing out to the side.

The sub-woofer uses a single 8-inch bass driver which is installed in a 15-litre ported cabinet, the whole system handling the frequency range between 38Hz and 85Hz. The amplifier unit is configured as a 70W unit with the same protection circuitry to look after the drive unit and similar performance figures. The maximum sound pressure level that the 1091A can achieve is 103dBA — more than enough to have the walls shaking!

The electronic crossover uses a 24dB/octave filter above the crossover frequency (85Hz), and has an 18dB/octave filter below about 40Hz to remove subsonic signals which would merely sap amplifier power without producing useful audible output.

Audio connections are the same as on the satellite speakers, with an XLR and TRS jack, and both are again balanced, of course. This is my only real area of complaint for the entire system, since it requires a rather strange collection of interconnecting leads. The inputs to the satellite 1029As are on XLRs, but one sub-woofer feed requires a TRS-to-XLR (m) lead, while the other needs a TRS-to-TRS lead. Not really a problem, I suppose, but it was an initial frustration to me because all my balanced leads use XLRs, and I had to make up some converters before I could get the

GENELEC 1029A & 1091A

▶ system going. Perhaps Genelec (or UK distributors Project Audio) could consider supplying a suitable set of leads with the sub-woofer?

The sub-woofer is a doddle to set up, especially since the most effective thing to adjust is its physical position in the room. A set of DIP switches allows the input sensitivity to be balanced to the satellites over a 12dB range, which helps to compensate for the bass-boosting effects of placing the sub-woofer near walls or corners. The instruction leaflet gives very good and clear advice on how to position the sub-woofer; I found its most effective location to be lying parallel to a wall between the two satellites with about 6 or 8dB of gain attenuation.

I knew from previous experience with Genelec monitors that you have to be very careful about matching levels between the satellites and sub-woofer (although this applies to *all* satellite/sub-woofer systems). It is very easy to set the system up with impressive but excessive bass output — the end result of which will be mixes that sound thin and bass-light on other monitoring systems. Remember, the sub-woofer should not draw attention to itself if it is properly set up, but switching it off will make its contribution clear! The other things to watch out for are peaks or dips at the crossover frequency. The supplied instruction leaflet explains how to match the units properly by adjusting the polarity of the sub-woofer to ensure that its signal is in the same polarity as the main speakers. This is simple to do, but has a profound effect on the end results.

When everything is set up, I like to listen to tracks that have 'walking' bass guitar lines to check the overall performance. The bass guitar should sound equally loud whatever notes are played, without any obvious loud or quiet pitches. Inevitably, the room itself will make this almost impossible to achieve because of standing waves (or Eigentones) where the room's dimensions relate to specific wavelengths and the whole room effectively resonates at those frequencies. Moving the sub-woofer or listening position a foot or so can often make dramatic improvements to these standing waves, so experimentation is very important in getting the best possible results.

The advantage of satellite/sub-woofer systems is that, by divorcing the bass driver from the rest of the system, you can find the best in-room position for the sub-woofer without compromising the stereo imaging (and vice versa). This is just not possible with full-range stereo loudspeakers, where the position for best imaging often puts the bass drivers at inappropriate positions in terms of standing waves in the room.

PERFORMANCE

These baby Genelecs are quite simply stunning! I have always been a fan of the Genelec sound, which is clean, neutral, accurate and well-balanced — everything a decent monitor loudspeaker should be. The 1029/1091 combination certainly lives up to the reputation of its larger and more costly siblings, yet in a surprisingly small package.

You really have to hear these diminutive loudspeakers in action to believe the scale and quality of the sound they generate. Stereo imaging is crystal clear, very precise and stable over a reasonably wide working area, with good depth and width. When they were set up as nearfield monitors, the sound stage extended well behind the monitors, and gave a very creditable, almost three-dimensional quality to the music. When they were set with a flat frequency response, I found the 1029As a little too 'sharp' for my tastes — not obviously bright or 'tizzy', just too detailed or analytical, perhaps. Many of you will probably think them fine when set to flat response, but after years of listening to rather smooth BBC-designed loudspeakers, I found the -2dB treble tilt a more acceptable balance for my tastes. In this condition, the speakers were just as tonally correct and precise, and still capable of revealing technical flaws in recordings, but less fatiguing and generally easier to work with over long periods. Although the 1029As tend to sound a little bass-light on their own (and I would definitely recommend they be used with the matching sub-woofer), they still managed to produce a very creditable sound, given their overall size and power rating.

Apart from the bass roll-off setting needed to match the satellites to the sub-woofer, I found no need for the bass tilt settings, although I'm sure they would be useful if the speakers were mounted close to rear walls. When the optimum position had been found for the sub-woofer, the bass integrated seamlessly and provided a very solid foundation for any instrument with a low-frequency content. Percussion of all kinds gained a weight and solidity from the 1029A/1091 system, and even singing voices became much more natural and believable — it is surprising what an enormous difference the extra octave at the bottom makes!

A good indicator of the quality of a monitoring system is how loud it actually has to be before you perceive it as being loud. Our brains use the distortion content of the signal as a measure of the actual loudness — the cleaner the source, the louder we will listen. The 1029As have no problem on this front: what appeared to be a comfortable listening level turned out to make conversation difficult and the system was quite capable of making all manner of things in the control room rattle when turned up. Even though these monitors are small, they can produce a very big sound indeed. In short, the 1029As are easy to install and simple to set up, and provide performance which is little short of stunning. If you are in the market for a monitoring upgrade, these need to be at the top of your shopping list! **SOS**

Hugh Robjohns is a lecturer at the Centre for Broadcast Skills Training at BBC Wood Norton. The views expressed in this article are the author's own and are not necessarily those of his employer.

The Sound

DAVID WEST

WEST PRODUCTIONS

West Productions' innovative work on the audio post-production of cult TV series The X-Files netted the company two Emmy awards in 1996. PAUL WHITE talks to company founder David West about the sound element of major TV programmes, including X-Files creator Chris Carter's new series, Millennium.

West Productions was founded by David West over a decade ago. The company's first premises were West's own garage: now they occupy 15,000 square feet of prime real-estate in Burbank, California. With 13 Timeline Studioframe workstations, digital ADR (Automatic Dialogue Replacement), Foley stages [where sound effects such as footsteps on different surfaces are re-recorded for dubbing] and three Dolby SR-equipped re-recording stages, this Emmy award-winning post-production company has worked on a huge number of major TV shows, including *Hill Street Blues*, *The Wonder Years*, *LA Law*, and currently, Chris Carter's *X-Files* and *Millennium* for Fox TV

As viewer expectations increase and the year 2000 approaches, the soundtrack has become a major part of the 'multimedia experience' that is television. I was keen to know how much creative freedom a company such as West Productions is allowed when working on today's major television sound productions. David West:

"My concept is that you bring your project here because you think my opinion is important and valid. As a mixer and editor, I take it as part of my job to add to a picture as I see fit, in order to offer the client and the director what I perceive their vision to be and beyond. Rather than just going in to provide a mix, I feel it's my responsibility to produce something more than just the obvious. The music is at the executive's/producer's discretion. I've been on the stage and fought to keep the music in, or sometimes I've fought to keep the music out, if it's right for the picture, because we are the last stop, and everybody else's job is over when the picture comes to my company. The control always rests with the guys paying the bills, but we do offer a lot of creative input. Every sound guy goes to the spotting-session [for the placement of spot effects in sync with the picture], where he sits with the director or the producer, who says what he wants, but if you can satisfy that, and then deliver a bonus on top, that's what makes the difference.

"There are tricks that people have been using for years in features, adding this sound to that

West Productions' Burbank HQ by night...

sound. You can layer up stuff — we've got cowboys and indians going in one scene but you'd never know. In every show there are lots of creative decisions — there were a couple of flashback scenes in *The X-Files* and we didn't know exactly how to play them. The producer had just visualised it with a voice-over, but by giving it a little bit of reverb, a little bit of the old tube sound, it works! I follow no rules: I'm just true to the movie."

JOIN THE Q
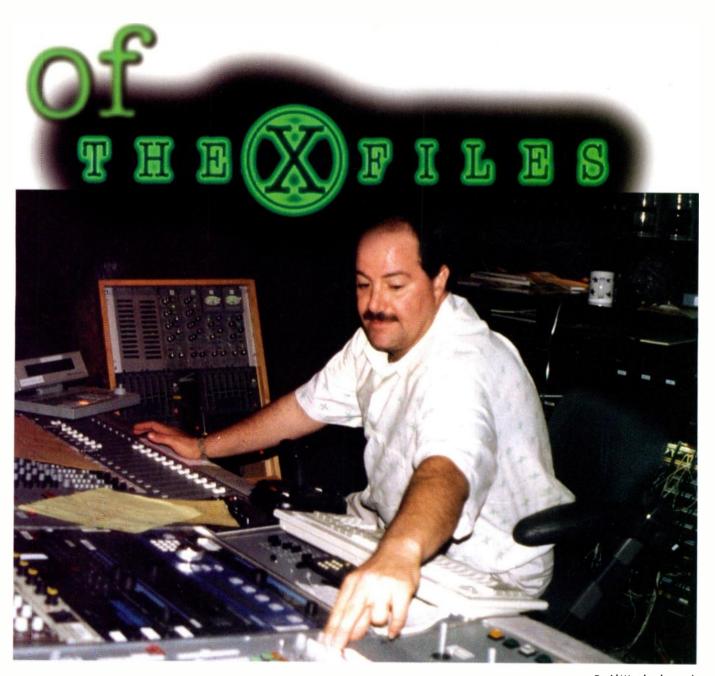
I understand that you're a keen proponent of QSound. How do you use it?

"I use it on everything! We have a nice manual Neotek mixer with a lot of inputs, and QSound gives me automated panning. It delivers a big bang for the buck — you can take a static track and give it some kind of movement. I put stuff on rotating panners all the time; I'll often use lots and lots of layers — 12 or so loop groups — but where can you pan them? I put them all on different rotating panners in different directions — and now I've got everything going around the room. It's one of those random acts of creativity that makes things work.

"We do the same thing with effects: take a couple of mono wind tracks, put them on rotating panners, take another track, put it on the surround channel, and those three mono tracks can sound like 10! At times, what QSound does may alter the quality and some engineer may say he hears some phasey, flangey stuff, but who cares? So long as the person listening to the TV can feel something happen, have their attention grabbed and focus on my work — it's OK."

Is there any technical conflict when using QSound with Dolby surround?

"We work in Dolby surround all the time, and I haven't had any real problems. If we have a slight mono compatibility problem, I say — buy a new TV! The people who pay my bills want it big, butch and awesome. The only way I can satisfy them is to do anything I can to get more volume within the level standards specified by the network. We're used to all of this dynamic range now, but with TV sound, you can't use it as effectively as you can in film. Now we're flat from 50Hz to 20Hz, and it's all dynamic, and it's real hard work giving the client the excitement and beef *and* providing the network with a track they can

# of THE X FILES

broadcast. QSound gets things moving around for me, it gives me the ability to take a few tracks and layer them so they've got spatial action.

"On a recent show, I had some old Martin Luther King dialogue, and I went through and did as much noise work on it as I could; I put it through the [Aphex] Compeller, then I put it into my reverbs. There are three main reverbs I use — Lexicon 300, PMC70 (which includes some of the useful effects not found in the PCM80), and PMC80, plus an Eventide DSE 3500, which has a lot of programs in it that we've created here. I put the dialogue into one of the Lexicons so that I could use some of the spatial processing to work up some low end and put some warmth back into it, then took the output from the Lexicon and put it into QSound so that I could localize the source. QSound works better than conventional quad panning a lot of times for me, but I have to be careful of the compromises it makes with the bottom end.

"I learned from my dad, a film guy, and basically he had to deal with 80Hz to 8kHz. He sat around the console all day messing with the tone of the dialogue by adjusting at 1kHz and 2.2kHz — there were less options then. I concentrate on trying to produce a warm, smooth, round consistent dialogue track. Give it some bottom, some beef and some localisation and perspective. Every sound guy has problems and they all have their own way of solving them, but we have to find a way to work with what they bring us. We've tried re-dubbing dialogue, but for *The X-Files* we want to use the original dialogue with its performance life and punch. We've had to work with very difficult sounds — but when we've finished with it, the network's approved it and the show is a major success. In many cases, it's better to live with the quality of the original sound rather than try too hard to fix it. It's a case of having the script and performances stand up."

THE MAGIC TOUCH

"I've worked with the best people in this business. These guys can walk on the soundstage and they don't care so much about the technical noise — they'll say, 'That music cue is 30 seconds out of

▶

▶ sync.' And the composer says 'What are you talking about? I wrote this!' And when you move the cue, a whole new movie appears and everybody goes, 'How did he do that?' They're not worried about the crickets in the background or all the bullshit we've been working on all day — they see the movie, the emotion, the big picture!

"We've had some big names in here, and when you get four or five big egos in the same room, it's not about mixing any more — it's about keeping people happy. In a lot of cases, you have to be able to let go of your own ego and buy into the vision of the people you're working with. Not only do you get the show done, you also get the respect of the people you work for."

MILLENNIUM

I hear you're working on a new show called *Millennium*, which comes from the same stable as *The X-Files*. What special challenges does this show present?

"*Millennium* is a really exciting turn — Chris Carter wrote it and David Nutter directed it. I'm hoping that the pilot for *Millennium* will be our Emmy next year. We explored the entire realm of creative sonic recording, and I think the pilot soundtracks are the best we've done. Yet nobody's heard it right! The network ran the first eight minutes in mono, the LA theatre we ran it in had tremendous problems with double-decoding a Dolby A print — it has yet to be heard outside a dubbing stage the way it is. Even so, it stood up on TV and Lance's [Lance Henrikson, lead character in the new show] voice is brilliant — he's got this wonderful voice with a thick, round, raspy flavour.

"*Millennium* is darker than *The X-Files*, and it has more of a focus, although I've only seen three shows, and who knows what Chris will come up with? Every aspect of it has been different and fresh — I don't know how Chris is able to keep creating so much genius on two shows with so little time.

"At our end, we're doing the same things — we're using QSound, but we've got different supervisors and we're taking care to look at it in a different way. All in all, every hit show comes down to the script — no music or sound effects can make a script work. It's the same as with recording music: can you put enough reverb on my voice to make me sound like I can sing?"

PROBLEM SOLVING

What kind of technology do you use for flying in effects — do you rely on samplers or hard disk recorders?

"The process I have for solving any problem is first to discover what the real problem is. I might have a guy who, for whatever reason, has thought of something he wants to do right now. He wants to throw in a new line, and I don't want to send him down to the ADR stage to do it, then come back up to my stage, because the moment is lost. Our task is to do the job in the fastest way — to get the sound into our system and put it in the right place in the mix ASAP. My head engineer came up with a method from many years ago — we run a half-inch, timecoded tape along with the mix, it has a dedicated buss, and we can bang sounds into there and get instant access. We don't have to open files, we don't have to crash our systems. We're putting so much information through our systems that we don't want to talk to them — we just want to change the reels and let it be. For me, any solution for a film, be it on 35mm, audio in cassette, 4-track, Pro Tools or a sampler, is the right solution, as long as you're really clear what you're trying to accomplish. The digital medium should be used as a tool in making successful films, not as a complete solution."

WORKLOAD

As the company grows, do you find enough time to keep doing work you want to do?

"As long as I don't sleep I'm OK! Work is my passion. However, I'm not getting enough time to explore other avenues for my company. I'm mixing 60 hours a week most of the time, and I live a long way from here. I have an excellent team, including our president Dave Rawlinson. We've got around 55 people at the moment, all of whom are given a great deal of responsibility. Delegation, communication and co-ordination are the key to making this work, and as I expand, I'm going to need more good people. We need people who love what they do. There are so few people who really know what they want to do and who are willing to put in the effort to get there.

"I love to expose and train new people in this business. I started with the kids from Hugh's Market who were stocking shelves. It was a new job, an exciting job, and they loved it. As a team we succeeded. My future relies on new blood, anxious to be the best or make the most of an exciting opportunity."

What do you look for when you take somebody in?

"I look for life and zest — people who look me in the eye. I want people who are on the same page! Desire followed by consistent action — I can show up, I listen, I learn, I grow, I explore, I get my job done and save your ass. That's all it takes — everything you have and just a little more.

"People should be prepared to work endless hours — I'm here endlessly, I'm signing the cheques, I've been in the business for 20 years, and I'm here. I sleep on the floor trying to get stuff done. I'm working till two o'clock in the morning all the time. People have to sign up for the fact that it's going to take their life. We are prostitutes to this business; clients will pay us any amount of money to work all night, with or without food, until we drop. That's what the business is about — unless you're on the screen, you're disposable. If you sign up for that, you can become the greatest thing there ever was — for a little while."

David with Sound Supervisor Thierry Couturier, receiving Emmy awards for West Productions' sound work on The X-Files.

SOUND EFFECTS

How do you come up with original sound effects?

"Record, record, record. File recordings always benefit a picture, often even by mistake — you never know what you'll get. I've had some crazy things happen to me when I've gone out to record. I've been attacked by turkeys, chased by police... And in the end they all turn out to be absolute gems — accidentally. Plus, new movie, new sound effects always made sense to me.

"You have to look for things that aren't obvious. Anybody can go out and record a gun shot — people tell you what they want, and theoretically you can figure it out. It's when they say they want a new spaceship sound that you have to be creative. Or you go out to get a better gunshot sound by mixing something in with it, like a backwards lion growl or firecrackers. In a lot of places, doing nothing makes something different — there are places when we've used just the reverb return on gunshots, or used a close-up shot of the barrel and just had the sound of the bullet going through the rifling. Anything that's not expected can be right. We don't do much with synthesized sounds. I leave that to our composer, synthesizer assistant or sound designer."

MARK SNOW

SCORING *THE X-FILES*

Mark Snow's quirky yet haunting X-Files theme tune was last year's surprise worldwide chart hit, and his atmospheric scoring plays an important part in setting the mood for this darkly compelling television phenomenon. DEREK JOHNSON discovers that the music is out there...

When talking about the sound of the *X-Files*, there's one very important element that deserves separate consideration: the music. Soundtrack composer Mark Snow's simple yet effective theme grabs the attention and sets the mood for the episode to come — and that episode is likely to contain more original music than the average TV show, with about 30-35 minutes of music in each 45-minute programme.

I first asked Mark about how he got the job of scoring *The X-Files*: "Well, one of *X-Files'* co-executive producers, RW (Bob) Goodwin, is an old friend of mine, and he suggested me to the show's creator, Chris Carter, who I didn't know. Chris and I both live on the west side of Los Angeles, so he came over, saw my studio and listened to some of my music. But he played it very close to his chest, so I had no idea of what he was thinking. He visited twice, and it was the same both times: very nice, very respectful, but he left saying 'We'll be in touch' or 'I'll call you...' So I was thinking 'No problem, this will happen or not'. I didn't get a sense of his intentions one way or the other, and two weeks later, I got the job."

What kind of brief did Chris Carter give? "At first, he didn't want story-telling in the music. He didn't want obvious melodic, traditional, over-the-top scoring. He wanted very supportive, very sustained atmospheric stuff that didn't get in the way."

Carter provided a temp reel — a copy of the pilot with a soundtrack made up of music from other sources. "It was sort of my direction to go in, and I did that at first. It worked out very well, and set a tone for the show. As the shows progressed, I definitely think the music has become more involved, more musical, more melodic, more emotional, but still in that honest, minimal approach, which I think is the key to it."

The X-Files was almost an instant hit; was Mark aware of what was coming, or did he see it as just another job? "It felt a little bit better than just another job, working with a really intelligent group of people. I thought it could be successful, but I didn't think it would turn into this! Especially — selfishly — when the theme became a big hit. The music got to be so noticed; I never thought that would happen."

The Emmy-nominated Snow has a long list of pre-*X-Files* credits, including TV shows such as *Hart to Hart*, *Crazy Like a Fox*, and some episodes of *Cagney and Lacey*. Mark's traditional, orchestral background is further exemplified by his time at New York's Julliard music school. He studied oboe there, and still professes a great respect for baroque and other pre-classical music. His musical interests pass by the romantic era and get straight into the 20th century, with the likes of Schoenberg, Stravinsky, Ravel, and Bartok: "Maybe I've become more mature about these things, because I certainly respect Brahms, Mahler and Bruckner and so on, but there's something inherently schmaltzy about their music, which at times doesn't quite feel honest — in today's sensibility, that is — whereas some early music is so potently beautiful in its simplicity." Mark's musical background isn't completely restricted to the classical: several years at the beginning of the '70s saw him touring and recording with the ▶

X-Files stars Gillian Anderson and David Duchovny.

▶ cult-ish New York Rock and Roll Ensemble, which also featured fellow soundtrack composer Michael Kamen.

ALL GEARED UP
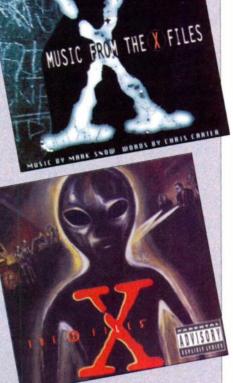
Despite his extensive experience of working with real orchestras, Mark does all the *X-Files* music at home, in his private studio. "First of all, there isn't the time to do a live orchestra score with this show. The copying, orchestrating, getting the musicians and so on, would be just impossible." Mark's equipment of choice is a Synclavier system: "About six or seven years ago, I got into the Synclavier (with Direct-to-Disk module). That's still my main piece of gear, and I find it the most elegant, fast, best-sounding thing available. People think that it has a 'sound', and it doesn't!

It's just a storage device, an architecture and a digital sampler that's so fast and immediate. The sound is what's in the library, what's on the disks. People talk about all of these samplers that cost nothing compared to the Synclavier; well, that's fine, but this is still by far the best. Actually, even if there were months to do each score live, working from here is still preferable, because the Synclavier provides a marvellous collection of regular acoustic instrument sounds, in combination with all the atmospheric, textural things."

But, of course, New England Digital, the company behind the Synclavier, is now no more: "It was unfortunate, but I've had very good luck with my machine: it hasn't broken down too much at all. And now there's a company called Demas, based near the original NED site in New Hampshire, that is offering Synclavier support and software updates. Demas even employs some ex-Synclavier people."

Despite having a Mac in his studio, Mark actually also sequences with the Synclavier, in a linear fashion: "It's not a hugely powerful sequencer, but luckily I don't have to do a lot of dance arrangements or pop things, so it's just fine."

The Synclavier system may be at the heart of the studio, patched through a Soundcraft Sapphyre LC mixer, but that's not it for "MIDI boxes", as Mark refers to them: the studio also accommodates a Roland S760 sampler, Kurzweil MicroPiano module, an Emu Proteus 1, 2 (source of the famous *X-Files* whistle, in combination, apparently, with a sample of Snow's wife whistling!) and 3, Emu Morpheus, Korg M1R and Wavestation SR. But these are adjuncts to the core sound produced by the Synclavier and its large library: "I have all these great samples on the Synclavier's magneto drives." Mark doesn't do much sampling of his own: "There are so many sample disks available that I have people transfer them onto Synclavier format." And is there anything else he'd like to add to the studio? "Well, the only thing that means anything to me is new sounds or new combinations of sounds. All I care about is if somebody has a great new sound library. That's what I'm looking for. Since my background is music, rather than electronics and technology, I just know as much as I have to. There are so many other composers who know about sampling rates and all that technical stuff, but none of that really interests me. So apart from new sounds, I could maybe upgrade the RAM and the number of voices on the Synclavier..."

THE FINAL SCORE

I wondered if Mark's classical background helps him to create more convincing electronic simulations — anyone who has listened to an *X-Files* soundtrack will vouch for its organic feel, even at its most impressionistic. "That's a really astute question, because in this day and age where these machines are so widely available, anybody can be a composer. But coming from an acoustic background, I tend to hear things in a very human,

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SINGLE FILES

The single version of the *X-Files* theme last year proved to be a surprise hit, so I asked Mark what prompted the release. "I think it was the record company's idea. Someone called me up and said: 'We need a four-minute version *today!*' I said 'I'm in the middle of doing the show music!' They said 'Put it aside, you gotta do it!' So I just elongated the track, added another section with the whistle sound, and made the strings do some oddball triads over the top — I must have done it in half an hour. I just wanted to appease these guys at the label and get back to work. The four-minute version originally appeared as the first track on the *Songs in the Key of X* CD, and it was just luck that Warner Brothers thought it would be good to release it as a single. They were just amazed that it took off. To me, as a musician, it was hilarious: here's a hit record that has no singing, no *bass*, no *drums*, no guitars, and that stays in D minor for four minutes! It's unbelievable."

And the single version of the theme tune, along with several of the dance remixes, did rather well worldwide — so well that there has been talk of a follow-up, not to mention bizarre requests for European tours. What did Mark make of the remixes? "I think 70% or 80% of those remixes are pretty good. But I've heard a couple that were just awful: they didn't even get the melody right, and you can't get much simpler than that! But I think most of them have been great." Mark specifically approves of the DJ Dado remix, and the Flexifinger mix that appears on the CD *The Truth and the Light: Music from the X-Files.*

There are actually two CDs of *X-Files*-related material out now: *Songs in the Key of X* is a Don Was co-production [*SOS* talked to Don Was back in December 1994] featuring a variety of bands performing series-inspired music, while the recently released *The Truth and the Light: Music From the X-Files* contains Mark Snow's music from the series, plus some new material, pieced together with moody snippets of original dialogue. "I had nothing to with the *Songs* CD; Chris Carter is a real fan of modern music, and all the people on the album are fans of the show, and they all

wanted to be on it.

A lot of the hardcore fans thought it was really bogus because none of the songs (save for a track by Nick Cave and the Bad Seeds) had been on the show. But it still sold pretty well. Now the thing that's disappointing to me is the CD with the show's music, which a lot of the fans have been waiting for, isn't getting much publicity. I mean, it's selling better in France than it is in America. I questioned the label about it, and they said 'Madonna's not on it, REM's not on it, so we can't really put much into it.' I said, 'But this is *The X-Files*! This isn't just any ordinary TV show...' We're hoping that it'll just stay around long enough to still be selling in years to come. It feels like that'll be the case, as word of mouth spreads."

emotional, real way, and I strive for that in the samples and my electronic mock-up of the orchestra. It's a very important thing for me: the music has to have a sense of life to it."

When it comes to the final music mix, analogue two-track is definitely a thing of the past: "We use Tascam's DA88 8-track digital recorder — I actually have Sony's PCM800 version. The music editor stripes a tape with timecode, and we mix down to that. We just drop the cassette into another machine at the mixing stage."

Given the extensive use of QSound 3D processing and Dolby Surround on the final mix, I wondered if any allowances had to be made at the music stage: "At first, we maybe had a little bit of trouble getting that right, but I think the two most important points are that things shouldn't be split to extreme left and right and that the lows and the highs — especially the lows — shouldn't be overdone. You can't think of it as a movie score where the lows can be just really packed in: you have to be a little discreet, because the low end tends to spread and wash everything out on television."

TOWARDS THE MILLENNIUM

Of course, Chris Carter has recently launched another series, *Millennium* [mentioned in Paul White's interview with David West, starting on page 154], which debuted on satellite in the UK at the end of December last year, and for which Mark is also providing the soundtrack: "It's doing pretty well. At the moment, I think it doesn't quite have the broadness of *The X-Files*. It's sort of a one-note piece, more of a murder mystery, but it's really expanding now, and the lead character, played by Lance Henrikson [of *Aliens* and *Terminator* fame] is really great: he plays it so quietly and deliberately, it's very, very cool... Musically, it's a whole different approach. The theme features a live violin playing a sort of sad Irish melody, with big Japanese drums in the background, a nice change of pace from *The X-Files*. The music started off fairly one-dimensional, just low, sustained synth with drum accents, but the show is broadening out, which is good."

Snow's obviously in demand: how does he handle the workload of two network TV shows? "It's not like two shows happening every week at the same time — there are breaks. Over Thanksgiving, it got pretty bad because there was a two-part *X-Files* that had a very short turn-around and needed a lot of music, but that's as bad as it's been. The workload seems to be manageable!"

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Allows you to accurately monitor the levels of your recordings on each track.

3. Left and Right Locator Displays

Shows you the currently selected in/out points. Values can be easily edited with click/drag mouse operation. Values can be readily selected and dragged into any of the 20 positions in the Position Memory bank and vice versa.

4. Cycle Button

Enables cycle playback and record between the left and right locators.

5. Track Switches

Besides the standard switches such as Solo, Mute and Record, the Stereo Buttons allow you to group 2 tracks together as one track for easy handling.

6. Wave Display

The positive waveforms of your recorded tracks smoothly scroll from right to left during playback and recording. This allows you to easily navigate through your recordings.

7. Set-Locator Button

These buttons allow you to write the current "tape" position into the locator displays on the fly. Controllable either with the mouse or with a keystroke, these buttons allow you to quickly generate new left and right locators.

8. Tape Indicator

Indicates how much space or "tape length" you have left for your recording. It also indicates your current position within the recording.

9. Auto Drop Button

With this button enabled, the VMR will automatically switch to record mode using the currently displayed left locator as record IN and the right locator as record OUT.

10. Tape Button

You'll need to choose a "tape" before making your first recording. Pressing this button gives you a variety of tape length options depending on the size of your hard drive.

11. Copy Button

This button allows you to copy, move or mixdown the data of "soloed" source tracks between the left and right locators to the tape position on a record enabled destination track.

12. Input Selector

Press "A" to choose the Analog Input, or "D" to choose the Digital Input on your Audiowerk8 Card.

13. Pitch Variation

Clicking on this button opens a display where you can simply enter the desired pitch value by clicking and dragging. Pitch ranges are from -9.99% to +9.99% in steps of 0.01%.

14. Position Memory Bank

Up to 24 positions can be stored with each tape. 6 can be displayed simultaneously. To view others, simply click on the slider and drag left or right. Clicking on the "pair" button between 2 position memories gives you a "pair selection" which can be dragged into both locator displays simultaneously. Position memories can also be set and recalled via keystrokes.

15. Position Display

Optimized for legibility, the position display shows you the current tape position. A maximum tape length of 1 hour, 59 minutes and 59.59 seconds can be displayed.

16. Tape Controls

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Digital Discovery

SONY MDMX4 DIGITAL MULTITRACKER

Sony's new MDMX4 digital multitracker heralds the beginning of a conscious attempt to woo the home recording market, and uses the company's own MiniDisc technology to put a digital spin on that home studio mainstay, the compact 4-track recorder. **HUGH ROBJOHNS** *checks it out.*

The Sony MDMX4 is the third in the current trio of MiniDisc-based 4-track recorders to come on to the market in the past year, the others being the Yamaha MD4 (reviewed *SOS* September '96) and the Tascam 564 (previewed *SOS* August '96 and reviewed December '96). Although all the machines are clearly designed to do the same job and to appeal to the same sector of the home recording market, each has slightly different facilities.

Sony are relatively new to the world of home-recording systems, and are therefore some way behind both Yamaha and Tascam in terms of practical experience of this market place. However, MiniDisc technology is Sony's own invention, after all, and they have manufactured audio mixers for a great many years, so there is nothing in the MDMX4 which is entirely new to the company.

FIRST IMPRESSIONS

The MDMX4 is split roughly down the middle, with the 10-input mixer section on the left, monitoring in the centre and the MiniDisc transport on the right hand side. It is intended to work with MD-Data discs, although straight audio MiniDiscs may be used if some of the functionality of the system is sacrificed (ie. you only get two tracks instead of four, and limited MIDI facilities). Audio is recorded onto the disk as a series of Songs, and you can have up to 255 of these per disc — provided the total record time doesn't exceed 37 minutes for 4-track audio, or 74 minutes for stereo-only material, as this is all that will fit on a MD-Data disc.

The mixer has two balanced inputs with XLR/jack sockets, capable of handling microphone levels (there's no phantom power), two more unbalanced channels and one stereo channel. Fixed three-band equalisation is available on the first four strips, and there are two auxiliary outputs available from the single send level control on each channel. There are also two pairs of effects returns, which may be used in mono or stereo. All input connections are accessible from the top of the mixer panel.

The rear panel of the MDMX4 carries two pairs of phono sockets, providing a stereo monitor output (from the monitoring buss) and the main stereo output from the mixer. There are no digital audio inputs or outputs at all, but the horizontal section at the back of the mixer carries direct outputs from the four MiniDisc replay tracks, allowing mixdowns through an external desk, or direct transfers to another multitrack medium. Needless to say, there is a full set of MIDI connectors on the rear of the machine, together with a couple of footswitch sockets on the front which may be programmed to perform a variety of functions. Also on the front is a single, standard-sized headphone connector. The mains lead is captive on the rear panel, and the power switch is next to it on the back of the machine.

THE MIXER

The mixer section of the MDMX4 is pretty much par for the course in terms of its facilities. It has four full-facility inputs, two more arranged as a stereo pair, and two stereo effects returns, making a total of 10 individual input channels. The first two inputs have combi-XLR sockets which accept quarter-inch jacks in their centres. These are wired to accept balanced sources, whereas the other inputs (all on quarter-inch jacks) are for unbalanced signals.

There is no mic/line switching, merely a level knob which provides a 40dB gain range. Immediately below the gain trims on the first four channels, push buttons select the mixer input from either the mic/line input socket or the appropriate MiniDisc channel for mixdowns and track bounces.

Next down
on the channel strips are
the three knobs for the equaliser
section. The mixer has fixed EQ curves, with only
the Boost or Cut control available to the user. The
top and bottom section are standard shelving filters
(with turnovers at 50Hz and 15kHz) with up to
15dB of boost or cut, while the mid section is a bell
(peaking) response with a 12dB range, and is
centred at 2.5kHz. Unfortunately, EQ bypass
buttons have not been provided, so you can't easily
compare the EQ'd signal with the original.

Below the equaliser section, a single knob is
provided for the two auxiliary sends, with a centre
detent which represents the zero point for the
auxiliary controls. Turning the knob to the left
increases the send level to Aux 1, whilst turning it
to the right sends to Aux 2. Although I have not
seen this kind of economy on send controls before,
it actually works very well, and the compromise of
only being able to send to one auxiliary buss at a

time is reasonable in the
circumstances.

Finally, at the bottom of the channel strip, two
push buttons and a pan pot provide the means of
routing channels to recorder tracks. It is not clear
from the panel markings, but the post-fader
channel output is always available to the main
stereo mix buss, the push buttons merely adding
the track destinations to the channel output. The
faders have an optimum working position indicated,
and the handbook stresses the importance of
setting the input gain correctly to optimise this
position (minimising noise and avoiding distortion).

Channels 5/6 are configured for a stereo source,
and so have only one set of shared controls. There
is no input gain control on these channels, only the ▶

Sony MDMX4

The MDMX4's 10-input mixer.

"...the MD format is so far ahead of any cassette multitracker in terms of original recording quality, low noise, and dynamic range... that it is not really fair to make a comparison."

top and bottom shelving sections are available in the EQ section, and there are no auxiliary sends. The pan pot becomes a balance control, there is a single stereo fader, and the normal track routing is retained. This section is clearly intended for stereo keyboards or maybe pre-recorded backing tracks, although I found the absence of an auxiliary send facility rather frustrating.

Adjacent to channels 5/6 is the master stereo output, with main fader at the bottom. Above the main fader is the monitoring section, which has four sources, selected on push buttons labelled Cue, Stereo, Buss 1-2 and Buss 3-4. The Stereo button picks up the main stereo mix buss, whilst the Buss 1-2 and 3-4 buttons monitor the track routing busses. The Cue button selects the output of a separate monitoring mixer, where independent level control knobs are provided for each of the four MiniDisc replay tracks. A Volume knob controls the monitoring level available to both the phono connectors on the rear panel and the headphone socket on the front edge.

At the top of both the master strip and that of channels 5/6 are facilities for the two stereo effects returns. Each section has a level knob and a set of buss assign buttons for recording the effects (both returns permanently feed into the main stereo mix buss). Immediately below the effects returns, on the 5/6 channel strip, two knobs control the master send level for the two auxiliary outputs. In general, the

mixer is well laid-out, the function of each control is clear, and everything performs as expected.

DISPLAY & TRANSPORT

The MiniDisc transport has a pleasantly neat and simple collection of controls, with an excellent display screen. The disc drive is mounted on the right-hand edge of the machine, with an Eject button above the display, so the MDMX4 will have to be installed in such a position that the right-hand side is not obstructed, to allow discs to be loaded and removed.

The transport section is dominated by the display panel, which incorporates six bargraph meters (for tracks 1-4 plus main stereo output), a large numeric time display (minutes, seconds, and frames, or bars, beats, and clocks), various function indicators, and an alphanumeric section for edit and system instructions.

The meters are a little small and of limited resolution, with 3dB steps around the peak levels (the top bar is red and indicates clipping), increasing to 6, 8 and finally 10dB steps down to -40dBFS. In practice, however, setting appropriate recording levels was not a problem, provided a generous headroom was allowed (I tended to peak things at around -10dBFS). Very brief transients illuminating the clip lights were often found to be OK on replay.

Immediately below the 4-track meters, bright red record tallies show which tracks are recording. These are activated by pressing one or more of the four arming keys just below the display. On the right-hand side of the transport section, a vertical block of eight buttons provides facilities to configure the machine, and performs various editing functions on the disc. These include facilities for selecting the time display modes, accessing the editing modes, adjusting the transport speed (pitch), setting the system parameters, rehearsing editing functions, and punching in and out; and there's also a very useful Undo button which erases a previous edit or punch-in action, retrieving the original data.

On the left-hand side of the transport section, eight more buttons in a horizontal block provide locator facilities and in/out marks for the editing functions, and below, six more buttons provide the transport control functions. The Play and Stop buttons are both large and easily identifiable (the Play button has a small green LED to indicate its status), and there are a couple of smaller buttons above the Stop button to locate to the start of each track on the disc. A useful control above the Play button, labelled Top, locates the start of the current track, or finds blank space to record a new track. The Record button is circular to separate it visually from the others, and it lights up red when active.

Finally, a large jog dial in the bottom right-hand corner completes the control facilities. This wheel is a dual-concentric affair, with a continuously-rotating inner section and a spring-returned outer section. It is used to search through songs (providing brief and infrequent snatches of audio rather than a continuous fast replay), select items and set parameters in the various control menus, and adjust the locators and marker time values. ▶

Sony MDMX4

IN USE

The MDMX4 is initially very intuitive to use. The mixer is easy to set up, and even a complete novice can be recording tracks within a couple of minutes, without having to read through the manual first. Having said that, the 65-page manual is crucial to

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MIDI

The MDMX4 has all the usual MIDI facilities you might expect. Synchronising a sequencer to the MDMX4 may be performed using either MIDI Time Code (MTC) or MIDI Clock. In the case of the latter, a tempo map must be created first (specifying the bar, time signature, and tempo), but this is very easy to do. Up to 50 tempo maps may be stored with each song, but not on standard audio MiniDiscs — MD-Data discs must be used for this. The MDMX4 also accepts MIDI Machine Control instructions (MMC), so that the whole transport may be controlled from a suitably-equipped sequencer system. All the MIDI aspects of the MDMX4 worked flawlessly in the time I had to play with it.

The back panel connections, including MIDI sockets for MIDI sync and MMC control.

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mastering the many complex track editing functions and the MIDI facilities, and has some very important information hidden away amongst its pages.

Recording individual tracks is a trivial process — you simply route the appropriate signals to the right tracks at sensible levels, arm the desired recording tracks, and press the play and record buttons. Lamps flash until the system is up to speed, and then away you go.

During testing, I found the mic inputs on the review sample to be disappointingly noisy at the higher gain settings necessary for typical moving-coil microphones (I was using mainly Beyer M201s and AKG D202s, not untypical of the sort of mic this unit is likely to be partnered with). This problem was brought to the attention of UK distributors HHB, and after investigation, they were able to report the discovery of a faulty batch of units. We were assured that this fault would not be present on any unit sold in the UK. At lower gain settings for line-level inputs, the first-stage noise on my unit was more acceptable, so I used an outboard microphone preamp for all of my demo and test recordings, thereby feeding only line-level signals into the MDMX4.

Overdubbing additional tracks is no harder than initial recording, once you have mastered the monitoring system to hear the replay of earlier recordings. The interesting bit comes when

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ver ponder, as you gaze wistfully into your screen, editing and re-editing for the umpteenth time, just why you started playing music in the first place? Get your hands on an SU10 and it will all come flooding back — the SU10's intuitive ribbon controller with filtering, cross-fade and real-time scratching positively begs you to experiment with sound. Though with 16-bit sampling at up to

you have recorded four tracks, but wish to add more. Imagine a situation where you have stereo drums on tracks 1 & 2, bass on 3 and a guitar part on 4. You now need to reduce these to a stereo mix, leaving space to record new keyboard parts or vocals. The process on the MDMX4 is to perform the mixdown of all four tracks (using the input selector switches to route the replayed tracks through the mixer), replacing say, tracks 1 and 2 in the process. Tracks 3 and 4 can then be over-recorded with the new additional tracks.

This works well enough, but as you erase the original recordings in this process, what do you do if the mix goes wrong halfway through, or you later decide to change something? Well, Sony recommend that before each stereo mixdown, you copy the entire song to a new part of the disc. This copy is an internal digital clone of course, so there is no degradation of the original tracks, although it takes time, and seems to be a real-time copy (I was expecting it to make a copy faster than in real time). The copy acts as a backup if you have problems later, in which case you can try your stereo mixdown again, working from the copy (though you ought to make another backup before you do this, of course).

To put this whole process into context, I set out to record a demo as Song 1. Having recorded the stereo percussion and an organ part across four tracks, I copied the whole thing to Song 2. I then bounced the four parts of Song 1 down to a stereo track (1/2) and overdubbed tracks 3 and 4 with a couple of acoustic guitar parts. All four tracks were then copied to Song 3 and I bounced down the guitars and backing tracks in Song 1, leaving tracks 3 and 4 free once again — this time for the vocals.

This recording, copying, and bouncing procedure allows you to go back and change things at a later time should you need to, and provides the very important protection against losing your original material if a bounce goes wrong, but it is time-consuming. In comparison, Tascam's 564 MiniDisc Portastudio (reviewed *SOS* December '96) offers a 'bounce-forward' facility, where a combination of four tracks in the original song can be mixed down and re-recorded as a new song on a different part of the disc. This allows you to retain the original tracks without having to make copies first. I know the bounce-forward technique used in the Tascam requires a very fast disc drive, so that the system can replay from one part of the disc and record on another, but if Tascam can do it, why not the original inventor of the MD format?

Punching in and out of recording is easy to do, either manually or by preset editing points. The machine can be set up to rehearse an automated drop-in, and the pre- and post-roll times can be set to suit the material or performer in one-second increments up to 10 seconds. At the front of the ▶

"...certainly represents a major step up for any cassette-based home recording setup."

Sony MDMX4

▶ machine, two footswitch jacks allow a variety of user-configurable start/stop and punch in/out operations to be remotely activated.

As with other MiniDisc formats, obvious audible degradation occurs after about five generations of bouncing down, and in my case, the rather dynamic percussion track I used in the demo was not quite what it had been after only one generation (a slightly dull, 'fizzy' effect appeared on the hi-hats). However, to put the quality in context, the MD format is so far ahead of any cassette multitracker in terms of original recording quality, low noise, dynamic range and the number of generations which can be tolerated, that it is not really fair to make a comparison. It may not be up to the best of professional digital multitrack formats, but it is not that far off, and certainly represents a major step up for any cassette-based home recording setup.

EDITING

One of the real strengths of the MiniDisc format is its ability to act as a kind of musical scratchpad. You can very quickly bash out a few ideas and knock up a demo, just like an analogue cassette system. But the real advantage is that you can then try altering the complete structure of a song. Move the second verse, insert another chorus, try different endings — it is all possible by simply editing the table of contents (TOC) on the disc, so that the machine plays the recorded data in different orders. This is an excellent facility, and one which, to many people, would make a MiniDisc multitracker worth its purchase price alone! The operational aspects of this kind of editing on the MDMX4 are quite good, although I found trimming an edit point with the jog wheel to be a little fiddly, and achieved the best results by marking edit points 'on the fly' (ie. hitting the marker buttons as I heard the appropriate beat).

Editing is not limited to complete sections of songs across all tracks, either. You can very easily remove a section from a single track, or copy a section to another track. Parts can be also exchanged (so the bassline in one verse could be exchanged with that in another, for example). All you have to do is to mark the beginning and end of the source material (using dedicated In and Out Mark Locator buttons), and then store the new position with another dedicated button. A Rehearsal function allows the exact in and out points to be checked before performing the edit, and an Undo button cancels the edit, restoring the previous audio sequences. Once all the timing information is logged, the Edit mode is entered, and the appropriate source and destination tracks are identified. After dealing with the usual 'do you really want to do this?' type of message, the job is done!

The editing functions are extremely useful, but

there are a couple of points which are not entirely obvious, and serve to catch out the unwary. Firstly, the nature of data storage on a MiniDisc is such that it is not possible to move or copy sections shorter than about four seconds (eight seconds in 2-track mode). Secondly, it is not possible to punch in after a song has been edited (the whole edited song has to be copied to a new location first). Lastly, the point which caught me out on several occasions was that song editing can only be performed once the disc's table of contents (TOC) has been updated after any recording. This is easily done by simply pressing the Stop button twice, but it had me scratching my head for a while! The TOC is not automatically updated on the disc — the machine keeps its own version in memory, only updating the disc's TOC when the Stop button is pressed or the disc is ejected. Another consequence of this is that if the power is turned off while the disc is still in the machine, the TOC may not be fully up to date, and there is a very real risk that the entire disc's contents may be unretrievable! Although this problem affects all MiniDisc recorders, it's pretty scary...

Other TOC editing functions allow a song to be divided into two (or more) separate songs, or a number of songs to be combined into a single one. Songs can also be deleted, of course, or named to make finding them easier (which is especially useful with all that song copy business before overdubbing, as it helps you to keep track of what instruments are on each song). The entire disc can also be named, which could be useful if you want to identify particular selections of material.

CONCLUSIONS

As Sony's first foray into the world of home recording, this is a very creditable effort. Overall, the mixer section is a good compromise between functionality and cost, the transport section is easy to use, and the display is very clear. Finding your way around the machine is really pretty intuitive for the most part, although I had to resort to the handbook once or twice to fathom out the detail of some of the editing functions.

The need to copy songs to preserve tracks prior to bouncing down is tedious, and interrupts the flow of work — especially since the Song copying process seems to be a real-time operation. Tascam's bounce-forward system puts it ahead of both the Sony and the Yamaha MD4 in this respect. On the whole, however, I think the Sony MDMX4's good solid construction, attractive looks and comfortable user interface will make it a worthy contender in the MiniDisc multitracker race. **SOS**

Transport controls and display.

Hugh Robjohns is a lecturer at the Centre for Broadcast Skills Training at BBC Wood Norton. The views expressed in this article are the author's own and are not necessarily those of his employer.

Equal Opportunities

ALL ABOUT EQUALISATION

PAUL WHITE looks at the various types of equaliser, and gives some tips on how EQ should best be applied to your recordings.
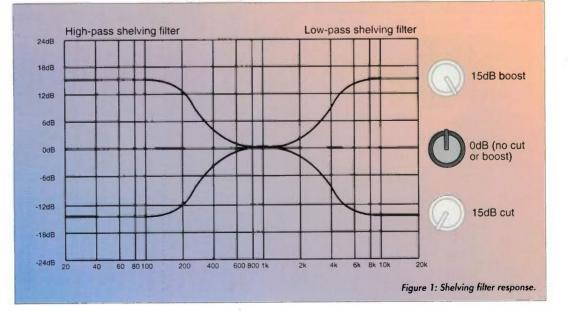
Equalisation can come in many forms, from simple treble/bass controls to multi-band, parametric equalisers, but, however you dress it up, equaliser is just another word for 'tone control'. The term equalisation came about because the very first equalisers were developed to help counteract or 'equalise' shortcomings in telephone systems, but today equalisation is used creatively as well as to fix problems. Equaliser circuits are based on electronic filters — hence the term filter, which crops up quite a lot when you're talking about EQ. Strictly speaking, a filter is a device that removes something, but in the context of active equaliser circuits filters can boost frequencies as well as cut them.

Check any textbook on audio, and you'll see the limits of human hearing quoted as around 50Hz to 20kHz, though those very same books will also point out that very few individuals, other than young children, can hear pitches anything like as high as 20kHz. A more realistic figure might be around 15kHz for an adult, decreasing further as the years pass. What is really puzzling, though — and this is scope enough for a feature in its own right — is that even if your measured hearing response starts to fall off well below 20kHz, it is still possible to hear the effect of equalisation applied at the top end of the spectrum, where you wouldn't expect to be capable of hearing any change. The plot thickens further when reputable studio engineers claim to be able to differentiate between two otherwise identical circuits if one has been modified to handle frequencies up to 50kHz and one handles frequencies only up to 30kHz. In theory, both limits are well above the limit of human perception, so it seems that what goes on outside the audible spectrum has a way of influencing what we perceive within the range of our own hearing systems.

SHELVING EQUALISERS

Though equalisers all do essentially the same job, there's a great deal of difference between a simple 2-band treble/bass tone control and a multi-band studio equaliser. The simplest is the shelving equaliser — a device that applies cut or boost, rather like a volume control, but only to the frequencies above or below the cutoff point of the equaliser, depending on whether the equaliser is based around a high-pass or a low-pass filter.

A low-pass shelving filter, as its name suggests, passes all frequencies below its cutoff frequency, but attenuates all frequencies above its cutoff frequency. Similarly, a high-pass filter passes all frequencies ▶

Figure 1: Shelving filter response.

ALL ABOUT EQUALISATION

▶ above its cutoff frequency, but affects all frequencies below its cutoff frequency. Figure 1 shows the frequency response graphs of a typical treble/bass EQ using high- and low-pass filters. Note that the filter graph shows a slope at the cutoff point — it isn't possible, or desirable, to have a filter that does nothing at one frequency, and then comes in with full effect when you move up by just 1Hz. Simple shelving filters typically have a 6dB-per-octave slope, so that their influence is felt more progressively — though it is possible to make much steeper slopes if required. The gentler the slope of the filter, the more frequencies outside the range of the filter will be affected.

BAND-PASS FILTERS

A filter that passes frequencies between two limits is known as a band-pass filter, and on a mixer with a mid-range control, the Mid knob controls a band-pass filter. On a typical mixer, the band-pass filter will have variable cut and boost, and on more flexible mixers it will also be tuneable, so that its centre frequency can be varied. This is known as a sweep equaliser, because although the filter

adjusted. The width of a filter response is sometimes described as its 'Q' value, where Q is the filter frequency divided by the number of Hz the filter affects — in other words, its bandwidth. The formula for Q is:

$$Q = \frac{\text{centre frequency in Hz}}{\text{bandwidth}}$$

Because the filter response is curved, the actual frequency width is measured between the points on the graph where the signal level has fallen by 3dB. A high value for Q corresponds to a very narrow filter; a low value of Q corresponds to a wide filter. High Q values are useful for picking out sounds that occupy a very narrow part of the audio spectrum, whereas lower Qs produce a smoother, more musical sound.
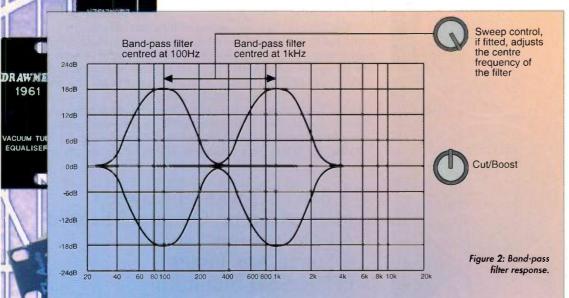
A studio parametric EQ may have several filter sections, enabling three or four parts of the frequency spectrum to be treated simultaneously. Parametric EQs can be time-consuming to set up properly, but they are the most powerful and flexible of the conventional EQ types. Figure 3 (overleaf) shows a typical parametric equaliser response.

Figure 2: Band-pass filter response.

frequency can be changed, the width of the filter cannot be adjusted. Figure 2 shows a typical band-pass filter response, including a sweep control function. In a typical mixer, the high and low shelving equalisers are used to control the high and low end, while a band-pass filter controls the mid range. However, some mixers use band-pass filters for EQing the low end too. The argument for doing this is that a shelving high-pass filter will also boost all frequencies below the ones you want to work on, whereas a band-pass filter attenuates frequencies both below and above its cutoff points.

PARAMETRIC EQ

A parametric EQ is very similar to a sweep band-pass EQ, except that a third control is added to allow the width of the filter response to be

GRAPHIC EQUALISER

A graphic equaliser can be recognised by the row of faders across the front panel, each fader controlling its own narrow section of the audio spectrum. For example, a 30-band graphic equaliser provides independent control over 30 different bands spaced one third of an octave apart.

Other than the highest and lowest faders, which control shelving filters, each of the filters in a graphic equaliser is a fixed-frequency band-pass filter, where ▶

ALL ABOUT EQUALISATION

Figure 3: Parametric EQ response.

Sweep or Frequency control adjusts the centre frequency of the filter

Cut/Boost

Q control varies bandwidth of filter. The narrower the bandwidth, the higher the Q value.

▶ boost is applied by moving the fader up from its centre position, and where cut is achieved by moving the fader down. Graphic equalisers have the advantage of being very easy to set up, but must be used sparingly, as if they're not very well designed, they can have an adverse effect on the sound. They are also less flexible than the parametric EQ, which can be tuned exactly to specific frequencies. On the graphic equaliser, the range covered by each fader is fixed, and the width of each individual band of a third-octave equaliser is actually rather wider than a third of an octave, to allow a smooth overlap between bands. Figure 4 shows the response of a typical graphic equaliser.

EQ TODAY
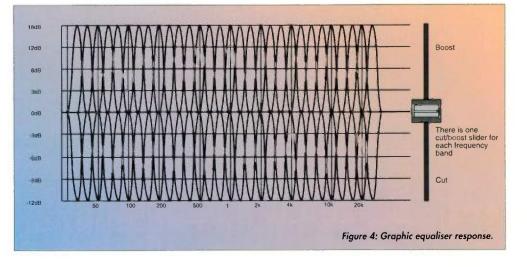
While early studio equipment and, to a greater extent, telephone lines needed a lot of corrective EQ to make them sound natural, modern recording equipment is capable of storing and reproducing sound that is virtually identical to the original. Nevertheless, the original sound isn't always what we want to hear, so EQ has evolved to take more of a creative role. What's more, the

simple bass and treble 'tone' controls found on early equipment have been replaced by sophisticated multi-band equalisers which place far more precise control in the hands of the user.

At heart, no matter how complicated an EQ is, it is still really just a frequency-selective volume control, but its subjective effect on the sound is often more profound than this description might leave you to expect. In February 1996's SOS I wrote an article that explored some of the psycho-acoustic aspects of EQ, and concluded that the most logical place to start searching for reasons why EQ has the effect on us that it does was in nature — specifically, the Earth's atmosphere. Here, low frequencies travel slightly faster than high frequencies, so the further away a sound source is from the listener, the more delay there'll be between the low frequencies (which arrive first), and the upper harmonics, which follow. The higher up the spectrum you move, the greater the delay will be, and, in effect, the harmonic structure of the sound becomes progressively more smeared as it travels. Air also absorbs high frequencies more readily than low ones, so the further away a sound is, the less bright it is.

Obviously, we're used to sound behaving like this, so we accept the effect as natural, which, by definition, it is. Our brains don't recognise the phase distortion — they just recognise the sound as being distant. By the same token, louder sounds that have little or no phase smearing are perceived as less distant. In recorded sound, however, we're faced with the problem that all recorded sounds travel the same distance — if the hi-fi speakers are three metres from the listener, then that's how far the sound travels. To fake the illusion of distance in a mix, therefore, we need to recreate the smeared harmonics and reduced HF response that occur in nature.

LEVEL & PHASE

We've already established that EQ acts as a selective volume control, targeting only certain parts of the audio spectrum. However, it's well known to circuit designers that EQ doesn't only change the level of specific parts of the spectrum, it also changes the phase of the affected frequencies relative to those

Boost

There is one cut/boost slider for each frequency band

Cut

Figure 4: Graphic equaliser response.

that aren't being cut or boosted. In other words, the equaliser is creating a similar effect to changing the distance of the sound source — it's affecting both the frequency response and the phase relationships of the signal. As I suggested in last year's psycho-acoustics article, this may be one of the reasons why brightening up a sound makes it seem closer, and why taking off some high end might make a sound seem more distant. Of course, every design of EQ affects the audio spectrum and phase response in a different way, and — leaving aside technical criteria such as noise and distortion — this might explain why some EQs have a more natural, musical sound than others.

IMPRESSIONS OF LOUDNESS

It's a well-documented fact that the human hearing curve isn't flat, but instead is more sensitive to mid-range sounds than to frequencies at the extreme high and low ends of the spectrum. Of course, we don't notice this, because we've heard sounds this way all our lives. However, as the level of sound we're listening to increases, the mid boost of the hearing system becomes less, and the result is that high- and low-frequency sounds seem proportionally louder. This is yet another of those interesting physiological facts that can be exploited to fool the ear into believing something that isn't entirely true. For example, if we know that extreme high and low frequencies stand out more when we listen to loud music, we can create the impression of loudness at lower listening levels by attenuating the mid range and boosting the HF and LF ends of the spectrum. The loudness button on a stereo system does exactly this, and if you look at the graphic EQs used in a club or PA system, you'll often see them set up with a smile-shaped EQ curve to promote the illusion of loudness and power. Of course, this works just as well in the studio, although it's always most effective just to treat some of the sounds in a mix, to maintain a contrast between the different sounds.

OVERVIEW

As a general rule, equalisation should be employed only after all efforts have been made to obtain the best sound at source. What's more, there's a huge subjective difference in sound between a budget equaliser and a top-quality studio equaliser: if you have to work with a budget EQ, or the EQ section built into your desk, you'll probably have to use it sparingly if the overall sound isn't to suffer. Though the character of a really nice equaliser is difficult to quantify, the best equalisers allow you to make more drastic changes without the sound appearing unnatural.

Most often, a combination of cut and boost is required — but always use the Bypass switch to flip back and forth between the equalised and unequalised sounds, to make sure you really have improved matters. Equally, if you feel the need to EQ an instrument in isolation, check again with all tracks playing to make sure that the settings you're using work in context with the rest of the mix. More often than not, you'll have to make further adjustments, but it really is worth striving to get your sounds right at the outset. EQ is an invaluable ally in shaping well-recorded sounds, but even the best equalisers have their limits when faced with difficult material. **SOS**

Saber Dance

DANNY SABER

"At the end of the day it's all about making the vocal that you're working with shine, and building things around that vocal. That's what the average guy listens to. He doesn't give a damn about what I'm playing on the guitar. Maybe some other guitar player does, but nobody really cares how interesting the noise is. People either like it or they don't, and they get a
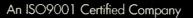
He's an accomplished producer, remixer, writer, programmer, guitarist, bassist and keyboard player — is there no end to Danny Saber's abilities? RICHARD BUSKIN talks to him about his blending of pop, rock, hip-hop and trip-hop sensibilities, as well as the commercially unorthodox results that have placed him in international demand.

vibe. The music sucks them in but the vocal is what keeps them there, so everything has got to be supporting that vocal."

In a nutshell, that's the musical philosophy of Danny Saber, and one that, in light of his success working on records by Black Grape, Garbage, Madonna, Terrorvision and Chuck D, not to mention upcoming releases by David Bowie, Michael Hutchence and Agent Provocateur, is well worth noting.

Saber began his musical career playing in a band with vocalist Issa Joone. "I started out as a guitarist," he says, "but from the first time I ever went into a studio I was always telling everyone else what to do." During the early '90s, the garage at the Joone family home housed a studio setup that Saber subsequently used when recording demos for a wide variety of LA acts, including names such as Bronx Style Bob and House Of Pain. "I was supplying the music, collaborating with the musicians and playing a lot of the instruments," he now recalls, "and that's pretty much what I do now with Black Grape. When required I'll pick up the bass or play some guitar, and I also do all of the programming myself. I don't have any programmers who work for me. I learned that stuff by just having a room, buying the equipment and finding out how to work it. Totally trial and error, and that was really cool. I was real lucky in my timing because as things were coming ▶

Danny Saber in the Neve Room at his preferred studio, Westlake, in Hollywood.

DANNY SABER

Terrorvision — just one band to have received the Saber touch.

▶ out I was right there learning how to use them. The gear was advanced, yet affordable — not like the early '80s, when you needed $200,000 to buy a Fairlight.

"I was forced to learn. If some singer or rapper came in and said, 'I want to sample this, this and this, and I want to hear it all in the same song', as a musician I'd be thinking, 'That doesn't work!' But they weren't interested in hearing that, so I'd just have to take their ideas and somehow *make* them work. By doing that, and by having all of these different kinds of people with these different ideas coming to me, I really began to bring things to fruition. At first I was this real technical kind of person, but then I grew into a writer and that's where I'm at now."

TRIP-HOPPING ALONG

Take David Bowie, for instance. The original mix of his single, 'Little Wonder', was sent on 2-inch tape to Danny Saber with the request for him to produce full-on remixes. The result: the main vocal remained and everything else was rewritten.

"I kept the basic melody of what he sang and then I totally restructured the song around what I wanted to hear, what I liked and where I saw it going," says Saber. "I took the vocal, put it in *Logic* — I'm using Pro Tools interfaced with *Logic* for my programming — and then I built a whole new track around it. In fact I did a couple of mixes, and the first one consisted of what I wanted to hear from Bowie. He had done this real cool track with jungle rhythms and heavy guitar and I liked it, but I wanted to hear something different: some weird 1996 'Space Oddity' trip-hop kind of thing... It was all in my head before I did it, and for once it actually came out the way I imagined it.

"I started with an acoustic guitar, referencing many of the people who grew out of him, such as Love & Rockets, and then I came up with some changes and ended up writing a new acoustic song around his vocal. Once I get that initial spark I can go anywhere I want to. From there I put a loop on it and then I just built it up with *Logic*, the Akai S3000XL sampler and the Emu Orbit. Towards the end of the original mix it went into

David Bowie, whose 'Little Wonder'
has been remixed by Saber.

half-time, so I decided to put the whole song in half-time and make it into a ballad."
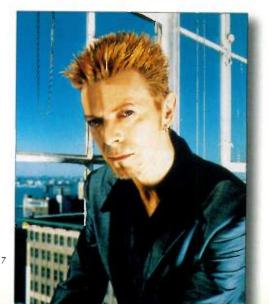
For the first half of the record Saber also opted to have Bowie's voice sounding as if it's coming out of a small transistor radio. "We just filtered the vocal. On the Drawmer gates you can input the audio into the key — there's actually a high-pass and a low-pass filter — and there's also a program on the Eventide called 'Critical band' which I like to use a lot. You dial in the frequency and it will thin it out and make it real small. It's just an EQ thing, and there are lots of different ways to get that. You can do it on the console, rolling off all of the low end and cranking up the high mids."

Among the sounds on the slow mix are a distorted drum loop, tuned down an octave and going through a dbx 363X gate, triggered on sixteenth notes via the Nord Lead keyboard; and strings courtesy of David Coleman and his electric cello, as well as half a dozen sounds — including that of a Mellotron — that were MIDI'd up with a Roland JD990. "I just thought 'Ah, Bowie — Tony Visconti!'" Saber recalls. "I grew up on that stuff — I learned how to play listening to it, so it was great to be able to do that and, in a way, it was also really easy for me."

WORKING WITH MR X

In the work sense, home to Danny Saber is Westlake Audio in Hollywood, where he and engineer John X usually employ an SSL console (or a Sony in the tiny production room that they use there) together with Saber's own assortment of toys.

"John is like the creative mind on the console," says Saber. "He comes up with a lot of the vocal noise sounds, recording things backwards and so on. In fact, on the slow Bowie remix he ran the vocal through a really long reverb and then recorded it backwards, producing this sort of 'whoosh' sucking sound. He's a real integral part of what I do because, once I've done all of the music and come up with the sound, John then mixes it, and in that sense he's not a standard engineer. He's very creative and very musical — a lot of times he'll put live drums through an Eventide H4000 and tune the kit to the song. He comes up with all of these mad ideas, and the beauty is that he's just as creative in what he does ▶

THE WISH-LIST

"Some computer program that will interface with your brain, so that if you heard a song in your head it would be converted to MIDI. That would be cool!"

DANNY SABER

▶ as I am in what I do, or as any of the artists are in what they do. He's not like an engineer in a white lab coat. He's real technical but he's real musical too, and at the end of the day we'll leave things in if they've got a vibe... Plus he's a nutcase, so we have a lot of fun!"

On the Bowie record, as on many others, Saber himself played a DI'd 5-string Modulus bass and a BC Rich Bitch guitar. "What I was going for was Tony Visconti, Mick Ronson and my own style mixed in," he says. "The guitar was going straight through a Marshall amp which I've had since I was 15. I've also got a Mesa Boogie Dual Rectifier that is real versatile, but it's a little more modern sounding. I travel a lot, so I can't take seven guitars with me."

Still, for the 'Little Wonder' remix Saber did rent a Martin 12-string, and then, for the sound of an orchestral 'rush' similar to that which distinguishes The Beatles' 'A Day In The Life', he took a sample of an orchestra tuning up, created a sound, spread it over the Nord Lead keyboard, and then triggered it at different points within the eight bars of the rise. For the second remix of 'Little Wonder', Saber was looking more to the feel of Bowie past masters such as 'Fame' — combined with certain 'Saberish' elements, of course.

"I wanted to do something a little faster and kind of funky. However, the tempo was too slow for what I wanted and if I moved it up it was too fast, so I used *Logic* to time-compress the vocal to the new tempo — I took it from around 75 beats per minute to 116, which was quite drastic. I also changed the way he was singing, by taking the lines and moving them around into a rhythm that fit the music. Then John X ran the vocal through a Digitech Vocalist to build the weird Vocoder-type harmonies that blend around the melody.

"Meanwhile, for the sort of 'transforming' sound that you can hear, I ran a Juno 106 through a distortion pedal. Again, I was coming from the DJ point of view, triggering it with a gate and then playing it on the keyboard in any kind of rhythm I wanted, while pitch-bending at the same time."

A MUSICIAN FIRST

For this remix, Saber had friend and bassist Jeffrey Conner come in and play. But, while it is evident that Saber is deeply into the techno sound, what also distinguishes him from many of his peers is the fact that he is first and foremost a musician rather than a DJ. ▶

THE TRAVELLING STUDIO SETUP

"This system has been built up over the last five years. I move around and so I have to keep everything streamlined. There's not one thing that doesn't get used every day, and that's the way I like it."

- **APPLE MAC QUADRA 800**
- **PACIFIC COAST TECHNOLOGIES HARD DRIVE**
- **DIGIDESIGN PRO TOOLS**

"This setup has been invaluable. Being able to put a vocal in and just cut it up any way I want it makes everything easier. Before this, putting vocals in the sampler could be a nightmare."

- **EMAGIC *LOGIC AUDIO* SEQUENCING SOFTWARE**

"This is great, the best programming device that there is. It interfaces with Pro Tools really well and it's really reliable. The sequencing is fantastic, because you've got 16 tracks of audio and as many sequences as you want. And the audio page is really bitching!"

- **AKAI S3000XL**

"This is basically the staple of everything I do, because I use a lot of samples."

- **MARANTZ CD RECORDER**

"This is my newest piece of gear and I'm very proud of it. It is awesome. It enables me to back up all of my Pro Tools stuff. I've got a 4Gb drive, so when I run out of space, for $7 I can make a 650Mb CD-ROM. So, even though this thing cost me $3000, it's the cheapest storage device and the most reliable.

"I've got an external 540Mb hard drive which I've formatted to the Akai S3000; when it's full, I just run off a CD-ROM to my Marantz. So now all of my samples are on CD-ROM, everything is labelled and it's far better than having to shuttle through 80 disks. On top of that,

none of the artists have DAT machines, so now I can run off CDs of the rough mixes for them, and that's great."

- **EMU VINTAGE KEYS**

"I've had this for a few years now, and it's a good multitimbral device for writing. If I want an organ sound as an idea it's there, but if I want a B3 I'll go out and get a real B3."

- **EMU ORBIT 9090**

"Again, another thing for when you want a quick sound. It's got some really cool noises in it."

- **ROLAND JD990**

"This has got killer bass sounds, really good strings, nice piano... excellent."

- **ALESIS ADAT XT**

"A cool machine. One ADAT's fine; if you need more than that, get a 2-inch!"

- **dbx 363X**

"I use this for gating things and screwing them up."

- **EVENTIDE H3000 SE**

"A beautiful piece of gear. It gives me an opportunity, even when I'm doing demos, to still have at least one good sound."

- **dbx 160X**

"I use this for vocals."

- **ROLAND MC303 GROOVE BOX**

"I've just got this, and so I have nothing to say about it. It's kind of a toy, it's a stand-alone thing and it's got really cool sounds, but I don't know how well it's

going to interface with my gear yet. It's only got two outputs, which kind of sucks — I bought it sight unseen and I thought it would have eight — but if I don't want it, I can sell it."

CLAVIA NORD LEAD

"I'm using this as my controller keyboard and for sounds. I was going to buy a Minimoog, and then, just as I was about to get it 18 months ago, this thing came out and I went to see it. It's got four channels — you can upgrade it to 12 voices — and every knob on it is recordable to MIDI. It's got all of these real techno sounds and it looks cool. The Moog wouldn't have been as practical."

DANNY SABER

"I've been playing guitar since I was 13 years old," he says. "I wanted to be Jimi Hendrix when I grew up. That's where I was coming from. In fact, when I got my first guitar the great studio gear wasn't accessible to most people. At the same time I also knew that I wanted to be more than just a guitarist. For one thing, I realised fairly early on that I was never going to be the world's greatest guitarist, and I was also aware that I was good at putting things together."

> "It's twice as hard for you to remix your own stuff, because you've already made it the way you want it. Now you've got to come up with a twist."

So, live bass, live guitars. The same often goes for the drum sounds, as played by Black Grape's Jed Lynch on any one of his Gretsch, Yamaha or Premier kits, though Danny Saber again sets himself apart from the rest of the field by combining such percussive efforts with loops. "It's really just a matter of trial-and-error with me," says Saber. "I mean, I'll get a loop, I'll know where I want the groove to be and I'll build everything around that. I'll take maybe 10 loops individually and tune them to a click at the right tempo. Then I'll just start turning them on and off to see what works together, resulting in a groove and a beat that I've probably never heard before. After all, it's basically like having five different drummers playing different beats all at the same time — if, somehow, you get them tight and they feel right together, you end up with a rhythm that's more than just a kick and a snare. Then if you throw live drums on top of that you may take it a step further."

Loops, samples, layering... Yet, in the face of temptation, how does the objective producer decide when enough is enough? Or is that like asking the proverbial question, 'How long is a piece of string?'

"Well, we get an all-in budget to do a remix, and so everything has got to be planned out to a tight schedule." says Saber. "I usually only do one mix, because you can hear the quality of what I'm doing. I'm not taking a vocal and putting a drum loop under it. I'm giving them something that's on a par with what they've done and probably spent a lot more time doing. In fact, with Bowie I spent a lot more time than normal; four days of studio time to do all those mixes, with a day or two to just think up the ideas, whereas usually what we'll do is take the tapes, go straight in and spend two or three days in the studio.

"So there's always a plan. The first day I'm coming up with the idea, the second day we're mixing it. That's how we limit ourselves. We know that we can't just sit in there forever and do stuff. That's the problem that I think some people run into when they have their own studio. They end up spending months on things instead of just going with that initial vibe and running with it. Having the luxury to analyse everything can actually hurt you in the end."

STUPID GIRL

Nevertheless, one luxury that Danny Saber certainly does appreciate is that of being left totally to his own devices when producing a remix. His work has earned him enough respect for the artists to trust his judgement and leave well alone. Communication is sparse, save for situations such as that with Garbage: Saber, the band and their producer, Butch Vig, all share the same management (Los Angeles-based SOS Management Inc).

"When it came to the remix of 'Stupid Girl', the only thing they said to me was that they wanted something to put on K-ROCK [an LA 'new wave'-type radio station]. So I just went for a Soft Cell

garbage

Stupid Girl

vibe combined with a kind of house/dance vibe. I basically took the vocal, I knew what the tempo was, I knew the key, and I got some loops going and started jamming over the track. I came up with a bass line that had a slightly different feel and I just kept building around that. They already had some really good noises on there, like guitars ▶

▶ feeding back and some real weird shit, including a Clash loop that they probably paid a load of money for and which they wanted me to use. So I started with that Clash loop and I guess I just made the track a little more housey and clubby and dancey. I did that in a day, and then we took another day to mix it.

"It's all down to the vocal. When there's a good vocal to work with it's a piece of cake, at least for me. But when you've got something that needs to be forced into some weird genre that doesn't even come close to what the artist is about, that's when it becomes much harder." Such as? "Melissa Etheridge! It was a song called 'I Want To Come Over', and I was trying to take a country rock kind of vocal and turn it into a club record! First off, we put her vocal into *Logic*, played with the words that we liked, and in the process turned the chorus from 'I want to come over' into 'I want to come over and over...' Then we put the vocal through a vocoder and totally replayed the melody to this track that I'd come up with. The original melody she was singing was really strong and it was hard to get away from the chord structure, so we put it all in the vocoder and it came out sounding like some weird, Donna Summer, trip-hop gay hit!

"What I didn't want to do was the standard thing of putting a house beat under the song. I could have done that and got away with it, but I really wanted to do something that made a statement with her. I think it's cool. So is Melissa Etheridge, and in that sense I knew that she'd hopefully appreciate whatever I did. She's probably got a sense of humour and from what I understand it was totally her idea to do this, so that was OK, whereas if I was doing it for somebody who took themselves too seriously it would probably turn into a ball-busting nightmare."

MR DO-IT-ALL

Having co-written, produced, programmed and played on Black Grape's 1995 album *It's Great When You're Straight...Yeah*, which went platinum in the UK, Danny Saber is not averse to remixing his own efforts.

"I did the first two remixes for them," he says, "and although it was kind of hard in a way, it was cool in a way too. I mean, I don't think that's the easiest stuff to remix anyway, because it's already so 'out there' to begin with. In fact it already is a remix.

"However, after the second remix I decided to give the third one to somebody else. I couldn't do it any more. It's twice as hard for you to remix your own stuff, because you've already made it the way you want it. Now you've got to come up with a twist. So I think it's good and it's also fun to give stuff to other people for them to remix, as long as there is a point to it.

"When I do remixes, people definitely get their

"It's all down to the vocal. When there's a good vocal to work with it's a piece of cake, at least for me."

money's worth, and I also get sent songs to mix for radio, so it's definitely legitimate, but I also think it can be a big pile of crap too. It just depends on who's doing it and what the motive is. If you're putting out a single, you've got to give people stuff that's not on the album, otherwise why should they buy it?"

A new Black Grape album is currently in the works, as is an on-going solo project by Michael Hutchence of INXS, which Danny Saber is once again producing, in addition to serving as co-composer, programmer and instrumentalist. This kicked off in 1995 at Nomis Studios in London, after Saber had just finished working there on the forthcoming album by Agent Provocateur. Nine songs were recorded with Hutchence in the space of six weeks — which also took in a spell at Peter Gabriel's Real World studio complex — and although sessions with INXS then took precedence, a recent five-day spell at Westlake saw the demo'ing of three more solo numbers.

"Michael's album is nearing completion and it's a lot harder-edged than what he does with
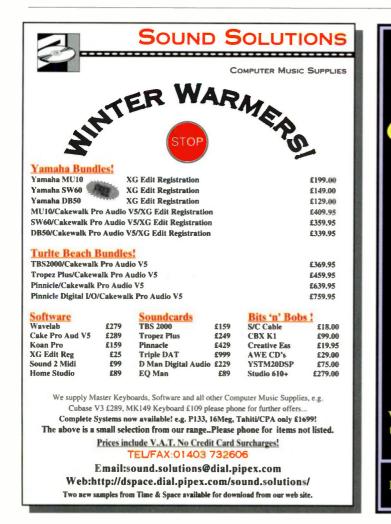
Danny Saber (right) with Black Grape's Shaun Ryder.

INXS," says Saber. "It's more like what I do, mixed with rock." That it certainly is — the demos I heard feature Hutchence's unmistakable vocal style, while all else around it is imbued with that Saber touch. "All of my equipment is used in writing to create a vibe, and what helps me is the fact that I've got all these different things to fall back on," he says. "For instance, on one of the songs I had a groove and a bass line, Michael came up with a vocal part and then I just jammed with the guitar for five minutes. After that I went through and edited myself, and I took one bar and kept looping it over and over again. It was really just a lick while I was doing a solo, but that turned into the chorus and so that basically came about completely by accident. With me it's always a case of trying things, playing and jamming, recognising something and then editing myself.

"Almost 90% of the time everything that I do on the demo ends up on the record. I might go back and re-record some of the guitars, the vocals will be improved upon, there will probably be some live drums on there, and I might pull a couple of the loops that I have just for the idea, but essentially it isn't going to be that much different. That's the way I write; the production and the song are one with me.

Photo: Richard Buslan

"OK, so who put the Smurfs on channel 12?"

"I've got something like five different angles from which to come up with a song: if I can't think of something on the bass, I can pick up the guitar; if I can't get the guitar right, I can play the keyboards; if I can't come up with anything on the bass, the guitar or the keyboards, I can sample something; and if I can't find anything cool to sample, I can take some loops and loop them, put a noise in and create something musical. So I've got all of these different places to pull from, and that is really what helps me." **SOS**

HIGHLY STRUNG

Roland continue to champion the cause of the guitar synthesist, refining their respected line of guitar synths in search of the best combination of facilities, usability and price. PAUL WHITE finds out whether they've got the balance right with the new GR30.

ROLAND GR30 GUITAR SYNTHESIZER

The MIDI guitar has always been fraught with technical difficulties. One of the main problems that beset the designers of MIDI guitar systems is the restricted speed of MIDI, which can affect how quickly sounds are triggered. Roland's recent instruments have largely side-stepped that issue, by providing good on-board sounds that are addressed directly from the processor, not via MIDI. Of course, a MIDI output is also available for driving external sound modules, but the reality of the situation is that tracking is both faster and more precise when you're using the internal sounds.

In the new GR30, Roland have built upon the ideas and techniques that made the low-cost GR09 possible, and the result is a slightly cheaper unit that incorporates new sounds, new performance controls, automatic harmony generation and a really neat arpeggiator. Requiring either an optional Roland GK2A (or GK2) split pickup, or one of the few 'Roland-ready' guitars on the market, the GR30 occupies the same shape of box as the older GR09 but is black rather than blue, and the control knobs

carry different functions. As you can see from the photograph, the control system is very simple, with just four pedals, four knobs and three buttons. As on the GR09, the GR30's display is simply a three-character alphanumeric window, so you can't name your patches. This also means that you need the manual when programming new sounds, in order to find out what sound corresponds to each of the 384 Tones in the list. If you can memorise all these, you could probably clean up on *The Generation Game*, but believe me, the GR30 is far more fun than a cuddly toy and a fondue set!

Designed along the lines of a 'stomp box'-type floor unit, the GR30 is powered by one of those irritating external PSUs and connects to the GK2A system via the provided multicore cable. The guitar's normal output may also be routed along the multicore to cut down on cabling, and this also provides the option of switching to guitar, synth, or both, from the GK2A. The guitar output may then be taken separately out of the GR30 or mixed with the synth sound. In the latter case, the guitar output may be used as an effects send to an external guitar processor; a separate jack is provided to return the effected signal to the GR30. MIDI In and Out connectors are present for sequencer users, and provision is made to connect an optional Bank Select switch and an expression pedal. The pedal may be programmed to do a number of different jobs, including changing the synth volume, tone, interval and arpeggio rate. The output from the GR30 is in stereo, though either jack can be used to plug in a pair of stereo headphones.

GETTING STARTED

The first step is to attach the GK2A divided pickup to the guitar, as close to the bridge as possible, and spaced from the strings in accordance with the instructions. Several shims, bits of sticky-backed plastic and Velcro are provided, though for a permanent job, screws and springs are also included. Once the pickup is in place, you need to go through a routine to set up the sensitivity of each string in turn, so that they are roughly equal in sensitivity. The row of LEDs to the left of the display serves multiple functions, including showing the string sensitivity, acting as a display for the on-board tuner, and indicating the status of the Local Off, Effect Bypass, Arp/Harm, Gtr Env Follow and Edit modes.

CONTROLS

The volume control regulates only the main mix output — the guitar output jack is unaffected. Certain functions (such as transpose and layering) can be programmed to operate on specific strings, and the String Select control allows independent selection of strings 1-6, strings 5 and 6 together, or all strings. A button immediately to the right of the display puts the GR30 into Edit mode, and once you're there, the parameter to be edited is selected using the two large dials, and the value incremented or decremented using the plus/minus buttons to the far right of the display. All the edit functions are clearly printed around the control knobs, so there's no need to keep referring to the manual.

In normal play mode, the four pedals may be used to switch patches or to access the Wah, Pitch Glide, Hold and Arp/Harm on/off functions. There are two modes in which the pedals may operate, indicated by screening on the panel, and the current mode depends on the setting of the S1 and S2 switches on the guitar pickup system: S2 accesses Performance pedal mode, while S1 is used mainly for patch selection. Patches may also be incremented or decremented using the S1 and S2 buttons exclusively, and bank switching can be accomplished via an external footswitch. However, if the GR30 is set to select patches directly via S1 and S2, the pedals remain in Performance mode.

In all, there are 256 patches, arranged as 128 user patches and 128 factory patches. As supplied, the user patches are copies of the factory sounds. Because of the limited display capability, patches are designated as a letter followed by two digits corresponding to Group, Bank and Number. There are four patches (corresponding to the four pedals) in a Bank, and eight Banks in a Group.

PEDAL FX

Pedal 1 serves as a Wah controller in Performance mode, but, since it is a switch rather than a pedal, the wah effect operates on an envelope rather than under direct user control. Several preset options are available, which include a rate setting so that the wah occurs over a short period of time rather than abruptly. It is also possible to set Pedal 1 to bring in vibrato modulation instead of wah and, again, there are several preset mod types to choose from. Each of these choices is saved as part of the patch data, along with a parameter for playing response,

the idea being that you could program specific patches to be suitable for finger-picking, and others for plectrum work. Pedal 2 controls Pitch Glide, and the rate of glide is selectable. There are nine preset rise- and fall-time patterns to choose from.

Pedal 3 functions as a Hold pedal, so that any notes sounding will continue to do so until the pedal is released. This is great for long Floydian intros where you play and hold a Gm string pad, then doodle over it with the regular guitar sound for 10 minutes or so. There are several Hold options, allowing you to hold both tones, just one of the tones, only the lower two strings, and so on. If the arpeggiator is being used, the Hold function operates according to the way the arpeggiator patch has been set up — for more on this, see the 'Take Note: GR30 Arpeggiator' box.

The fourth pedal switches the harmony or arpeggiator function on or off (if it is included in the current patch). It's also possible to call up the tuner function using a combination of pedal and pickup switch presses, providing the GR30 isn't in 'Patch Inc/Dec by S1/S2' mode.

The internal patch effects are based on chorus and reverb, though there are 18 different reverb types to chose from, and 25 chorus types with variations that stray into the realms of delay, flanging and special effects.

PATCHES

Each of the patches in the GR30 can be built up from either one or two Tones chosen from a list of 384; the degree of sound editing available is very limited. The two Tones can be adjusted independently for attack, release and brightness, arpeggiator or harmony functions can be set, and the pedal functions can be defined on a 'per-patch' basis. Effects can be added, though these comprise fairly conventional reverb and chorus-type treatments, and there's an option for a parameter of the sound to change with the guitar level envelope. The dynamic response of the guitar synth can be set to hard, soft, tapping, no dynamics, finger style or normal. Both volume and brightness can be made to follow the guitar envelope, if required.

Various 'Patch Common' parameters may also be set, including volume and an arpeggiator or harmony effect, and there's provision to specify the MIDI Send To and Receive From settings of an external device. Also part of the patch are independent transpose options for the two Tones, the balance between the two Tones, and the relative pan positions of the two Tones. Normally the synth sounds will follow string bending and hammering, but a Chromatic mode is available for emulating instruments that normally stay fixed in pitch, such as the piano or organ.

Edited patches may be saved to any desired user location; user patch locations may be exchanged, and patch data may be bulk-dumped via MIDI SysEx, for storage in a sequencer or MIDI data filer.

IN USE

Tracking is still the prime concern with MIDI synths and, when it's using its internal sounds, the GR30 is about as good as it gets. Unless your playing is excessively messy, tracking is both fast and accurate, though just occasionally you'll catch a ▶

▶ string with the side of your thumb and be greeted by a ricochetting whine as the synth tracks the rogue harmonic instead of the note you thought you'd played! There is some delay on the lower strings, but not enough to put you off playing fast bass lines — something that wasn't true just a few years back. In fact, unless you play very quickly, you probably won't notice the delay at all.

Strumming is followed with surprising accuracy, but only if you strum at a slow or moderate rate — the 'Pinball Wizard' intro is a definite non-starter! Trilling also works up to a point, but as you approach 'Just One Cornetto' speed, the system starts to miss notes, due to its inability to discriminate between successive pick strokes. Also very impressive is the accurate way in which vibrato-arm gymnastics are tracked, though all of these wonderful attributes are compromised to some extent when the outside world of plodding old MIDI gets involved. If you're working with a sequencer, you really do have to play cleanly and moderately. A further tip is to sit well away from the computer monitor, because the divided pickup is affected by interference in exactly the same way as a regular guitar pickup. The difference is that you don't hear the interference — you just experience its effects as the tracking accuracy takes a turn for the worse!

Chromatic mode works exceedingly well on organ or piano sounds, but as the system is designed to re-trigger whenever it sees a new pitch (from a hammer-on, for example), you have to be very careful how you lift your fingers off the strings, otherwise this is interpreted as a pull-off and the open string triggers. Perhaps a switchable option to ignore pull-offs to open strings would be useful for clumsy sods like me! Another feature I'd like to see is the ability to add vibrato to only the last note picked, rather than to all six strings. This is a trick I used to use with my old Roland GR500, which was linked to an external monosynth with a vib depth pedal fitted. However, the ability to follow the guitar envelope for volume or brightness is really most useful, and helps emulate some of the hybrid guitar/synth sounds possible from the old GR300.

The sounds for the basic Tones have been chosen largely to complement the electric guitar style of playing, with lots of piano, guitar, organ, bass and synth lead-type patches. There's also the usual set of brass, reed, strings and choir sounds, as well as many very effective pads, a few world sounds, and a few dance-orientated sounds. Sound effects are kept to a merciful minimum, and there's a good range of gentle, New-Agey washes, tinkly bells and ethereal voices. Though editing is limited, by layering these sounds, changing their envelope characteristics and adjusting their brightness, you can get a huge tonal range. The quality isn't bad either — about on a par with a decently-specified GM module, but

without the GM sound set.

The arpeggiator is most welcome, not only for electro-pop applications, but also for dance and ambient effects. It's very easy to use (unless you want to enter really fancy rhythmic stuff), and it sounds utterly convincing. The harmony generator is also a wonderful addition (see Harmony Angel: GR30 Harmonist' box) though, as always, you have to be careful to pick a harmony style that will work with what you're playing. You can also come unstuck if you wander too far from the main key or use oddball scales.

Both the Wah and the Glide are genuinely useful real-time functions, and I particularly like some of the softer, fatter Wah options that allow you to create very analogue-sounding filter sweeps, which work particularly well on rich pad sounds. Better still, some of the Wah options include an automatic triggered sweep, with which you can interact, using the footswitch. The Modulation option is part of the Wah menu, but I couldn't find any way to change the vibrato delay time, rate or depth, which is slightly limiting. Even so, the default values are generally useful. Compared with the GR09, the GR30 is far better suited to live performance.

SUMMARY

At around the same price as a half-decent effects unit or MIDI synth module, the GR30 is excellent value and works as well as any guitar synth I've heard, if you stick to the internal sounds. The arpeggiator, real-time controls and harmony functions work brilliantly, the operating system is so simple that anyone can knock up unique patches, and the tracking is both fast and positive. As mentioned earlier, tracking isn't quite so bullet-proof when you're dealing with external MIDI gear or sequencers, and for sequencing you need to have the option to be able to record on six different MIDI channels at the same time. Poly mode is very limiting (see 'MIDI' box).

The only real criticisms are in areas that could only be improved by increasing the price or the complexity of operation, or both. I'd like a better display with an edit grid, and the ability to name patches; a little more modulation flexibility, and a built-in expression pedal — but then I'd also like to win the lottery, live forever and have to keep putting the phone down on Pamela Anderson! They say you get what you pay for, but in the case of the GR30 you get an awful lot for very little outlay, even when you add on the extra hundred quid or so for a GK2A pickup system. **SOS**

MIDI

Like the vast majority of guitar synthesizers, the GR30 allocates a separate MIDI channel to each string (Mono mode) to allow independent string-bending; the MIDI bend range would normally be set to 12 semitones rather than the two or three usually adopted by keyboard players. This is necessary not only to accommodate bends, but also to provide sufficient range for hammers and slides.

MIDI data may be sent on any six consecutive MIDI channels. It's also possible to send all the data on a single MIDI channel (Poly mode), but this precludes the use of many guitar-specific playing techniques. Patch Change and Bank Change messages can be sent and received, and it's even possible to set different Program Change messages for each string.

HARMONY ANGEL: GR30 HARMONIST

The harmonist can work on either or both Tones, or on an external synth, just as the arpeggiator does. The user can select any major or minor key, and the GR30 will automatically work out and play a musically-correct harmony — providing you don't stray out of key. If you want to do fancy key changes during a song, you can use a pedal to switch from major to minor, and external MIDI notes may be used to set a different key in real time. The way in which the harmony is constructed depends on the harmony style chosen, and there are 12 possible intervals from which to choose, as well as a diminished option.

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SOS 2/97

STEREO LAB

STEREO MICROPHONE TECHNIQUES EXPLAINED

PART 1: HUGH ROBJOHNS takes a historical look at stereo miking techniques and explains the whys and wherefores of the various methods available.

The first documented stereo microphone system was used (entirely by accident, in fact) at the great Electrical Exhibition in Paris in 1881. A French designer by the name of Clement Ader was demonstrating some improvements to an early telephone system, and stumbled across what we would now call the spaced-microphone stereo technique! Unfortunately, no one realised the significance of Ader's discovery and he went on to invent the inflatable bicycle tyre before playing with aeroplanes, calling his first plane 'A vion', which became the generic name for aeroplanes in the French language.

Most of the development of stereo recording as we know it today happened in the very early '30s, and almost simultaneously in America and the UK. In the USA, Bell Laboratories were working on systems using spaced microphones under the direction of Dr Harvey Fletcher. Meanwhile, in the UK, a very clever man called Alan Blumlein, working for EMI, was developing an alternative system which relied on coincident microphones.

Both methods were years ahead of their time and both had advantages and disadvantages. It was not until the invention of PVC in the '50s (which

allowed micro-groove vinyl records to be produced) that either of these techniques were adopted commercially, but today both formats are alive and well, and are often used in concert with each other.

In this article, I'll be looking at what stereo microphone systems are trying to achieve, also taking a closer look at the coincident stereo ideas which have become the mainstay of many practical recording techniques. Next month, I'll talk about spaced microphone systems and combinatorial techniques.

WHAT IS STEREO?

The word 'stereophonic' is actually derived from Greek, and means 'solid sound', referring to the construction of believable, solid, stable sound images, regardless of how many loudspeakers are used. It can be applied to surround-sound systems as well as to simple two-channel techniques — indeed, in the cinema, the original Dolby Surround system was called Dolby Stereo, even though it was a four-channel system! However, most people are conditioned to think of stereo as a two-channel system, and this is the definition I'll adopt in these articles.

There are basically three ways of creating stereo sound images over a pair of loudspeakers:
• The first is an entirely artificial technique based on Alan Blumlein's work, and uses pan pots to position the sound images from individual microphones by sending different proportions of each microphone to the two channels.

THE HUMAN HEARING PROCESS

The whole idea of stereo recording is to try to fool our auditory system into believing that a sound source occupies a specific position in space. So how does our hearing determine the positions of sounds around us in real life?

Without getting bogged down in the psychology and biology of the subject, we use three principal mechanisms to identify the positions of sounds around us. The first and probably most important one is that of differing arrival times of sounds at each ear, followed by level differences between the ears for high-frequency sounds, and finally, independent comb-filtering effects from the outer ear (the pinnae).

Since our ears are spaced apart on opposite sides of the head, any sound source off to one side will be heard by one ear fractionally before the other. Also, because there's a large solid object between the ears (the rest of the head), a

'sound shadow' will be created at high frequencies (above about 2kHz) for the distant ear.

Both these mechanisms highlight the possibility of confusion between the direction of a sound source at any given angle in front or behind the listener, since both the timing and level differences would produce the same results for both directions. To overcome this ambiguity, an automatic reflex action causes us to instinctively turn or tilt our heads slightly and the resulting changes in timing and level immediately resolve the confusion.

The third mechanism was discovered relatively recently, and is the reason for the bizarre shape of the pinnae. (I always knew they had to be there for something other than supporting glasses and earrings!) As sounds arrive at the outer ear, some of the sound enters the ear canal directly, while some is reflected off

the curved surfaces of the outer ear and into the ear canal. Since the reflected sound has to travel fractionally further, it is delayed, and in combining with the original sound, produces a comb-filter effect, resulting in characteristic peaks and notches in the frequency response. These frequency-response anomalies depend on the particular direction of sound arrival, and it is thought that we build a 'library' memory of the comb-filter characteristics which can be used to help provide crude directional cues.

This whole concept of directional perception is the foundation of the sophisticated signal processing used in systems like QSound and RSS, which try to create surround sound information from a conventional two-channel stereo system. Modifying the frequency response of recognisable sounds to simulate the effects of the pinnae can trick us into perceiving sounds from locations outside the normal stereo spread between the loudspeakers.

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STEREO MIC TECHNIQUES

▶ • The second technique (and one we will look at in more detail next month) is the use of two or more identical but spaced microphones. These microphones capture sounds at differing times because of their physical separation, and so record time-of-arrival information in the two channels.

• The third system is that of coincident microphones, and this has become the backbone of all radio, television, and a lot of commercial stereo recordings. This technique uses a pair of identical directional microphones , each feeding one channel. The microphones capture sound sources in differing levels between the two channels, much like the pan-pot system, but this time the signal amplitudes vary in direct relation to the physical angle between microphones and sound sources.

COINCIDENT MICROPHONES

Blumlein developed coincident techniques to overcome the inherent deficiencies (as he saw them) of the spaced microphone systems being developed in America. Since our hearing

mechanism relies heavily on timing information (see 'The Human Hearing Process' box), Dr Harvey Fletcher thought it reasonable to use microphones to capture similar timing differences, and that is exactly what the spaced microphone system does.

However, when sound is replayed over loudspeakers, both ears hear both speakers, so we actually receive a very complex pattern of timing differences, involving the real timing differences from each speaker to both ears, plus the recorded timing differences from the microphones. This arrangement tends to produce rather vague positional information, and if the two channels are combined to produce a mono signal, comb-filtering effects can often be heard.

Blumlein demonstrated that by using only the amplitude differences between the two loudspeakers, it was possible to fool the human hearing system into translating these into perceived timing differences, and hence stable and accurate image positions. We all take this entirely for granted now, and are quite happy with the notion ▶

TERMINOLOGY, RIGGING AND CALIBRATION

Blumlein performed all his experiments using microphones with figure-of-eight polar patterns (only these and omnidirectional mics were available at the time). Most of the time, the figure-of-eight microphones were arranged at 90 degrees to each other, such that one faced 45 degrees left, and the other 45 degrees right. The angle between microphones is called the 'Mutual Angle', and 90 degrees is the most commonly used. It is possible to change the mutual angle over a small range, to adjust the precise relationship between the physical sound source positions in front of the microphones and their perceived positions in the stereo image, although the effect is often very subtle and few people find it necessary to make such adjustments.

The usable working area in front of the microphone is defined by the polar patterns of the microphones, and is called the 'Acceptance Angle'. The diagrams below show the typical acceptance angles for figure-of-eights and cardioids crossed at 90 degrees. Note that because the figure-of-eights are bi-directional, with opposite polarity lobes, they have two acceptance areas and two out-of-phase areas at the sides.

It is essential to calibrate the microphones and their channels at the desk before attempting to record anything in stereo. Even nominally identical microphones will have slightly differing sensitivities, and the input channels in the desk could be set up completely differently — so it is important to run through a line-up procedure (which is far quicker to do than to read — honest!)

What we need to achieve is identical signal levels in the left and right desk channels for a given sound pressure level in front of the microphones. The easiest and most accurate technique starts with setting the microphones' polar patterns to the desired response (if using switchable mics) and connecting them to two desk channels (or a stereo channel, if available). Turn the pan pots on paired mono channels fully left and right and use a fader clip (or some other means, such as a large bulldog clip) to mechanically fix the two faders together so they

track accurately. Rig the microphones one above the other with their capsules as close together as possible, and turn them to face in the same direction while someone speaks in front of them (about two feet away and at their mid-height, if possible, to ensure minimal level differences).
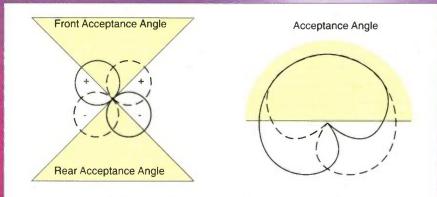
In the control room, switch the loudspeaker monitoring to mono (do not use the channel pan pots, because their centre positions may not be accurate), and adjust one mic channel for the typical operating gain you expect to need, with the fader in its normal operating position. Check that there is no EQ in circuit in either channel and switch a phase reverse into the second channel. Adjust the second channel's gain until the combined output from the microphones is as quiet as possible — there should be a very obvious null point (it will never completely cancel, because of inaccuracies in the microphones and desk channels, but it should get extremely quiet).

Next, remove the phase reversal and loudspeaker mono-ing, and with the two mics still facing forward, have your talking assistant wander in a complete circle all the way around the microphone array. If the stereo image moves away from the centre, the mics have incompatible polar patterns and will not produce accurate stereo images. Select another pair of

microphones and start over.

Finally, rotate the microphones to face 45 degrees left and right (make sure the microphone connected to the panned-left channel is turned to face the left of the sound stage) and have your assistant confirm the image boundaries and left-right orientation. Having completed the line-up, do not re-plug the microphones, or adjust the channel gains, as the calibration will be destroyed and you'll have to go through the entire process all over again! In practice, this whole procedure should take about a minute and should become routine.

A lot of engineers use a 'stereo bar' as a more convenient way to mount a pair of mics from a single mic-stand. Although this technique introduces small timing differences into the recording, it is a perfectly acceptable technique, provided the microphones face outwards rather than inwards after the line-up process. The reason for this is that each microphone casts a sound shadow at high frequencies across the other, and if they face inwards this is likely to degrade the stereo image (particularly if the mics in question are physically large, such as C414s, or U87s). If the mics face outwards, the sound shadow will fall on the rear of each microphone, where it is relatively insensitive anyway (assuming cardioid or hypercardioid patterns) and will not cause imaging problems.

Front Acceptance Angle

Rear Acceptance Angle

Acceptance Angle

Acceptance angles for pairs of figure-of-eight and cardioid mics crossed at 90 degrees.

Joemeek /dʒəʊ-miːk/ *noun*

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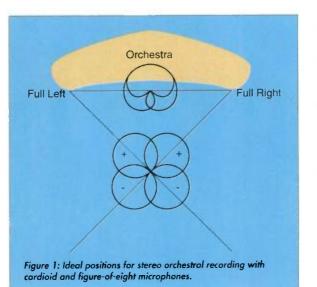
STEREO MIC TECHNIQUES

▶ that moving a pan-pot or balance control to alter the relative amplitudes of a signal in the two channels will alter its position in the stereo image in an entirely predictable and repeatable way.

This process is used every day to create artificial stereo images from multi-miked recordings, but contrary to popular belief, the level difference between the two channels which is necessary to move a sound image all the way to one loudspeaker is not very much. Typically, a 12 to 16dB difference between channels is sufficient to produce a full left or right image, and about 6dB will produce a half-left or right image — although the exact figures vary with individual listeners, the monitoring equipment and the listening environment.

To create stereo images directly from real life, Blumlein needed to develop a microphone technique which captured level differences between the two channels, but no timing differences. To avoid timing differences, the two microphones must be placed as close together as is physically

Figure 1: Ideal positions for stereo orchestral recording with cardioid and figure-of-eight microphones.

possible — hence the term 'Coincident Stereo'. The normal technique is to place the capsule of one microphone immediately above the other, so that they are coincident in the horizontal plane, which is the dimension from which we are trying to recreate image positions (despite hi-fi magazines' claims to the contrary, conventional stereo recording does not encode meaningful height information!). Amplitude differences between the two channels are created through the microphone's own polar patterns, making them more or less sensitive to sounds from various directions. The choice of polar pattern is the main tool we have for governing the nature of the recorded sound stage.

If you read books on stereo techniques, you'll find a variety of alternative terms used to describe the various methods in use. The kind of coincident stereo discussed here is also known as 'XY' recording (in America and parts of Europe), 'AB' recording (in the BBC and most other European broadcasters), 'crossed pairs', or just plain 'normal

stereo'. The term 'AB stereo' takes on a different meaning in the USA, where it is often used to describe spaced microphone arrays — beware of the potential for confusion!

PRACTICAL TECHNIQUES

In general, we aim to place sound sources around stereo microphones such that they occupy the complete stereo image. If you consider an orchestra, for example, it's usual to have the back row of the violins fully to the left, and the back row of the cellos or basses fully to the right.

To create this spread of sound using crossed cardioids to record the orchestra, it would be necessary to place them directly above the conductor in order to achieve the desired stereo image width. To take another example, crossed figure-of-eights would have to be positioned a long way down the hall to achieve the same stereo width (see Figure 1).

It should be obvious from these comments that in choosing the polar patterns for the microphones, you also determine the physical separation between sound sources and microphones for a given stereo width, and therefore the perspective of the recording. In the example above, the cardioids would give a very close-perspective sound, with little room acoustic and a distorted orchestral balance favouring the close string players over the musicians towards the rear and sides of the orchestra. In contrast, the figure-of-eights would give a much more natural and balanced perspective to the orchestra, but would also capture a great deal of the hall's acoustic, which might make the recording rather more distant than anticipated.

It's quite possible that neither of these basic techniques would produce an entirely satisfactory result, and a compromise might be to use crossed hypercardioid mics (with an acceptance angle of about 150 degrees). More likely, a combination of the two original techniques, plus a scattering of close 'spot' mics to reinforce the weaker sections of the orchestra (using pan-pots to match their stereo images to the main crossed pairs), would have to be used. The crucial point is that there is no absolutely correct technique, only an array of tools which you must choose and use to obtain the results you want.

COMBINING CROSSED PAIRS AND SPOT MICROPHONES

A very commonly-used technique is combining a crossed pair (to form the basis of a stereo image) with a number of close microphones (to give particular instruments more presence and definition in the mix). This applies equally whether we're talking about recording a philharmonic orchestra or a drum kit — only the scale of the job changes; the techniques do not.

There are three things to consider with this combination technique: image position, perspective and timing.

The main stereo pair will establish image positions for each instrument and the close microphones must not contradict this, if we're to avoid confused and messy stereo images. The best ▶

STEREO MIC TECHNIQUES

▶ technique I know for setting the panning for the close microphones is to concentrate on a particular instrument's image position in the main pair, then slowly fade up the corresponding spot mic and hear how the image changes. If it pulls to the right, fade the spot mic down, adjust the pan-pot to the left (or vice versa) and try again. With practice, you should be able to match image positions in three or four cycles, such that fading the spot mic up only changes the instrument's perspective, not its position.

Clearly, a microphone close to an instrument will have a completely different perspective to one further away, and this contrast is usually undesirable, as it draws undue attention to the instrument in question. The relative balance between the 'spot' mic and the main pair is critical, and it's surprising how little a contribution is required from the close mic in order to sharpen the instrument's definition, which is normally all you're trying to achieve. Remember, if you're aware of the close mic, it's too high in the mix.

The last point is relative timing, but this is usually only a problem with large recording venues. Consider an orchestral recording again, where the main stereo pair of, say, hypercardioids, may be 50 or 60 feet away from the orchestra. As sound travels at about one foot every millisecond, the sound from the stereo pair will be about 60ms behind that from any close spot mics. The human hearing system is geared up to analyse the first arriving sounds, which means we naturally tend to be aware of sound from the spot mics before the main pair — almost irrespective of how low they are in the mix. This is not the situation we want — the spot mics are supposed to assist the main stereo pair, not the other way around!

The solution is to route all the spot mics to a stereo group (having balanced and panned them appropriately) and send the combined signal to a stereo delay line. Dial in a suitable delay (one millisecond per foot for the distance between the main pair and the most distant spot mic, and then add five to ten milliseconds for good measure). The output of the delay line is returned to the desk and mixed in with the main stereo pair to produce the final mix. By delaying the spot mics, you can cause their signals to be heard after the main stereo pair (by the five or ten milliseconds that were added), and they'll consequently be much harder to perceive as separate entities. In fact, delaying the close mics makes their level in the mix slightly less critical, as the hearing process takes less notice of them, although

their panning is still crucial, of course.

This technique is extremely effective, but is rather time-consuming, and few people would bother with it if the main stereo pair was less than about 20 feet from any spot mic.

M&S COINCIDENT TECHNIQUES

There is an alternative coincident stereo technique, again developed originally by Alan Blumlein. This is the M&S, or Mid & Side, technique, mainly used by television sound recordists, but definitely worth knowing about, whatever you record.

M&S is a coincident technique in exactly the same way as the conventional systems already described. Instead of having directional microphones facing partially left and right, the M&S technique uses a pair of microphones, one of any polar pattern you like facing forwards and the other, a figure-of-eight, facing sideways. These two signals have to go through a conversion process before being auditioned on loudspeakers or

Figure 2: Pickup patterns for a cardioid mic and figure-of-eight mic arranged as an M&S pair.

headphones as normal left-right stereo.

The M&S system offers a number of practical advantages for television sound recordists (which are outside the scope of this article), but the single most useful aspect of the system for everyday recording tasks is that the perceived spread of sound sources across the stereo image can be controlled very easily from the desk.

The most common arrangement is to use a cardioid microphone facing forwards (the 'M' mic), together with a figure-of-eight microphone (the 'S' mic) facing sideways, and when these are converted into normal left-right stereo, they produce an identical acceptance angle to conventional crossed cardioids (see Figure 2). One important point to note: the polarity of the S lobe facing left should be the same as the polarity of the M mic. If this is not the case, the stereo image will be reversed.

As the balance between the M and S microphones is altered, so is the apparent distance between sound sources, as heard on the speakers (the effect is similar to adjusting the mutual angle between a conventional crossed pair of mics; see 'Terminology' box for more on this). This can be used to great effect, and it also allows the image width to be pushed outside the speakers by introducing an out-of-phase element to the signal,

DECODING M&S PAIRS

To decode the M&S signals to normal left and right, pan the M microphone to the centre and split the S microphone to feed a pair of adjacent channels (or a single stereo channel). Gang the two S channel faders together, pan them hard left and right, and switch in the phase reverse on the right channel.

Listening with the monitoring switched to mono, balance the gains of the two S channels

for minimal output (make sure there is no EQ switched into either channel). Once the two S channels have been aligned, revert to stereo monitoring, fade up the M channel and adjust the balance between the M and S signals for the desired image spread.

Putting a phase reverse in the M channel will swap the stereo image over — left going to the right and vice versa — and the image width can be varied from mono, through normal stereo, up to extra wide, simply by moving the S fader up and down.

although this should be used with great care, as it will affect mono compatibility.

SOUNDFIELD MICROPHONE

This concept of M&S was extended in the design of the Soundfield microphone and its baby brother, the ST250. These microphones were originally developed for Ambisonic recording — a technique which captures and reproduces true surround sound, with height information as well as 360-degree horizontal imaging (as opposed to the entirely artificial spatial positioning of the various cinema surround systems).

Unfortunately, Ambisonics has never really caught on and although a few companies are producing material suitably encoded (such as classical recordings from Nimbus), most people use the soundfield microphones as glorified, but stunningly accurate, stereo mics.

The soundfield microphones have an array of four cardioid capsules, arranged as the sides of a tetrahedron (two pyramids joined base-to-base), and these are combined electronically to produce four 'virtual microphones' called W, X, Y and Z. The first output (W) is designed to have an omnidirectional polar pattern, while the other three are figure-of-eights facing left-right, front-back and up-down. The way in which the W, X, Y and Z virtual microphones are created simulates extremely close spacing between capsules, so the stereo imaging is phenomenally accurate.

These four signals are combined together to produce a stereo output according to the settings on the control unit, in much the same way as the basic M&S arrangement described earlier. The omni (W) signal can be thought of as equating to the M microphone in a simple M&S pair, and the X, Y and Z signals equate to the S microphone, albeit with separate microphones for each direction (up/down, left/right and front/back).

The control unit allows the user to manipulate the Soundfield mic's characteristics to unprecedented degrees. The effective polar patterns of the simulated stereo pair can be selected, as can their mutual angle, and then this virtual stereo array can be pointed and tilted in any direction, simply by manipulating the way in which the four signals are combined. One of the most amazing aspects of the soundfield microphone is that by changing the balance between the W signal and all of the others, the mic can be made to appear to 'zoom in' to the sound source! It is even possible to record the four base signals individually (called the B-format) and then use the control unit to manipulate the microphone's characteristics on playback.

Next month, we'll look at spaced microphone arrays such as the Decca Tree and Binaural recording, as well as some of the more popular combinatorial techniques. SOS

Hugh Robjohns is a lecturer at the Centre for Broadcast Skills Training at BBC Wood Norton. The views expressed in this article are the author's own and are not necessarily those of his employer.

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RACK & ROLL

FREELANCE ENGINEERS & THEIR PERSONAL RACKS

It was once taken for granted that freelance engineers would use the standard outboard provided by the studio they were working in. But the explosion in the amount of gear available means that now they often have to assemble personal racks to be sure of getting access to their tools of choice. DAN DALEY explores the phenomenon...

O nce upon a time, the piano player was the only person who came to a recording session not carrying anything. Then the age of synthesizers arrived, and pianists became as encumbered as everyone else, weighed down with tons of gear. That left only the engineer hauling nothing heavier than a cup of coffee.

But times have changed once again. The rapid proliferation of outboard signal-processing equipment has led to an expensive, but seemingly necessary, trend among engineers: a personal equipment rack that travels with them from session to session. With studios finding it impossible to purchase everything the market demands, with project and personal studios rarely having the funds to buy more than basic pieces of equipment, and with much of today's gear having a distinctly ephemeral quality as technological fads come and go, engineers are finding that they need to be able to say 'have rack, will travel'. Talking to a number of US-based engineers on the subject, I soon discovered that not everyone is necessarily pleased with the new need to show up at sessions bearing what used to be the studios' responsibilities. However, other things also become clear: with a few exceptions, the notion of 'standard' gear has become quaint, to say the least. There is so much equipment around, in terms of signal processing, for example, that standardisation has become impossible — at least for more than a couple of months — as everyone rushes to acquire the flavour of the month. The flip side of that equation, though, is that you don't necessarily have to spend a small fortune on outboard — there's an increasing amount of gear available that sounds good and is inexpensive.

PERSONAL SERVICES

Jim Zumpano, an Atlanta-based engineer who has been the primary tracking engineer for legendary R&B production team LA Reid and Babyface (who have since parted company), as well as for Eddie Murphy's *Boomerang* film soundtrack and for singer Toni Braxton's records, says that the personal rack has become very important for engineers as they move from one studio to another. "There's so much gear out there, you never know what you're going to find — or not find — in a particular situation. Having a personal rack brings with it a comfort factor that a lot of people like: my cables, my mic preamps, and so on. You're more familiar with the way your signal processors work and what to expect from them. As you start working in these smaller studios, you never know what to expect in terms of outboard gear. Your own rack gives you consistency."

Zumpano's basic rack is loaded with a Neve 1081 mic preamp/4-band EQ and Neve 33609 and 32264 compressors. What's interesting, he says, is that as he makes the transition from almost exclusively MIDI-based R&B and dance work to more rock recording, his rack is changing. "With R&B, most of the backing tracks are MIDI-generated," he explains. "You're mainly recording vocals, so you're mainly concerned with mic preamps and compressors, and maybe EQs. I like the Amek/Rupert Neve 9098 mic preamp/EQ for that. But once you move into rock, the rack gets

> "The personal rack has become very important for engineers as they move from one studio to another."

bigger. At that point, I'm also carrying around things like SSL mic preamps, which are great for snares and drum overheads. And Tube Tech DI units for guitars and basses."

Zumpano likes the idea of being able to bypass the mixing desk completely with his rack gear, which is relatively simple for vocal overdubs. He'll set up a signal chain, typically ending at the compressor, and go straight to tape. On overdub

sessions, he simply goes from one track to the next by moving the patch cords on the studio patchbay. For track-laying sessions and mixes, he'll use a custom snake. "The less wire the better, but for tracking and mixing you have to have a more complex interface between your rack and the console," he says. One cautionary note that Zumpano sounds is that, while pro audio gear operates at +4dB, many project studios and some pro studios are using equipment that operates at the semi-pro -10dB level, necessitating level-matching boxes. "For instance, if you play an [original model] ADAT back through the quarter-inch outputs, it comes back -10," he explains. "If you send that to a +4 console, you'll have a level difference. You have to kick up the line trims and try to level-match the signals. The times we've had ADATs interfaced with the SSL at La Coco Studios [which Zumpano manages] I didn't notice a tremendous amount of noise resulting from the level mismatch. But it can become a problem if you're doing this with a lot of connections on a single track." Zumpano uses an Aphex stereo level-matching box in his rack, which also doubles as an interface for CD players when sampling.

Chuck Ainlay, who has engineered for Mark Knopfler, Wynonna Judd and George Strait, among others, has been a personal rack user for nearly a decade. "Basically, it started out as a need for reliable equipment," he explains. "Ten years ago in Nashville you never knew what you were going to get in terms of outboard gear at a studio." Ainlay's racks — he has several — are based around vintage Neve and other mic preamps, EQs and compressors. Secondary racks contain digital effects, including the Lexicon PCM70 and 300, and a rack full of samplers. He chooses a rack based on application: he'll bring the Neve racks for track-laying and overdubs, the sampling and effects racks for mixing. He also carries his own speakers — Dynaudio PPM3s.

Barry Sanders, former owner of Nashville's The Sanctuary Studios and now head of that city's Dreamhire rental outlet (where he has no shortage of outboard toys to contemplate), remembers that there was a significant increase in both the number and size of racks coming into his studio during his last few years there. "Even for demo sessions, I noticed that engineers were bringing in pretty extensive racks." Sanders mentions one Nashville engineer who used to pride himself on being able to make a session sound good with what was at hand, eschewing a personal rack. Even he's coming to sessions these days with a rack of gear, Sanders says: "It's a pretty rare day when the engineer doesn't walk in with something under his arm."

MARKETING PLOY?

Sanders, however, brings up another point. "I sometimes wonder how much of the gear is really necessary and how much of it is a marketing attempt on the part of engineers to increase their rates." In response, Ainlay agrees that this is part of it. "Nowadays people sort of expect it," he admits. "It impresses the client. You get the feeling that if you don't walk in with a huge rack, people think you aren't doing all that well. And it's a security blanket for the engineer too." Ainlay does

charge for the use of the rack — $100 per day, which he says is a bargain for the client, considering what it would cost to rent that amount and level of outboard gear. Clients also pay for the rack's transportation to and from sessions.

But rental houses are increasingly the beneficiaries of this trend towards more elaborate engineers' racks, notes Sanders. The hot items continue to be the classic ones, which may say something about the state of current digital signal processing. Sanders says demand never lags for

items such as the blackface Urei 1176, LA-2A and vintage Tubetech compressors, as well as vintage microphones such as U87s and U47s. The one new piece of gear that seems to be in consistent demand is the TC Electronic Finalizer [reviewed in *SOS* December 1996], a digital multi-function processor which Sanders says engineers are referring to as a 'mastering studio in a box'.

"It's especially popular with project studios because it gives them so many functions at once, and all in the digital domain," he says. "Other than that, I don't see any one new piece of gear dominating the market. The theory seems to be that the vintage stuff costs a lot of money but that there's always demand for it, so the rental houses will always carry it. decreasing the need for engineers to go out and buy it. On the other hand, the prices of the newer digital processors tend to be so low that it's not a problem for people to simply go out and buy them in order to check them out. If they don't care for them, or if their clients' demand for them diminishes, they're not out a lot of money and they can go and buy the next happening piece of gear that comes along. What people seem to want really does tend to shift from week to week."

GOING WITH THE FLOW

Some engineers have managed to avoid feeling compelled to acquire a personal rack at all — until recently, anyway. "I'm just getting ready to take the dive," said Jay Healy several months ago. Healy, a freelance engineer in New York who has worked with Mariah Carey, John Mellencamp, Billy Joel and REM, adds, "I had been able to resist it up to this point. When I was assisting, I noticed that the time it took to hook up an engineer's rack sometimes was as long as it took to bring in rental gear and hook that up. So I never saw a speed advantage in a personal rack. But it became

Chuck Ainlay, pictured with three of his personal mobile racks.

RACK & ROLL

▶ more apparent that you need one. So instead of accumulating one over the course of time, I'd been making a shopping list, bought everything all at once and did it right." That shopping list included a GML compressor, a Lexicon PCM80, Avalon 4-band parametric equalisers and a Yamaha SPX990. However, Healy says that he remains aware of another pitfall of the personal rack syndrome: "There's a tendency to use the same stuff and the same settings over and over on different projects and on every song. Things can start to sound a little similar to each other. Sometimes I like the idea of going into a new project and not knowing exactly what I'm going to use. It helps keep things fresh."

RACKED WITHOUT PAIN

A rack does not have to become attached at the hip, however. So says David Z, producer/engineer/mixer/arranger for a very eclectic assortment of artists, from early productions with Prince (for whom he produced early demos and singles such as 'Kiss' and engineered or mixed other records, including 'Purple Rain', in their shared hometown of Minneapolis), through Janet Jackson and Sheila E, Billy Idol and Fine Young Cannibals (whose 'She Drives Me Crazy' garnered Z the Grammy nomination for Best Producer in 1990), to, most recently, the master of eclecticism himself, guitarist Leo Kotke.

"I carry a bunch of gear, but it really depends upon the project," says Z. "If the studio has what I think works for the record, that's fine. If not, I just add stuff of mine that I think I'll need. There's not a set formula as to what gets used. The thing to remember is that you're making a record of the artist, and you want to make it sound like them, not your rack."

Z's personal arsenal includes items such as vintage API 550 EQs, whose number he augments via rentals if he needs more for mixes; the same goes for Focusrite EQs. He also uses his vintage Emu SP1200 drum machine and his vast and growing collection of home-made samples to customise records. The rack's pedigree drops precipitously from there, though; Z is an aficionado of cheaper gear, ranging from Yamaha SPX90s, which he likes to use as a chorus on low-frequency instruments like kick drums and bass guitars, to the Musitronics Mutron and other pedal-like devices that he throws in as needed. He also uses a Tech 21 Sansamp, as an added distortion device for guitars and basses.

"One of the key items, though, is a set of Keypex gates," he adds. "I have them wired for gate triggering through the key input. I often synchronise things to each other on tracks, especially on mixes, so the Keypex is critical. In fact, on Prince's song 'Kiss', the rhythm is coming

David Z.

Fernando Kral.

mainly from a 12-string guitar sync'ed via the Keypex to the record's hi-hat track."

SOUNDS INDIVIDUAL

When asked if he plans to acquire a personal outboard rack, engineer Fernando Kral (Joe Jackson, Talking Heads and Jane's Addiction) responds "I decided to buy a house instead". He's aware of the emphasis being placed on personal racks, starting in the late 1980s, but after assisting producer/engineers Bob Clearmountain and Hugh Padgham, neither of whom used a personal rack back then, Kral decided that he'd rather invest in microphones. However, as his engineering career began to include more overseas work, including Europe, South America and Asia, Kral has come to feel the need to have gear he's familiar and comfortable with at hand. In a way he's relieved, he admits. "Until now, I've always had to be extra careful in booking studios I wasn't familiar with, and I was less likely to go somewhere I hadn't been before, simply because I wasn't sure of the outboard and how well it was maintained. But now that I'm going overseas more, it's harder to come by the esoteric signal processors, like Echoplexes and Boss choruses, that I've come to rely on. And as studios get pressured on rates and pare down, and as budgets eliminate some rentals, it's nice to know I'll have the tools I need for sessions."

But to this day Kral resists the notion of a formal rack. "I understand why people feel the need to have them," he says. "But I also see racks becoming the equivalent of marketing tools for engineers — 'I get this sound and this is the gear I get it with'. That sort of thing. That's OK too, but I find that these days I'm working on such a wide range of budgets for record projects that the idea of a rack simply doesn't give me the sense of security that you might think it would. I find that if I have to do a $30,000 or $40,000 record in a smaller studio that doesn't have the same level of outboard equipment as a larger, more expensive studio might, then that's alright, too; the studio's sound simply becomes the sound of the record. It's a lot more organic and in some way more sincere that way. The record isn't in the outboard rack. It's in the band and the walls of the studio." **SOS**

The profusion of new Macs may have you wondering which one to go for. MARTIN RUSS helps out, pausing only to dispense a spot of advice on optimising your serial port use...

I f, like most of us, you use your Mac OS computer for more than just music, its serial ports will tend to be used for more tasks than there are ports. There are a number of compromises that may need to be made — or else you'll need to do some re-plugging of cables. Here are some of the major points to consider:

• If you have a simple MIDI Interface and a non-AppleTalk (serial) printer, connect the MIDI Interface to the Modem port, and the printer to the Printer Port. Deceptively simple.
• If you have a multi-port MIDI interface which needs both serial ports, using a printer means

Chooser enemies.

that you'll have to remember to change some switches on the MIDI interface every time you want a print-out. Trying to print to a MIDI interface by accident when you've forgotten to change over the switches can result in a frozen Mac which will have to be restarted.
• If you have a multi-port MIDI interface which uses both serial ports, and an AppleTalk printer, not only do you need to remember to change the switches on the MIDI interface, you'll also need to turn AppleTalk off when you use MIDI. And put the switches back again to use the printer!
• If you're using AppleTalk between networked Mac OS computers and File Sharing is active (Check the 'File Sharing' Control Panel), then this can affect MIDI — MIDI Manager is very susceptible on my Centris 610. The best advice is to turn off File Sharing before you use MIDI.
• Changing between MIDI Manager and OMS or FreeMIDI is often required because some pieces of software will only work using MIDI Manager (Yamaha's otherwise excellent freeware VL70m editors are a recent example). This is not always straightforward. I use Extensions Manager 2 to make it easy to move System Extensions and Control Panels around, but the basic rule runs like this: *for MIDI Manager, remove the OMS or FreeMIDI Extension and re-boot.*

Now I know that MIDI Manager, OMS and FreeMIDI are supposed to tolerate each other, but this strategy works for me almost every time. Every other strategy seems to waste my time, including this favourite one: *remove all the extensions and re-boot.* This is fine for solving many Mac problems — unless you want to use MIDI.

BUT WHICH ONE?

There are now so many Macs and clones, and they change so rapidly, that it is becoming increasingly difficult to decide which one to buy — rather like buying a PC, in many ways. Competition has a nasty tendency to provide too much choice!

So which Mac OS computers should you be looking at for use in musical projects? Perhaps the first piece of advice is to carefully consider avoiding the new PowerMac 4400 — although made by Apple, this Mac is very much a PC in Mac clothing, since it uses rather more PC-type technology than most other Apples. I have heard stories of confusion about its use of EDO memory, for example. But the clincher has to be the PC-style box, which is probably the closest Apple have ever come to a plain box. Price is not everything.

Moving along to the Performas, the tower-styled 6400 in its various guises will probably catch your eye. The 3D audio and built-in woofer may not be as important to musicians as Apple might like to think, and the bundling of *Cubasis AV* in the top-of-the-range 'Creative Studio' model may not suit a purchaser who already has digital audio and MIDI sequencing software. But, at the price, the combination of processing speed, built-in Internet capability, a nice selection of bundled software and a CD-ROM may be hard to resist — it's very high on my shopping list.

Ethernet networking becomes standard in the PowerMacs and many of the clones, but unless you intend to have your Mac OS computer as part of a network, it isn't really needed. There's a tendency for the Performas

to have better clock speeds than the PowerMacs, and although the Performas use the 603e processor rather than the PowerMacs' 604 or 601 processors, the actual differences in processing power don't seem to be that great — we're talking percentages rather than several times faster. The one exception is the PowerMac 4400, which uses the same 603e processor chip as the Performa 6400, but at a slower clock speed. The conclusion seems to be that unless you need a very expandable, dual-channel SCSI, Ethernet-networked workhorse with more than two PCI slots, for digital audio or hard disk recording, the PowerMac approach may be over the top.

ON THE NET — NOT!

The Internet is chock-full of nice little pages carefully hand-crafted by enthusiastic individuals — and consisting of little more than links to other pages. These themselves are often quite interesting pages, but sometimes *they're* merely more pages of links to other

"So which Mac computers should you be looking at for musical projects?"

pages... As if this wasn't bad enough, there's also the dreaded 'Add Bookmark' or 'Add to Favourites' command, which notes down the URL (Internet address) of the current page and adds it to the huge list that you've already accumulated. If you do manage to sort through all of your bookmarks, just about the only thing you can do with them is turn them into a page of your own — of links!

But there is a way to stop this time-wasting behaviour. Not only do links go out of date because pages move, never get updated, or vanish into the twilight zone, but trying to find the link you half-remember out of a huge list of bookmarks wastes time: hours and hours of it. There are some who claim that the Internet

is nothing more than an excuse to become a computer couch-potato, and producing pages of links from bookmarks probably reinforces this. But there is a way out of link fetishism: Don't store any!
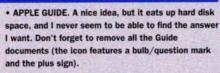
Cold Turkey is the way forward: *do not* store bookmarks to interesting pages so that you can visit them again at some unspecified time in the future, because there are already more pages out there in cyberspace than you could ever visit in 10 lifetimes. And no-one ever really needs to revisit a page, anyway. Instead, use a searcher, like Digital's AltaVista site, and find pages by typing in key words. This method avoids most of the 'pages that are no longer with us' and very often locates the page you were looking for, or a close approximation. But, best of all, it often throws up pages with no apparent connection at all with the subject you were looking for — which is, of course, the basis of all successful TV soap-operas and sit-coms.

Check out **www.webpagesthatsuck.com** for how not to do it!   **SOS**

TIP OF THE MONTH: EX-EXTENSIONS

Your Extensions folder is almost certainly full of them. But what do they all do? Do you actually need them all? Probably not. Here are some likely candidates for moving to a 'Disabled Extensions' folder inside your System Folder, or better still, to a Non-System Folder, perhaps not even on your hard disk at all.

• SPEECH MANAGER. Do you need your computer to talk to you? No. And in my experience it hates MIDI too. Two reasons not to have it.

Do you need these Extensions?

9 items   190.3 MB in disk   40.6 MB available

Apple Guide   TokenTalk Prep   Sound/Monitors Guide Additions

A/ROSE   Apple Built-In Ethernet   Personal LaserWriter SC

LaserWriter   TokenTalk Phase 2   Shared Library Manager PPC

• APPLE GUIDE. A nice idea, but it eats up hard disk space, and I never seem to be able to find the answer I want. Don't forget to remove all the Guide documents (the icon features a bulb/question mark and the plus sign).

• QUICKTIME POWERPLUG. If you haven't got a PowerPC processor, you don't need this, but you would be surprised at the number of 68K Macs that have it busily doing nothing except occupying space.

• PRINTER DRIVERS. Few of use have more than one or two printers, but most Extensions folders have a huge selection of printers that you can't use. If you have a Stylewriter, you need the driver for that and nothing else.
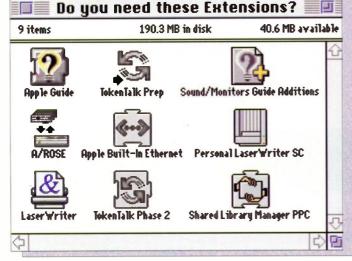
• PRINTER SHARE. Do you need a networked printer?
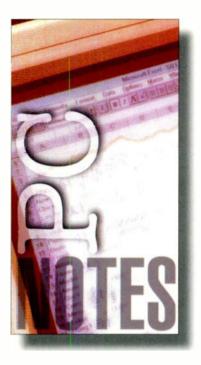
• NETWORKING FILES with names including words like TokenTalk, EtherTalk and Ethernet are only useful if your Mac OS computer is connected to a network using that protocol. If you use a MIDI interface and a

serial or AppleTalk printer, all of these can be moved to an alternative location.

• A/ROSE. This is a good one. I've never had any NuBus slots or cards in any of my Macs, and yet this mysteriously-named extension is invariably installed each time I update the Operating System. If you haven't got NuBus cards, you don't need the Apple Real-time Operating System Extension (A/ROSE).

• THREADSLIB OR THREADS MANAGER. You only need one of these if you have a pre-System 7.5 operating system. ThreadsLib is for PowerPC machines; Threads Manager is the 68K equivalent.

• MATHLIB. Not required for 68K Macs.

• MONITOR-SPECIFIC FILES. Any file with a name like LC Monitors Extension, Quadra Monitors Extension or IIsi Monitor Extension points to a specific model or range of Macs. If you haven't got one of them, you don't need the file!

• APPLESHARE AND FILE SHARING EXTENSION. Only needed if you want to share files (or applications!) over a network. File sharing is not recommended for anyone who also uses MIDI. If you do need both, be prepared to do lots of re-booting.

• NETWORK EXTENSION. If you have just a serial (non AppleTalk) printer, you don't need this.

There are other 'not needed' files. More information next time.

Is it worth investing in a stand-alone CD-R recorder? BRIAN HEYWOOD explains how it can make life easier when you're burning gold...

Anyone who has followed these columns for a little while will know that I'm quite a fan of CD-R recorders. If you're a PC user, the cheapest way to start making your own CD-Rs — known as 'gold discs' — is to add a drive such as the Yamaha CD102 to your existing computer setup, but this may not be appropriate if you work in a busy professional studio with tight time-scales or need to do a lot of small-run CD duplication. Why is this? Well, the problem with burning CDs from your PC is that the machine is completely tied up during the whole process of creating the gold disc — one of the 'features' of CD creation is that you can't interrupt the Write process without destroying the disc. This means that even with a double-speed writer you would need to tie up your PC for at least half a day to produce half a dozen CDs. If your PC is the heart of your studio, this may mean that you can't do any

other work while you carry out the duplication process.But there is a solution to this dilemma: enter the Marantz CDR620...

A stand-alone CD-R recorder such as the Marantz has the huge advantage that it will let you perform the CD duplication off-line. In fact the CD-R620 gives you the best of both worlds, as it can operate as an autonomous CD-R recorder, as a peripheral CD-R drive via the built-in SCSI-2 interface, or even in tandem with another unit for double-speed copying via SCSI. You can even use the CDR620 to perform basic quality control on blank media, because it checks the pre-groove when you insert a CD into the tray and will reject any discs that don't meet the specification.

HOW IT WORKS

The CDR620 looks like — and is designed to operate in the same way as — a traditional tape-based recorder, but with additional buttons to give access to the extra functions required for creating CDs. So you could, for instance, copy your master to DAT and then simply stream the digital audio into the CDR620 whenever you need to make a CD copy of the material. If you need to do a lot of small-run CDR duplication then it might even be worthwhile to buy two machines and use the double-speed copying facility. The CDR620 isn't just a stand-alone unit, though: it can also function as a computer peripheral CD-Recorder. To use it in this way, you simply connect the unit to your PC via a SCSI-2 port and select SCSI mode from the front panel. The CDR620 emulates a Phillips CDD2000 CD-R drive, so it should be compatible with almost all the currently available CD-R mastering software and

software/hardware setups (such as TripleDAT, *Gear* and SADiE, for instance). Under SCSI control the unit supports a wide range of CD formats including CD-ROM(XA), DC-I, Photo-CD, Video-CD, CD-Plus and, of course, CD-DA (Red Book digital audio).

CONNECTIONS

The unit is pretty simple to operate and has a large fluorescent alpha-numeric display which gives an extremely clear readout of the operating modes and levels in a wide range of ambient light and viewing angles. The back panel holds the connectors for the analogue audio (XLR), and the digital audio (AES/EBU XLR and S/PDIF RCA phono cinch), as well as SCSI-2 (high density D-type), the remote control interfaces (D-type), and a standard IEC

> "The problem with burning CDs from your PC is that the machine is completely tied up during the whole process of creating the gold disc."

Whizz Jones gets his face on the internet in his Music-Network Classified advert — and it's free!
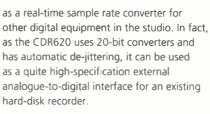
Web — and at a later date on CD-ROM. And this all makes the Music Network web site a more vibrant and interesting one for people to visit, which I guess is the whole point of the exercise.

The free entry consists of 40 words of text, one image (a photo or logo) and a 40-second audio clip.

If you're more ambitious you can buy extra space for more text, more images, a longer sound bite or even a 15-second video clip. To find out more about the service, call the Music Network on 0171 625 8758 or email entries@music-network.com. If you want to check out their web site, use the URL: http://www.music-network.com/.

power connector. Here you'll also find a set of four DIP switches for setting the SCSI address and termination. The only connector on the front panel is a standard stereo jack socket for headphones. One convenient feature is that the unit has built-in sample-rate conversion circuitry, so the digital inputs can take a wide range of input sample rates (from 32kHz to 58kHz). As the CDR620 can be operated in monitor mode, this means that you can use it

as a real-time sample rate converter for other digital equipment in the studio. In fact, as the CDR620 uses 20-bit converters and has automatic de-jittering, it can be used as a quite high-specification external analogue-to-digital interface for an existing hard-disk recorder.

REMOTE OPTIONS

While it's relatively expensive compared with CD-R drives designed to be connected directly to a computer, the CDR620 has a number of advanced features that can only be implemented in a stand-alone unit. The comprehensive remote control functions mean that the unit can be installed in a machine room, and the independent copying and recording functions could be a real boon in a busy facility where you don't want to tie up a studio when you need to perform these mundane tasks. The ability to operate independently also gives the unit applications in live recording and logging applications for

> **"A stand-alone CD-R recorder has the huge advantage that it will let you perform the CD duplication in real time."**

which a computer-based system wouldn't be suitable. In short, a versatile machine with lots of uses.

CONCLUSION

The basic CDR620 with remote controller costs £3,519.13 (inc VAT) — which may seem pretty expensive compared with a computer CD-R drive, but by the time you've added up the equivalent cost of the additional features it starts to look like quite good value for money. For more information contact SCV London on 0171 923 1892 (fax: 0171-241-3644). ▪

Atari NOTES

DEREK JOHNSON reports on a successful French Atari show, whets your appetite for the current issues of the Atari mags, and rounds up more software news and tips.

As an Atari user myself, I probably shouldn't be surprised at the platform's refusal to die, but it never ceases to amaze me that the ST and all its offshoots still manage to support magazines, web sites and, amazingly, shows. The shows run last year in the UK by Goodman's PDL are likely to be followed by more this year, and Germany is also a hotbed of this kind of thing. Early in December of last year, a large two-day show was run at the Bervy-Expo in Paris; the 2500 visitors to the second *Forum des Applications Atari* were treated to first-hand experience of Hades, the super-charged Falcon clone from Swiss Precision mentioned in last month's Atari Notes, plus the start of a new range from French company Centek: a pair of accelerator

boards (the 25MHz CENTurbo I and the 75MHz CENTurbo II) and a stand-alone computer, the Phenix, available with 66MHz 040 or 120MHz 060 CPUs. There's no news yet on whether they will be available in the UK, but on the show front there are already plans to run a third *Forum* next December. Thanks to French magazine *Keyboards* for the information.

'ZINE SCENE

The second issue of the sole remaining UK print magazine dedicated to the Atari in all its incarnations is out now. The 64-page *Atari Computing* magazine — a disk full of software is available as an option — offers loads of features and reviews of commercial and shareware software. Issue 2 includes reviews of Alexander Clauss's *CAB* GEM-based web browser and Michael Guse's *MG-Ftp* ftp client — just what the Atari user with a modem and an Internet account needs. On a musical level there are reviews of a couple of chip music packages and Ben Hall's *Dump-It* System Exclusive utility. There's also a review of a new version of *Gem-Rec96*, a 2-track direct-to-disk recorder for the Falcon 030. A subscription to *Atari Computing* costs from £9 (£11.50 Europe, £12 North America, £13 other) for three issues without reader disk to £30 (£35, £36, £38) for six issues, including disk. A sample copy costs £3. For more info, contact the magazine at Rois Bheinn, Overton Crescent, Johnstone, PA5 8JB (or email ataricomputing@cix.co.uk.com).

We've also mentioned *AtariPhile* before in these pages; this is the on-line magazine for Atari users, whose fourth issue was posted around Christmas — which accounts for the festive lighting on the main page! The 'magazine' manages to maintain some of the cohesive feel of something that's been printed, and is once again packed with information. This issue sees an increase in music coverage, with an article on scoring with Steinberg's *Cubase*, the start of a regular MIDI column and the first part of a music and MIDI series. The best place to check out *AtariPhile* issue 4 is at the web site (http://walusoft.co/fff, or ftp at ftp.metkonect/pub/atari/atariphile/), but those without Internet access can obtain a disk version of the mag from Colin Fisher-McAlum, at 11

Pound Meadow, Whitchurch, Hants RG28 7LG. Send a pair of formatted double-density disks, plus an SAE and two first-class stamps per issue; non-UK residents should send International Reply Coupons. PD libraries are also handling the mag (try Goodman's on 01782 335650; Floppyshop ST on 01224 312756; or the FaST Club on 0115 945 5250).

THINGS THAT GO DUMP IN THE NIGHT...

As promised after last month's mention of the *Dump-It* SysEx management and conversion utility, we now have details of the software's availability: it can be ftp'd from ftp.cnam.fr, nic.funet.fi, and Hensa (ftp://micros.hensa.ac. uk/micros/atari/atari.html. Note that because Hensa is an educational site with a heavy load during the daytime, access is only available between 8pm and 8am, UK time. For those without an Internet account, check Goodmans PDL, who should be able to help. And just to remind you, you can still browse *Dump-It* author Ben Hall's personal home page at http://www.city.ac.uk/~cb170/home.html.

Ben also told us about a new sample utility for the ST that should be available shortly. *SampleC*, by Richard Evans, will offer librarian, conversion and MIDI Sample Dump Standard facilities, with a similar interface to *Dump-It*. It's looking good, and we'll let you know where you can get it as soon as we know more. **SOS**

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WRH

This American classical composer writes rock songs as well as concertos and has a home studio complete with contemporary synths and a computer-based sequencer. PAUL TINGEN talks to him about the best of both worlds.

There was a time when classical music and rock music were continents separated by a virtually uncrossable ocean of prejudice, sectarianism, incomprehension, and musical languages so different that even attempts at communication between the two were doomed to fail. Many such attempts have been made over the years, from Deep Purple's ill-fated *Concerto for Group and Orchestra,* and ELP and ELO's flirtations with classical themes and instruments, to more recent efforts, such as Elvis Costello's collaboration with the Brodsky Quartet, and the Kronos Quartet covers of Jimi Hendrix tunes.

Although more recent crossovers between classical and popular musics are far superior to the awkward efforts of the '60s and '70s, the two genres are never completely comfortable bedfellows. To most rock musicians, the stylistic, technical and aesthetic demands of classical music are a closed book, and any efforts to incorporate them tend to sound pretentious or pompous. And few classical composers have succeeded in incorporating rock influences, despite classical music's long history of taking popular music as its inspiration

— look at Vaughan-Williams' use of English folk music, the jazz-influenced experiments of Ravel and Stravinsky, and Bartok's beloved Balkan folk music. What the 20th century has produced in the main is classical composers who employ some of the *tools* of rock music, using synthesizers to play classical music, like Wendy Carlos and Tomita, experimenting with electronics, like Stockhausen, and utilising the tape loop and sampler, like Steve Reich.

During the late '80s and '90s, however, a new wave of classical composers has emerged: these composers write genuinely in the tradition of classical music, yet are informed by rock aesthetics, and use popular music instruments and rhythms in a way that's more organic and integrated than ever before. In Britain these are composers like Steve Martland and Jonathan Harvey. A South Bank concert last Autumn showcased American composers Javier Alvarez (wildly and effectively experimenting with latin and jazz rhythms and harmonies), and Frank Zappa (exploring orchestral composition through abstract, atonal pieces with a rock edge).

POST-STYLE

And then there was, and is, John Adams. Though virtually unknown outside the classical music sphere, he is nevertheless regarded as one of the world's leading composers and is America's most frequently-programmed.

John Adams
CROSSING BORDERS

The South Bank concert mentioned above also saw the world premiere of his piece *Gnarly Buttons*, for solo clarinet and ensemble, and the London premiere of six instrumental pieces from *I Was Looking At The Ceiling And Then I Saw The Sky* (1995), a collection of 25 rock songs. Both works were played by the London Sinfonietta, and included samplers and synthesizers, plus additional electric guitar, bass and drums (for *I Was Looking At The Ceiling And Then I Saw The Sky*). Both works also revealed a classical composer reaching striking new levels in dealing with the heritage of rock 'n' roll. *Gnarly Buttons* falls very much into the tradition of other instrumental John Adams pieces, such as his violin concerto (1993) and his awesome *El Dorado* (1995), in that Adams has managed to blend the textures of synth sounds and samples with the timbres of the orchestra in a completely organic and natural way. Indeed, it's often hard to tell the samples and synth sounds from the orchestral sounds.

Not so with *I Was Looking At The Ceiling And Then I Saw The Sky*, a theatric song-cycle piece based on the lyrics of the African-American poet June Jordan. This is a radical departure from Adams' normal work, in which he attempts to write genuine rock songs, based around mainly black music styles like R&B, gospel, soul, blues, doo-wop, and so on — a bold endeavour for a classical composer. Thus it was that the Queen Elizabeth Hall was treated to the unusual sight of a classical ensemble having to play heavy-duty R&B brass parts and tight percussion, and accommodate a screaming electric guitar and midriff-punching bass.

Before the concert, Adams, a greying figure with horn-rimmed glasses and gentle eyes, who looks 10 years younger than his 50 years, held a public talk in the foyer of the QEH. He remarked, with a wide grin: "I'm often asked how I liked writing pop songs, and I answer that I found it the hardest thing I ever did. When you're an opera composer, and especially a minimalist opera composer, you'll take a long time to get off the runway, maybe 20 or 30 minutes. People will indulge you because they pay an enormous amount of money [public laughter] and they figure they're going to be there for four hours anyway. If they don't understand it, that's OK, because it's opera! [More laughter.] But with rock 'n' roll and pop songs it's completely different. People expect the message right up front. They want the very best idea in the third bar of the piece, and if they don't get it, the song is a complete bore-out, not good. You have to state your point immediately, and it also has to be intelligible. It was a very hard thing to do, and sweating out 25 songs was a long march."

Asked how this experience sat with his 'regular' music, Adams commented that "we're in a very interesting historical period. I think we're post-style. We're in a period during which we can't really place a stylistic label on the most interesting composers any more. All these pioneering, avant-garde inventions that happened earlier this century — like 12-tone music, aleatoric, minimalist music — but also popular music styles like jazz and blues and rock, have all spent themselves in their pure form. There's a vast synthesis happening at the moment. All genres are beginning to collapse, and the best testimony for that is to go into a large

music store and see displays of CDs by Hildegard von Bingen right next to Arvo Part, Steve Reich and music from Bali. And when you ask for a CD you can witness the frantic hysteria of the shop assistant: 'Is that New Age Crossover Hillbilly or something?' There's so much information coming at us that we can't process it."

AMERICAN SOUNDS

John Adams himself must shoulder some of the blame for the confusing state of contemporary music, for he has long been one of the more innovative and genre-crossing composers on the classical music scene. He grew up on the American East coast, where he learnt to play the clarinet, and studied music and composition at Harvard University. After moving to San Francisco in 1971, he was active as a music teacher and conductor at the San Francisco Conservatory of Music and became the San Francisco Symphony orchestra's first composer-in-residence. It was during the '80s that Adams sprang to international prominence, with minimalist-inspired instrumental works like *Harmonium* (1981) and *Harmonielehre* (1985), and especially his two operas, *Nixon In China* (1987), which won a Grammy award for Best Contemporary Composition, and *The Death Of Klinghoffer*, which was premiered in 1991.

Adams has always been categorised as a minimalist composer, something he called "a label that stuck to me like one of these monsters from *Alien*" during his pre-concert talk in London. A few weeks later, via a lengthy transatlantic phone conversation from his Berkeley home, he explains: "I never considered myself a minimalist. My early pieces were evidently influenced by minimalism, and people therefore grouped me in their minds with Steve Reich and Philip Glass, and they still do today. But I've always considered minimalism only as one of the many styles that I'm using. I moved away from it very strongly in the early '90s, with the opera *The Death Of Klinghoffer* and my violin concerto, in which my music became more chromatic and more harmonically complex. I've always been looking to extend my music expressively, using melody and lyricism, and taking a great deal of inspiration from jazz, rock and ethnic music. No open-minded composer can neglect the folk music of his or her time."

Even though Adams quotes a number of European composers who have incorporated popular music into their styles, he goes on to say that he feels that the more obvious appropriation of popular music in classical styles is "a key element in American art music, like Charles Ives using ragtime music, or George Gershwin using black music." Another "key element" of American music in general, according to Adams, is "the unwillingness to accept the heritage of European music wholesale, especially in terms of sound. We have concluded that music is not just about the way the composer puts pitches on the five lines on paper, but as much about the choices of the sounds themselves. We are the country of the real mavericks, like Harry Partch, who invented his own instruments and tunings, and we were the country in which most amplified and electronic instruments were invented, and their usage pioneered."

"I think that the main problem with sequencers is that they're largely used by people with very little musical ability, to create music that may appear to be accomplished and polished, but is in fact very superficial."

John Adams ▶

SEQUENCING IS WONDERFUL

And with that Adams leads us straight into a subject that is *SOS* home territory. For it turns out that he doesn't only conform to the profile of the archetypal 20th century American classical composer by taking his inspiration from both art and popular music sources, he also has a keen and long-standing interest in electronic instruments. This interest runs so deep that he even taught himself electronics: "My first encounter was in 1969 when the music department of Harvard University bought a Buchla synthesizer, which was the creation of an engineer called Donald Buchla. It was an analogue synth similar to the Moog, but creatively more imaginative. I immediately began composing pieces on the Buchla. After I moved to California in the '70s, I became very interested in what might be called low-tech electronics. Many American composers had built their own electronic instruments and created pieces with these, and that was my goal as well. So I built my own synthesizer, using analogue circuitry such as integrated filters and circuits."
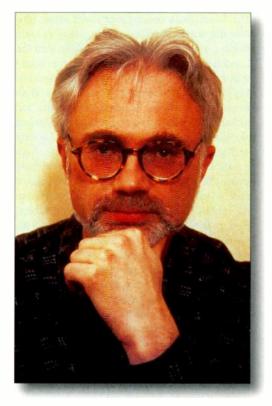
Adams admits that his self-built synth has since been "cannibalised for parts", but adds that "this was a very critical period for me creatively, not only because of the use of synthesizers, but also because I became intensely aware of how sound can be manipulated by gates and filters and oscillators. This had a profound effect on the way I viewed compositional structure and form. It showed me that there were other ways of creating musical pieces besides the conventional approach of harmony and melody, or using 12-tone rows, or whatever. I became very interested in making the form of my instrumental pieces imitate what went on in electronic circuits. Some of these pieces have titles that suggest electronic activities, like the string orchestra piece *Shaker Loops* or the piano piece *Phrygian Gates* — the loops being a reference to tape loops, and the gates to voltage-controlled amplifiers. With the exception of *Harmonielehre*, they're the only two pieces that don't have any electronic instruments. But the music behaves like electronics. *Shaker Loops* is built around the idea of loops, or sequences, of oscillating melodic cells, whereas the music of the piano piece behaves as if it were coming out of a synth circuit."

Adams agrees that the technique of composing with tape loops, as Steve Reich did in his influential voice piece *It's Gonna Rain* (1965), or with sequencers, is attractive for anyone working in the field of minimalist music. "But I use the technique in a very different way. I never made tape-loop pieces like that. I was more interested in mimicking the behaviour of electronic instruments in my instrumental pieces. And the arrival of the sequencer has opened up all sorts of possibilities

> **"I love working with electronics so much that I wanted to create an album that had nothing but synthesizers and samplers."**

that one could do with just pen, paper and a piano, but would be very difficult to try out. The whole concept of cutting and pasting and multiple transpositions and timescale changes is what makes sequencing wonderful. I'll give you a case in point. In my violin concerto, there's a bass line in the second movement that's very similar to the Pachelbel canon bass line. But it expands and shrinks all the time and is transposed into very unusual harmonies. I wrote that using the time scale and transposition devices from [Mark of the Unicorn's] *Performer*."

INDIVIDUAL LANGUAGE

In addition to integrating samplers and synthesizers into his instrumental classical music (see 'Key Facts: John Adams' Gear' box), Adams has also written, played and recorded two works that feature only electronic instruments: *Light Over Water* (1983, New Albion) and *Hoodoo Zephyr* (1993, Nonesuch). He calls *Light Over Water* "not as satisfying as *Hoodoo Zephyr*, because the technology was far more primitive at the time." *Hoodoo Zephyr* is a fascinating work, ranging from pulsating, multi-layered intensity, to slow, spacious, atmospheric sound effects, with much emphasis on sound processing. The harmonies are very simple, and melodies almost entirely absent. It's also a work that's clearly composed using a sequencer, with extremely tight musical performances and many repetitive patterns and structures. Adams: "*Hoodoo Zephyr* was done in my studio here, on the instruments I already mentioned to you. The only additional thing I hired was a big professional mixing desk. I love working with electronics so much that I wanted to create an album that had nothing but synthesizers and samplers."

Hoodoo Zephyr is nevertheless a rarity in Adam's oeuvre, because "I'm very interested in live performance, and in working with classically- ▶

John Adams

▶ trained players. So I suspect that most of my future work will continue to be in that format. I acknowledge that orchestras are not the easiest medium to work with, because of all the economics and the tradition that one has to deal with there. But I like working with them and with musicians, whether in the concert hall or in the recording studio. I often conduct my own music, again both in the concert hall and in the recording studio, and when I'm not conducting in the recording studio you can be sure that I'll be sitting there, choosing takes and doing the mix, and generally hovering over everything that the producer and engineer do. First recordings of one's own work are critical. It's very important that the tempi and balance and mix are completely right, because through it you should get the composer's ideal image of what he wants."

Adams was off soon after the interview for a first recording of his song cycle/opera/musical — he

still isn't sure what to call it — *I Was Looking At The Ceiling And Then I Saw The Sky* in Finland, with a collection of Finnish musicians "who were wonderful during the Helsinki performance of the piece in 1995" and American singers. *I Was Looking At The Ceiling And Then I Saw The Sky* (a drama centred around the 1994 Northridge earthquake — the title is a quote from an earthquake survivor) is expected to be released in the second half of 1997. Adams has also recently written a piece called *John's Book Of Alleged Dances* for string quartet and sampler, which will soon be recorded by the Kronos Quartet.

Despite all this activity, and despite his daring experiments, critical reaction to Adams's recent works has not been positive in his home country. His violin concerto was criticised as sounding "too European" (they have Euro-sceptics over there too, apparently), and *I Was Looking At The Ceiling And Then I Saw The Sky* was positively panned. Adams: "It was generally well received in Europe, but in the States there were people who felt that I was a classical composer dabbling in pop music, and there were politically correct protests from people who felt that I had no business venturing into black music styles. That was funny, because no black people criticised my piece, and June Jordan is an African-American herself, and she loved what I did. I pointed out that George Gershwin was a Jew from Brooklyn, who wrote *Porgy and Bess*, which was covered by every black musician from Charlie Parker to Miles Davis. So I felt that these criticisms were unfair.

"What worries me more is that I was recently criticised for no longer being interested in innovation. No composer wants to read that about himself. It's considered the kiss of death. But I think that innovation takes different guises. There have been composers who were innovative and were talking about that all the time, like Wagner, Beethoven, Stravinsky or Schonberg, and there are composers who weren't necessarily innovative in that same sense, of creating new styles of music, but who created great works of art nevertheless. Bach was seen as old-fashioned towards the end of his life, for example, even though he was deeply innovative. I think it's also a matter of the era we live in. I said that we're in a post-style period: there's no brand-new, single style that everyone is talking about and that you can become a true believer in, or opponent to. All styles and genres are blending, so all composers can do nowadays is create their own, uniquely individual language." **SOS**

KEY FACTS: JOHN ADAMS' GEAR

Adams does his composing at his home studio in Berkeley, which consists of several Yamaha keyboards, including an SY77, SY99 and Electone, Korg Wavestation, Emu Proteus 1 module and Emax II, Kurzweil K2000, Lexicon LXP15 reverb, Macintosh computer with software such as *Performer* and Blank Software's *Alchemy*, and a small 16-channel desk ("I can't remember the name. I only use this studio for writing, and I write as much with the sequencer as with pen, paper and piano").

The sequencer has been blamed in rock circles for much sterile, laboured music, and Adams agrees that they need to be treated with care: "You have to watch out that you don't make your decisions to coincide with the technology. It's better to make sure that the technology responds to you than the other way round. That takes hard work sometimes. I think that the main problem with sequencers is that they're largely used by amateurs and people with very little musical ability, to create music that may, on the surface, appear to be accomplished and polished, but is in fact very superficial. But in the hands of a really well-trained, skilled musician, a sequencer can be a very valuable tool."

At first hearing, this sounds a little like the arrogance of the classical music elite, but Adams' point appears to come out of a genuine concern that sequencers are often used to cover up musical ineptitude — which, of course, they are. And eating the proof that is in Adams's pudding, it must be said that his recent large compositions, such as the violin concerto and *El Dorado*, show the power of technology in the hands of such a skilled musician. Both are masterpieces that jump out at you from the speakers, especially *El Dorado*, with its first movement full of spooky synth sounds, tuned percussion, frantic hi-hat and snare rhythms, and haunting, chromatically-rising strings. Adams: "There's a Kurzweil K2000 and Yamaha SY99 in the recordings of both pieces. My usage of synth and sample sounds in these pieces is very subtle. I integrate them with the orchestral sounds, so

you really have to listen for them. There are a couple of synth solos, but most of the time I use synths and samplers as a colouring mechanism. I suppose you could call it a background sound. I use it like a painter who puts a colour wash on the canvas before he puts the actual paint on it. So it creates a kind of perspective against which the other instruments play. I also use samples to flesh out certain instrumental sections. For example, if I have strings playing pizzicato, I'll double that with pizzicato samples. This mixture of a sampled, processed version of the sound, and the real sound creates an interesting effect, which I like very much, and which I think is very much my identity. I spend quite a lot of time programming synthesizer sounds, for my pieces, and take samples from sample CDs, or record them myself. I will treat and process all samples here, using *Alchemy*.

"The problem I'm having at the moment is that when you write for a specific model of synthesizer, such as the Yamaha SY series, they eventually go out of production. This causes terrible problems if an orchestra somewhere in the world wants to play a piece of mine and can't find a Yamaha SY synth. I discovered that it's much easier when I have all the synth sounds in sampler format as well. They can take my sounds as a sample, and play it from that. So I'm in the process of transferring all my orchestral synthesizer parts, which were originally done for the Yamaha SY77 and SY99, to my sampler, the Kurzweil K2000. I like the Kurzweil very much, and it's compatible with the Akai S-series sampler range, so people can also play the sounds from that. But I realise that I'll eventually have to make my sounds available on all major samplers, like Emu and Roland. The whole transferral process already takes up an enormous amount of time. I've even had to hire an engineer to help me with it, because it's so labour-intensive. It also hugely frustrates my publisher, Boosey and Hawkes, because they're only used to dealing with printed music and they now have to get used to the concept of updating — so common in the world of software and

Two new big powerful MIDI merge units

When you need to drive a MIDI slave subsystem from many master devices at the same time, the new *5M* or *9M* will come to your rescue. For example, they will let the whole band play together via a single multi-timbral sound generator. They will also let a sequencer record many sources of MIDI data simultaneously and avoid the need to switch or re-cable.

5M has five MIDI inputs and *9M* has nine. Both units have one MIDI output which carries the merged datastream.

Philip Rees MIDI merge units have earned a first-class reputation. The advanced data handling features of *5M* and *9M* include intelligent interlocks on clocks and pitchbend. The firmware employs an optimum combination of the message-stream and state-machine models of the MIDI merge function. They can readily handle all types of MIDI data, including Time Code and SysEx.

These devices employ very fast, sophisticated electronic hardware. They require external dc power sources of 9-12V, which are connected via 3.5mm jacks, tip positive. The cases are 109mm x 109mm x 40mm.

5M MIDI Merge Unit £125.95
9M MIDI Merge Unit £169.95

2- and 3-input merge

If you just wish to combine two or three MIDI datastreams, you can't do better than employ our well-respected *2M* or *3M*. The *2M* merges two sources, provides two copies of the merged output and has thru sockets for each input. The *3M* merges three sources and has thru sockets for two of its inputs in addition to the merged output.

These neat units are both mains-powered.

2M MIDI Merge Unit £69.95
3M MIDI Merge Unit £99.00

Talented tape sync

You can use the *TS1* to sync your MIDI sequencer to any decent tape machine. When you start, stop or shuttle your tape back and fore, *TS1* tells your sequencer to play in time, just as if your MIDI voices were extra tracks on the tape.

The *TS1* can generate and recognise the usual four SMPTE formats. The *TS1* will convert SMPTE to MIDI Time Code (MTC). Alternatively, you can use the *TS1* by way of Song Position Pointer/SRT format.

The *TS1* merges MIDI data received with its own sync data. You won't need to swap around the MIDI wiring, as *TS1* has four MIDI ports and automatic signal routing.

The *TS1* has a built-in mains power supply.

TS1 MIDI Tape Sync Unit £99.00

Amazing MIDI to CV

For an amazingly low price, the *Little MCV* lets your MIDI system control your analogue synths with their great sounds and friendly knobs.

This versatile interface unit can generate control voltages for the 'one volt per octave' (logarithmic) or the so-called 'volts per hertz' (linear) systems.

The gate output can be set to five volts positive, ten volts positive or S-trig.
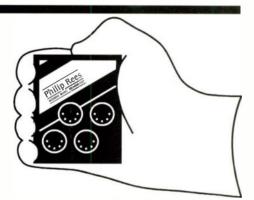
High resolution sixteen-bit conversion allows accurate pitch across the full 128 note MIDI range with smooth modulation, pitchbend and portamento. The CV output also has a wide bipolar voltage swing and a tuning preset is provided.

The MIDI sustain commands are comprehensively implemented. MIDI reception can be set to any channel, using the straightforward front panel rotary control.

There are MIDI IN, CV OUT and GATE OUT sockets. The mains power supply is built-in.

Little MCV MIDI to CV Converter.. £75.95

Functional Simplicity

Select a selector

These handy low cost switch-in-a-box gadgets solve many MIDI routing problems and avoid the inconvenience of recabling.
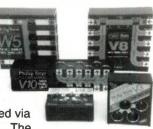
The *3B* is a novel changeover switch, which will let you bypass your computer or sequencer without moving cables.

2S MIDI Selector £11.95
5S MIDI Selector £29.95
3B MIDI Selector £29.95
9S MIDI Selector £39.95

Low cost thru units

Some MIDI gear may lack thru sockets. Chains of more than three MIDI devices can suffer from data corruption. You can solve these problems at low cost with *Philip Rees'* MIDI thru units.

The *V3* is a battery powered 1-into-3 thru box. The *V4* has four outputs and is powered via its MIDI input. The *V8*, which has two inputs and eight outputs, requires an external power source. The *V10* is a mains-powered 1-into-10 unit. The mains-powered *W5* has independent source selection for each of its 5 outputs.

V3 MIDI Thru Unit £11.95
V4 MIDI Thru Unit £19.95
V8 MIDI Thru Unit £27.95
V10 MIDI Thru Unit £39.95
W5 Dual Input Thru Unit £55.95

Californian company Zeta have made a significant contribution to the profile of the electric violin, with a range of instruments and hardware which have opened the world of sound synthesis to the string player. Since the first Zeta synth violin appeared in 1984, the company has grown enormously and has a string of world-class musicians as advocates, including 'fusion' violinist Jean-Luc Ponty.

ZETA MIDI VIOLIN

The Jazz model violin was the first synth design to appear from Zeta, and is available in both 4-and 5-string versions (CGDAE). This handsome instrument is constructed from lightweight bass wood, and

MIDI Out with a snug locking facility for the cable, and a volume knob. The volume knob acts as a mix control for the analogue signal and MIDI, but only controls the volume of the former.

The timbre of this violin is simply magnificent. When I became interested in electric fiddles, I doubted whether their sound quality could ever threaten acoustic territory, but no company has worked harder than Zeta to make this possible. I found the Jazz model's creamy tone a dream — even the open strings — and I played it for a whole day without even trying it with MIDI.

At the heart of the sound is Zeta's bridge system, which uses two piezo pickups per string and can be raised or lowered according to taste. Each string has its own preamp and trim pot for sound level control. The preamps are battery powered, but when using MIDI, remote power is supplied by the Synthony. If you happen to find a better pickup system than this, Zeta promise a refund! (See 'The Science Behind the Strings' box for more technical detail on the Zeta system.)

Space Age Strings

ZETA JAZZ MIDI VIOLIN & SYNTHONY MIDI CONTROLLER

For over a decade, Californian company Zeta have kept the flag flying for string players who'd like to bring their skills into the synth age.
ALAN McCLURE takes a bow...

although it's a little heavier than its acoustic brethren, its weight is conveniently concentrated on the player's shoulder for maximum comfort. Its most striking physical feature is the absence of traditional outward-curving shoulders; the Jazz has more of a double-bass design, with shoulders much lower on the neck. This allows easy access to high positions, especially on the lower strings, and avoids strain on the left wrist and elbow, but requires that your intonation is very good and that you don't rely too much on visual communication or physical contact with a shoulder for familiarity. Zeta have included a removable upper bout strut (or traditional shoulder) for those players who must have a reference point for their hand when shifting into higher positions. The body itself has been thinned, allowing the player to look down a little more on their fingers. Initially, I found the headstock heel and the neck heel a little offputting, as they have a less extreme angle than normal, but in general I have found playing very comfortable.

The violin uses unique ebony hex wrench tuners which point inwards and upwards, keeping the instrument in tune for long periods of time (in fact, I've only used these when changing strings). Normal daily tuning is handled at the tailpiece.

Electronic control functions include a line out for mono analogue output of the violin sound, a

SYNTHONY MIDI CONTROLLER

To get the MIDI violin system up and running, you'll need several things. In addition to the MIDI violin (or Zeta Retropak — a bridge and tailpiece MIDI pickup for acoustic violins), you'll need a Zeta MIDI control unit. Zeta manufacture two MIDI controllers: the VC225 and the more recent Synthony, which has built-in sounds. The VC225 is identical to the Synthony, except that it doesn't contain sounds, and would be a better buy for a musician who already possesses a good GM synth.

The Synthony has an internal GM synth, and is extremely easy to operate. Each of its 12 command parameters, with corresponding LEDs, is clearly printed on the front panel. I'm particularly fond of the Dynamics Mode parameter, which transmits the relative loudness of each string to the synth, in such a way that the player can control aftertouch or breath control from the bow for expressive sounds, or the initial attack volume for percussive or keyboard sounds. The Dynamics Scale command determines how much of a change in volume is controlled from the bow.

The front-panel edit button toggles from one preset to the next and also acts as an 'enter' key, for saving and deleting presets. The display is a simple 2-digit affair, and a there's a

useful built-in tuner. Also provided with the Synthony is the MS40 footswitch, which controls three types of hold, sustain, bypass, and a chain/step system for saving, deleting and changing presets quickly. On the back panel, there's a MIDI Out for driving other synths, and on the front panel there's a headphone input and a synth In/Out button for bypassing the internal GM synth.

The sounds on the internal synth are pretty basic and cannot be edited, apart from transposition. A serious musician would probably be better off using the Synthony or the older VC225 to drive another synth — to my ear, most of the Synthony sounds are cold and lifeless, especially drums and percussion. Authentic orchestral timbres are badly lacking, and strings are very 'synthy'. A handful of sounds — fretless bass, marimba, glockenspiel, xylophone, and the 'Warm Pad' patch — provide comfort, but in my opinion, a good external synth is essential for best results from this equipment. (Be warned: if you have an older MIDI synth that will only receive on one MIDI channel, and you have pitch bend enabled, pitch-bend messages will only be recognised properly if only one note at a time is being played. So when double-stopping, both notes will be equally affected by pitch variation in either one of them.)

I confess to being a bit cautious when I connected the Synthony to my Kurzweil 2500. I knew that tracking speed, which you'd expect to be good with the Synthony's own sounds, might have been considerably reduced with an external instrument. Not so! The results were breathtaking. Tracking was instant and accurate, and this was where the real fun began. Although each program required editing for best results (chiefly attack, decay, sustain and release) and this was a little time-consuming, I found using first-class sounds through a violin an incredible incentive for imaginative music-making. I discovered that it was best to leave pitch-bend at '0' wherever possible, especially with sounds that take a little longer to reach their full amplitude, because at increased playing speeds the synthesizer doesn't track and causes minute flaws in intonation.

To backtrack slightly, setting the unit up includes system learning, which is 'teaching' the controller your bowing style with the volume turned down. A few other twiddles are all that's needed to match up the Synthony to your playing style.

HOW DOES IT FEEL?

As long as your playing is clean and accurate, and not too over the top, the Synthony MIDI Controller

is as stable and reliable a unit for violin as you're likely to find. Tracking speed is 98% delay-free, losing the other 2% on the bottom C string (5-string instruments), so I'm pretty impressed. Most bowing styles can be faithfully reproduced, as long as the weight of a bow stroke is not too superficial, and the fingers of the left hand firmly stop the strings. You do have to be careful setting up a good threshold sensitivity, because the Synthony sometimes refuses to recognise open strings when they are played in the middle of a series of stopped notes using slurs, especially at high speeds. For best results, re-bow open strings. Slurs work best in fast passages and detaché is best kept in the upper half of the bow. Certain sounds will even work well with tremolo bowing. A controlled spiccato is successful with certain sounds of short duration, especially percussion, as long as each bow bounce is firm enough to sufficiently bring the string up to the correct pitch.
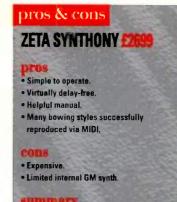
If you think that your intonation is pretty good, working with MIDI, with pitch bend at 0 and using no vibrato, may come as a shock. Notes will easily 'jostle' around a semitone if they are not 100% accurate, so it may take a while to build up your confidence. Intonation is more difficult on an electric stringed instrument because there are fewer overtones present than with an acoustic one, so be patient and results will be rewarding.

For more expressive sounds the violin is ideal, being a fretless instrument; bending notes is therefore a more natural approach than using a pitch-bend wheel.

CONCLUSION

The Zeta Jazz violin is a superb instrument, and the MIDI system should promise reasonable, reliable results in the hands of a good player (if he/she gets enough session jobs to afford it). Electric bowed strings are definitely gaining more and more respect and good players who have a system like this should benefit immensely. SOS

£ JV205 Jazz violin (5-string) £2379; JV204 (4-string) £2149; Synthony MIDI controller £2699; VC225 MIDI controller £2149. Violin Retropaks: 4-string £889; 5-string £969. Prices include VAT.
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The Direct Approach

ALL ABOUT DIRECT INJECTION

Most people have heard of direct injection, or DI, but what is it, how does it work, and what can it do for your recordings? HUGH ROBJOHNS gets directly to the point...

There are only two ways of capturing the acoustic sound of an instrument for recording or PA purposes. The most common is obviously by using a microphone, and plenty has already been written on that subject over the years (some of it by myself!). The other way is by using a method known as Direct Injection — or DI'ing — which relies on some form of mechanical or electrical pickup mechanism on the instrument itself. Incidentally, the term 'DI' is also used in the context of connecting electronic keyboard outputs or power amplifier feeds to a mixer — rather different requirements to the true DI, but important all the same.

The most important thing to remember is that the process of DI'ing an instrument is not a problem-free means of capturing its sound. DI has both advantages and disadvantages compared with a microphone, and the pros and cons vary depending on the particular situation. If you will excuse the metaphors, think of the DI as just another weapon in your armoury to be used in the war of getting a good sound.

THE BASIC CONCEPT

When you say 'Direct Injection' to most musicians, they probably think of a small black box into which they plug their instrument. The box is part of the DI, but the electro-acoustic pickup on the instrument itself is equally (if not more) important. The quality of the sound you get from DI'ing is the result of the combination of pickup and DI box, and although you rarely have control over the pickup, choosing the right DI box can often make an enormous difference to the end result.

Let's start by thinking about the pickup itself. There are various types in common use, the most common being the straightforward electro-magnetic pickups used in electric guitars. This type uses a coil of wire to sense any changes in the magnetic field created by a small permanent magnet. As the guitar string above the coil vibrates, it disturbs the magnetic field, and the coil generates a small electrical current, which is passed on to an amplifier and loudspeaker. Another common form of pickup is the piezo device, often fitted into the bridges of acoustic guitars, where the mechanical vibrations in the bridge cause microscopic distortions to the shape of a piezo-electric crystal, generating a small voltage in the process.

There is also a wide variety of more specialised forms of contact mics, or bugs, designed for stringed and wind instruments, which work along similar lines to the piezo pickup. In general, however, all types of pickup have unbalanced outputs at mic level, and are relatively weedy in terms of their ability to drive long cables and low impedances (see the 'Importance Of Impedance' box).

So, what does a DI box have to do, and what is inside one? Well, the idea of the box is to provide ▶

EMO produce a range of passive DI boxes like this one here, which is available in single- and dual-channel versions, and has a ground lift switch to minimise earth loop-related hum. The company also produce a 19-inch 8-channel rackmounting DI box (not pictured).

BSS's fully-featured AR116 active DI box offers a low-pass filter, a choice of three types of pad (for use with guitar pickups, line-level inputs, and power amp speaker outputs), a ground lift switch to help prevent earth loops, and an XLR output.

All About Direct Injection

▶ a convenient means of transferring the signal from the instrument pickup to your mixer as cleanly as possible. As the output from most types of pickup is small (-50dBu) and unbalanced, the DI box is needed to balance the signal (normally with a transformer) and send it down a microphone cable to the mixing desk, at a nominal mic level.

Most DI boxes also provide inputs which can accommodate higher-level signals, such as those from electronic keyboards (0dBu, or line level) or even power

MTR's active DI box is available in mono and stereo versions (both shown here), but they sport the same features: a high input impedance to prevent signal coloration, two pad settings, and a ground lift switch.

amplifiers (possibly +40dBu), and also offer a 'loop-through' facility so that the signal can be passed on — usually to the musician's own back-line amplifier.

PASSIVE OR ACTIVE?

The internal workings of a DI box vary depending on what sort of box it is — passive or active. Passive boxes do not have any form of electronic amplification built in, and use only passive components — normally a couple of resistors, a switch or two and a transformer which feeds a balanced mic-level signal to the mixer. The pickup is expected to drive all of these components sufficiently well to produce the required output, something they are not generally designed to do.

The resulting 'loading effect' is the main disadvantage of passive DI boxes — and as a result of this, such boxes often have a very dramatic effect, not only on an instrument's sound quality, but also on its playability — a point I'll come back to elsewhere (again, see the box on impedance below). However, the main advantages of passive DI boxes are that they do not need batteries or phantom power from your mixer to work, and they are significantly cheaper than active boxes.

Active DI boxes are essentially very similar to the passive boxes — they are intended to do the same job, after all! The big difference is that they incorporate a buffering amplifier which effectively isolates the instrument's pickup from the rest of the DI technology (note — the internal amplifier is only a buffer. It has no significant gain, and the DI box output signal is still at mic level rather than line level). This kind of DI box is far less likely to affect the sound or playability of the instruments you plug into it, and (in my experience) usually gives trouble-free results. The downside, of course, is the need for batteries or some other means of powering the internal amp.

SHOCKING PROBLEMS

Connecting your mixer directly to some unknown piece of equipment (especially keyboards and amplifiers) is potentially dangerous. Faulty wiring within the instruments or equipment could put hazardous voltages on the DI signal going to the mixer with disastrous — and possibly fatal — results!
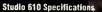
The most important advice here is that if something looks even slightly dodgy, don't plug it into your DI equipment — it is definitely *not* worth the risk. Things to look for in particular are earth wires lifted from the mains plug, damaged mains leads, and six-inch nails where the fuse used to be! Even if the musician says everything is fine, he might always wear rubber-soled shoes and be completely oblivious to the fact that his hair stands on end when he plays the guitar! Joking aside, an awful lot of expensive gear has been destroyed ▶

THE IMPORTANCE OF IMPEDANCE

Elsewhere in this article, I have mentioned how DI boxes — particularly passive ones — affect the sound quality and playability of some instruments. This is all down to the impedance, which is basically the loading placed on the pickup by the DI box (also referred to as the input resistance).

In simple terms, the pickup has to generate a current which changes in direct proportion to the original acoustic signal. The lower the impedance of the device the pickup is feeding, the bigger that current has to be, and therefore the higher the strain on the pickup.

Most pickups are designed to produce very small currents (or small voltages, which amounts to the same thing here), so they need a very high-impedance load from the amplifier or DI box. If the

pickup sees only a relatively low impedance, the effect on an electric guitar (to take one example) is that the sound often becomes rather dull and muddled. The impedance also affects the way notes can be sustained, and it is common for the instrument to feel dead — the pickups are unable to pass on the decaying string vibrations, because they simply can't generate enough current.

In a passive DI box, there are very few components — the major one is the transformer, which is used to create a balanced output signal destined for the mixer. The windings of the transformer are only long lengths of copper wire, so it is hard to create a high resistance. The better passive DI boxes usually manage something like 100KΩ. This might sound pretty high — and indeed it

is in the world of transistors and op-amps — but it isn't in terms of valve amplifiers. Most guitar pickups, being designed for the classic valve amplifiers of yesteryear, need to see something substantially higher than this if they are to give of their best.

Active DI boxes use a buffer amplifier which is designed to present a very high impedance to the pickup, but with enough output current to drive the balancing transformer properly. One of my favourite active DI boxes (originally made by Theatre Projects, but now manufactured and badged by Canford Audio, and available from their catalogue — see the separate 'Where To Find Them' box), manages an input impedance of 10MΩ — a hundred times higher than the best passive boxes — and this seems capable of preserving all the desirable characteristics of every type of pickup.

Sample shop

Our regular look at the hottest new sample CDs and CD-ROMs.

ROCK HISTORY DRUM LOOPS VOLUME 1
(AUDIO CD)

This CD from Black Box Music presents a collection of drum loops primarily aimed at the rock market and ostensibly covering drumming styles from the '70s to the present. Played by Heinrich Altenbroxter and recorded by Arthur Koll, the CD is divided into three 'chapters': Groove Sets, Rock Drum Tool Sets and Single Hits.

The 20 Groove Sets are designed to emulate the drum backing from some famous '70s rock songs. Unfortunately, the CD doesn't attempt to stray very far from Led Zeppelin territory — indeed, it doesn't stray at all! The good news is that the results are effective, with a solid choice of basic sounds and effects to take you right back to the days of cheesecloth shirts and unwashed flares. Tracks are named in keeping with their source — 'Immigrant' and 'Kashmir', for example — but those less familiar with these songs might welcome more descriptive sleeve notes. Each Groove Set features between 12 and 30 loops, including fills and variations.

Moving onto chapter B, the Rock Drum Tool Sets each contain three tracks with loops in dry, wet and hi-hat/cymbal-only versions. Within the dry and wet versions, each of the two basic style loops is provided in loud, medium and softly-played variants, before being presented again with ride or china/crash variants. A trio of fills and a final 'special' loop round off the wet and dry

versions. Each set is named although, again, how useful these names are is largely down to your rock history knowledge — a brief description would certainly have helped me! All of the patterns, with the exception of one 6/8 offering, are in 4/4 time.

The CD rounds off with the Single Hits chapter. I found some of these difficult to place with the looped patterns. The sleeve notes give no help at all here, failing to detail even how many samples are present on each track. What the documentation does suggest is that the single hits can be used to feed an effects processor to give the illusion of adding effects to specific sounds within a loop. This is a trick I have often used myself, but to program all of the single hits for a number of live-played loops would be quite a tedious process, I'd say. A set of MIDI files derived from the loops would be extremely useful in this context, especially if the documentation gave suggestions as to which single hits to use for each loop.

If you are regularly involved in producing rock music, or want to add some punchy rock rhythm loops to your own style of music, there are currently very few alternatives to this collection. For that reason alone, the *Rock History* CD would deserve praise, but the samples are also capable of producing convincing, powerful drum tracks. In use, I found that the sounds sat comfortably in a mix, to the point where few listeners would discern them as looped material at all — which is probably more important in rock music than the dance genre, which often attempts to flaunt the use of looping. I might have hoped for some forays into more diverse playing styles, and a few 7/8 or 5/4 time signatures would have been more than welcome in these stuck-in-a-rut 4/4 days, but the production and playing are generally excellent. The only problems lie with the slightly under-cooked documentation (the useful test tone at the end of the CD doesn't even get a mention!) and the daunting task of sampling and arranging so much audio material. With the latter in mind I'm keen to see how the CD-ROM version measures up. *Paul Ward*

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THE DEFINITIVE PERCUSSION SAMPLER VOL 1: STEVE REID
(AUDIO CD)

Just when you thought the world had seen enough drum and percussion sample CDs to last an eternity, Voice Crystal have added to their ranks with the boldly titled *Definitive Percussion Sampler Vol 1* (the title presumably encouraging us to keep our eyes peeled for Volume 2, out shortly and no doubt even more definitive). Titular ambiguities aside, this 78-track CD provides a massive selection of contemporary and ethnic percussion samples, presented not in the familiar loop and sample format, but as single instrument hits. The disc is divided into a number of sensible sections, starting with Gongs, Waterphones, Bells and Chimes, and moving through the whole gamut of percussive instruments from around the world, before ending with a highly usable collection of metallic effects, flute and woodblock sounds.

The excellent sleeve notes guide us through the many variations of each instrument — for example, track 11 (Congas), has 22 different recordings of single conga hits, ranging from slapped to muted and played with finger tones, palm tones, and even played using a stick! This fantastic attention to detail is continued throughout and you'll be impressed not only by the range of exotic instruments provided, but by the vast sonic changes that can be coaxed from even the simplest of drums. Frame drums, Shakers, Udus, Utars, Rainsticks, Guiros, Washboards, and Djembe drums are amongst some of the more unusual inclusions, and as a reliable source for these and other sounds, this CD is well worth the price. Interestingly, at the tail end of the CD there's a good selection of non-percussive sounds such as Slide Whistles, Clay Flutes, Jews Harps and Whistles. All of these are given the 'if it moves, sample it' treatment, and are presented in long and short versions with a number of other variations thrown in. The recording quality is good throughout, and all the sounds (both stereo and mono) are presented dry and without any unnecessary compression.

On the negative side, it's getting more and more difficult to distinguish between releases of this kind. Yes, it is very good indeed, and I can't imagine anyone wanting a percussion sound

that isn't included here, which makes it a stunning achievement, but at the same time there isn't much here that we haven't all heard before. The forthcoming CD-ROM version (for *SampleCell*, Roland, Ensoniq, Kurzweil, Akai and Peavey formats) will undoubtedly prove to be extremely popular, as it will cut down the time needed to sample, edit and present these sounds and will give a huge number of users a massive library of ethnic and standard percussion sounds ready to go. Until then, however, there is still a great deal here that will both inspire and delight you (particularly in the terrific effects sections), but not, unfortunately, much that will bowl you over with originality. *Paul Farrer*
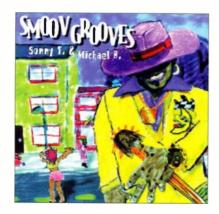
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SMOOV GROOVES
(AUDIO CD)

Smoov Grooves, from the acclaimed producers of *Funky-Ass Loops*, is a 74-minute audio CD packed with slow funk musical loops and grooves.

For those of you unfamiliar with this type of sample CD, the basic idea is this: a couple of extremely funky guys (namely Sonny T and Michael B, otherwise known as the rhythm section behind the artist formerly known as a member of the royal family, currently without a name and known only by squiggle) have put their heads together and recorded a selection of

short 'songs' lasting about eight bars each. These consist of drums, bass, guitars and keyboard parts, and are presented first in their entirety, then split up into their constituent solo lines. This seems to be quite a popular way of presenting sample CDs these days. Many producers have adopted the same approach, and although you could argue that all you're doing is reconstructing a finished piece of music, it does give you a good idea of the kind of results you can achieve by mixing and matching the various musical parts throughout the whole release.

This 62-track CD doesn't bother with any demos or clumsy spoken introductions: it just kicks off with the first of the loops — and what

a great way to start. Slow and meaningfully funky vibes seem to pour out of every note and, despite being more laid-back than an arthritic tortoise on Mogadon, the timings, as you might expect, are absolutely spot-on. The loops range from 60 to 108bpm, and vary from a sort of camp '70s cop show feel right up to the familiar spaced-out Jamiroquai-esque MTV style of the '90s. The recording quality is as good as the excellent playing, and the producers have wisely left most of the effects out of the mix altogether.

There are tons of fabulous bass licks, as well as a good number of funky strumming and picked guitar parts too. The drum and percussion parts are deconstructed as much as they need to be, yet the producers haven't filled up the CD with countless versions of the individual drum samples, and the warm and resonant Rhodes chords will be filling dance floors and TV ads from now until next Christmas.

With these guys as his rhythm section Prince (or squiggle) can justifiably sing "My name is Prince, and I am Fon-Keh", and with this CD in your collection you can now say the same thing. So turn the screen contrast of your sampler down low and immerse yourself in the understated glory of slow-burning precision funk. *Paul Farrer*
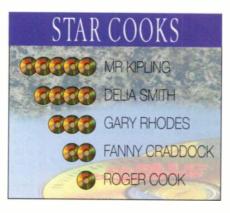
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- F **01273 328881.**

A POKE IN THE EAR WITH A SHARP STICK
(AUDIO CD)

At last — someone has realised that if you're going to have a digital max tone on a sample CD it should be at the beginning!

East West have a history of supporting industrial music. On this release, they've largely omitted the music! First we're treated to a large selection of metal sounds: hits, rolls and swells. Names like 'Alien Environments', 'Cyberpunk' and 'Kaos' don't tell you much. Think instead of oil tanks and saucepans being variously banged, scraped, stroked and dropped down lift shafts, cars being crushed, and so on. Samples are good,

and sonic content evolves incrementally. After a while more variety creeps in. Effects are variously synth-generated and found/transformed; the vibe is dirty and hard while retaining fidelity. Dirty and clean? I'm sorry if you find it difficult to hold two opposing views simultaneously — that's just how it sounds to me.

There are drones, chain runs, ethnic tones and atmospheres, spacey textures, a water section encompassing drip and drain, and a minute traffic noise bleed. There are big stereo bongs, metal swishes and bangs which could emanate from a plate, gong or oil drum. One sample has a reversed tail which stops abruptly. Synth and other effects are hard and dramatic. I heard electrical arcs and shorts, machine grinds, transformed vox and industrial textures and hits with digital feedback. I particularly liked the ball spinning in a bowl. There are samples of dry thunder, glass, eerie discordant tones and drones, piercing digital buzzes and feedback. However, wherever you drop into this CD, you're never too far away from it's stock-in-trade — metal noises.

Samples are generally mixed up, but occasionally strictly grouped, with the 'Percussive' section being more coherent than most. Then again, it's also less interesting, comprising single hits, including (yep) metal implements, along with pretty standard drums. There are loops too, but not yer standard drums: instead, they're well-produced and cutting ethno-electro percussion and effects. Last up come ethnic wails, followed by 'Oakland Jail Ambience'. One of our sound designers goes to jail?

This mixture of metal and out-of-control noises represents a narrow but useful repertoire. Applications? Well, the atmospheres are not really long enough for film use without looping. On the other hand, the sheer drama of many of the hits and tones would lend a helping hand to many a TV commercial and particularly, perhaps, the increasingly sonically-demanding computer games market. And I'll be dipping into this collection regularly for remix excitement. Nice one. *Wilf Smarties*

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demo DOCTOR

AURORA

Recording Venue: Home
Recording Equipment: Yamaha MT3X cassette multitracker, Yamaha SPX90 effects, Shure SM58 mic.

Aurora are a Torquay-based band going for a mellow dance vibe. The opening track is just working out fine, with a lightweight vocal from Kelly Brader floating across the drum- and percussion-oriented mix courtesy of plenty of reverb. Unfortunately the arrival of the male vocal breaks the atmospheric quality of the song: the sound is just too hard. I don't really think that these vocals gel together anyway — one soft and the other harsh — but when Kelly sings out the harsh quality appears in her voice too. A compressor would probably be a wise investment to control the levels, but until such time as you can afford one, try keeping an eye on the meters when you record the vocal and make sure that they hover around the same level, give or take 4dB or so. Brian could try singing more softly, and also closer to the microphone, to achieve a mellower
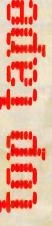
sound that will be more in sympathy with Kelly.

Vocal problems aside, the general sound of the mix is quite pleasing and probably good on a variety of speakers. During the instrumental sections one track in particular, 'Jealousy', is the perfect late-night mix, with nothing harsh jumping out of the mids to ruin the flavour of the recording. Such a shame that the vocal recording has to upset this mood, but a useful reminder of just how important it is to make vocals and backings work together — the vocal sound is so often an afterthought on demos! ❑

STEVEN COLLAZO

Recording Venue: Home
Recording Equipment: Alesis ADAT digital 8-track, Soundcraft Spirit 24:8:2 mixer, Atari 520ST running C-Lab *Creator*, C-Lab Unitor sync/output expander, PPS2 sync, Alesis 3630 compressor, Quadraverb 2 effects, Behringer SNR2000 denoiser and EX2100 enhancer, Sony DTC690 DAT and TC WR565 Cassette deck, Linear Phase 8812 studio monitors, Brand X amp, AKG C1000 mic.

Steven's basically trying 'to put across a kind of R&B with dance elements'. Immediately I was impressed with both the lead- and the backing-vocal performances — all by the man himself. His singing style is definitely American dance/soul, as you can hear from the mellow approach, choice of harmonies and control of vibrato. He obviously has a large vocal range, too: from the high harmonies on the choruses to the low vocal that underpins the lead lines. It's interesting that the microphone he's using, an AKG C1000, is excellent for the backing vocals but not quite as successful for the lead lines. This is because the coloration on the mic tends

to harden the mids, which means that you can place the backing vocals lower in the mix without losing them (I should point out that the mellow backings also conspire to allow this to happen). The lead vocals, in contrast, would benefit from a little more lower mid at around 500Hz and some cut in the classic 1kHz region, to just give the sound that extra touch of class it deserves.

Having said that, I'm not sure that it's really worth remixing, because the vocal performances on all the tracks is so good that I wouldn't have any hesitation in playing the demo to recording or publishing companies. My only criticism would be that, for this country, the backings could be considered pretty mainstream and it might be worth collaborating with a good hip-hop or garage programmer to take what Steven has got and play around with it. Given that he's used an ADAT with an Atari synchronised, there are no end of options available — anybody else just needs a copy of the tape with the vocals on it and some synchronisation details.

Even so, Steven's own mixing skills should be mentioned. Everything is well balanced — not easy with so many backing vocals to deal with — and there are some nice effects touches, but the effects are not over-used on the lead voice, allowing it to front the mix. A good demo. ❑

Recording Venue: Home
Recording Equipment: Apple Mac running Emagic *Logic*, Emu Proteus FX sound module.

Some of the Croonisms drawn from the accompanying letter make sense, such as "Technology's great in allowing you to do things more easily than before, but no amount of it is a substitute for musical ideas". Yet after the second musical truism the didactic stance begins to wear thin and you start to wonder whether Mr Croon does actually have "fun with music" as he states in his letter.

The teasing nature of the first composition — 'Nowhere to Come but Back' — suggests that he does. I particularly liked the use of what sounds like a marimba voice from the Proteus sound module to add attack to a piece that featured synthesized voice, wind and string instruments, too. Using only a single module and recording direct to cassette is a process

that requires some discipline, but here the combination of E-Magic's *Logic* software and the Proteus FX enables the MIDI side of things, at least, to be easily accomplished.

Despite the 'unfashionable' (his words) nature of some of the music — a demanding listen by any standards — the quality of the recording, even when copied from a cassette master, is high. Many of the sounds are very realistic, particularly the pizzicato strings, piano and marimba: a testament not only to the sound module, but also to the way the sounds are arranged. It's the string sounds that end up seeming the most unnatural, especially when they take the melody part. Even so, I suspect that Mr Croon has deliberately used the limitations of the Proteus to his advantage, especially on some of the more testing compositions where tension is built into the theme — for example, in 'Stranded Concerto

Music' the strings begin to thin out and grate on the higher notes and the attack is a little odd too.

This is a tape that is strong in compositional quality and worthy of a listen by anyone who would like to compose contemporary classical music or to rearrange older pieces using synthesized sounds. Croon's 'Mazurka' and 'Madrigal Remould' balance out the more avant-garde pieces, while the short 'Borrowed Chords' betrays the jazzier side of his nature. ⌐

Recording Venue: Live gig
Recording Equipment: Live recording — Spirit RacPac mixing desk, Alesis ADAT digital 8-track, 3 x AKG C1000 mics, unnamed hi-fi condenser mic. Mixing — DAT, LA Audio comp/lim, Alesis Quadraverb Plus effects.

Simon Jones is the man who recorded this live jazz gig by Blakey's Special Six, although he had some unforeseen circumstances to cope with. Well, isn't that always the way

with live recordings? If he wants to continue with a career in live recording he'll have to expect the unexpected! It's no surprise to me that there was no chance for a soundcheck, even if the circumstances (the band delayed by a snowstorm) were quite unusual for this country. Such things can be overcome by experience in the live gig situation — for example, getting to know the sort of level you'd expect from a jazz band when you have three open mics in the room, and working quickly with your input gain sections on the desk; as you get more used to their tolerances you will be able to set them up for what you think the level is going to be.

Regarding the levels to tape, the ADAT is quite forgiving if you under-record, but the opposite is true if you peak into the red for any amount of time, so it's not a bad idea to set your limiter up at home to make sure that levels at the gig don't get too hot for the ADAT. Making a note of

the settings, or putting on a little chinagraph mark, will take that worry off your mind in a pushed situation.

Listening to the recording shows up a few slightly heavy-handed fader movements — I noticed the bass on the first track, at least — although this is something only the experienced engineer would be listening for, and in general the sound is an excellent reproduction of the vibe of the gig. I have a few suggestions for improving the overall sound which Simon could try out if he has the time.

First, he might experiment with more compression. This could actually be done on the mix he has already and, if set up properly, would not compromise the sound but would even up any little problems in the balance. Secondly, a slight mid cut at 600Hz takes some of the hardness out of the room ambience and a 10kHz boost of 3dB helps with the presence. Both can be achieved using the EQ on the Spirit Racpac desk. ⌐

DARREN WHITE: Darren is looking for help in getting his songs to established artists in the hope that they may be covered. Quite honestly, the best way to do this is by sending tapes to publishing companies or the management of artists you think the songs might suit. I also find it interesting that Darren attempts to justify the 'ordinary standard' vocals on the tape. Remember that the vocals are the most important things in a song; if you believe strongly enough in the song and you can't manage the singing, it's worth paying someone else to do the job — try any of the bigger local studios and I'm sure they'd be able to put you in touch with someone. If this means that it takes longer, then you can still write songs while you're saving for the small fee that most struggling vocalists would require.

DUKUR: With mixes that are geared for underground clubland, Sheun Olatunbosun tends to start with the dance groove and add the rest later. His first track on this tape is well under control, with a pounding four on the kick, echoed cabasa and keyboard, and minimal conga loop. Add to this a vocal thinned by the use of a phase, some sampled guitar and a burbling bass, and you've got the picture. Yet it's all very understated and I can't help but think that suddenly it's going to get off the leash and give the listener a good kick in the face. Still, I guess that must be my rock background talking. Moving on to the sampling, every dance demo I've heard done with the Amiga computer and 8-bit sampler has this wonderful grainy low-tech sound, but all the owners find them limiting. Personally, I think low-bandwidth sampling will always have a place in the mix and Sheun's mixes are a case in point. Good use of effects too — phase and echo being the obvious ones for underground.
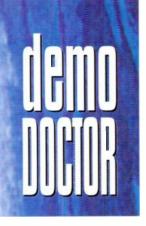
demo DOCTOR

CONVULSANT: David Jury is the programmer behind the name and he's certainly progressed since his last tape. Drum-beats hang together

better, although he still doesn't go for many breaks in the drum department, preferring to use looped strings, echoed burbling (excellent!), and low synth instead on the first track 'Banana Scoop'. Moving into jungle territory, the second track, 'Mugwump', is more adventurous. A neat kick-drum and note combination starts off a slightly dub style, but then you're treated to some nice snare breaks, conga loops and, of all things, a double-bass break!

JUPITER: This demo showcases guitar-based pop from Walthamstow-based Jupiter. The opening track has well-programmed drums and bass, with only guitar and vocals actually being played. The only time this doesn't seem to work is when the hi-hats erupt into broken 16th notes with very little in the way of accenting to make them more natural. I also found the overall sound a bit woolly apart from the guitar, which was DI'd via a Quadraverb GT that made it

sound as though the guitar had little body and was all upper-mid frequencies. This may have been caused by using too much of the direct sound as opposed to the preamp of the GT in the mix — it's easily done! Furthermore I'd try experimenting with the EQ on the GT. The graphic is a good place to start as it's so visual, but some boost around 200Hz is what you really need to give this kind of clean, pop guitar sound enough guts to kick the track along.

THE SCAM: Recalling the punk of my teenage days, I think that the Scam have something of the Buzzcocks about them. Some neat twists and turns in arrangements, sudden stops and key changes, plenty of energy, petulant vocals, non-lead breaks — yes, it's all there. Yet there's also something quite American in the feel; the Yanks have gone streets ahead of us with garage bands such as Presidents of the USA. The quality of the cassette sent by the Scam doesn't quite match the playing because of the drop-outs and distortion; however, through it all you can tell that the band are pretty sound and that they record OK. The guitar sounds are beefy and panned wide in the stereo for maximum width with minimal instrumentation. I also liked the 'lead' break on the second number, which had a great miked-up resonant honk to it. Good vocals too, and catchy choruses with harmonies — certainly a band worth catching if they're ever gigging in your neck of the woods.

WHEELWRIGHT'S BANE: Wheelwright's Bane are a folk/barn-dance/ceilidh band based in the Reading area who have been in existence since the early '80s. Having gone through the usual amount of personnel changes, they actually had to finish this tape off before one of the band left to take up a job in India... Even upgrading from a Tascam TSR8 to a Tascam DA88 (good choice) during the course of the recording doesn't seem to have upset the continuity. The recordings are of a consistently high quality with plenty of life in the playing and some interesting additions to the arrangements — mostly from keyboards and electric
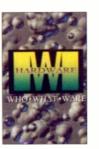
guitar. For example, Dave Fenner's electric guitar lead break on the opening 'Ceud Mile Failte' owes a lot to Mike Oldfield in terms of sound but has a more down-to-earth feel about it. The addition of a keyboard brass section to the tune set 'Bridge of Lodi/Humours of California' — complete with tongue-in-cheek big-band ending— shows that not all folk players are as po-faced about the tradition as some would have you believe.

A drum kit is also used on most of the tracks — although the full kit isn't always played: take the interesting 'Chill Out Polka' which uses the kick drum and the snare with the snares turned off for the back beat. Add a dub bass line, some classic low whistle, electric guitar, pipes and sax and we're almost in Moving Hearts territory. The kick and snare here would have benefited from some room reverb, though; they're just a little dry for my taste. On some of the other tracks I notice a little reluctance to mix the snare loudly enough — if the band compare the ballad 'A Strange Affair' with the track that follows it they should see what I mean. But apart from some small mixing idiosyncrasies the overall feel of the tape is good and there are excellent performances from all. I have to single out the excellent singing of Sylvia Barnes and the low whistle-playing (a devil of an instrument to record and play well) of producer and engineer Rex Morrey.

HARDWARE: There's a problem with thin guitar on this demo and it's probably down to the way in which the processor has been used. This time the Quadraverb GT program is skanky guitar and slapback echo. For such an up-tempo track the delay time is too long and

the mix of wet signal against dry biased in favour of the effect. This really throws the rhythm out and must have made the guitar part very difficult to play if, as I suspect, it was printed to tape with the effect. Given the space in the track in terms of frequency range, the guitar sound should have been bigger. It lacked lower-mid, and even when the keyboards joined in there still seemed to be a big hole between the upper-mid frequencies and the bass guitar. Incidentally, the bass was well played by David Sheppard and I liked the use of bass chords on the sparse 'Needle in my Vein', where the full sound in the lower mids offset the thin guitar somewhat.

BLUE GALAXY: Blue Galaxy are a duo who have worked together in talent competitions but seem to play in separate club bands. The material is well constructed but they really need to get away from their working-men's-club and talent-show roots, which tend to date the arrangements and instrumentation. At times it seems as if they're about to attempt a rap or a decent break, but then suddenly we're back into glitz and grins and cheesy endings. Within the confines of what they do Andrew has a good voice — with strong control of those long held notes in the ballads — but a little too much vibrato for my taste. This should get them far on the circuit and they could even cross over into mainstream given the right production (look at George Michael...). Not that the production's bad on these recordings: engineer and keyboard player Andy Meachin has done a very good job of arranging the songs and mixing them without that much equipment. Occasionally the vocal swings into overload (so watch those levels!), but generally the voice has been well recorded, using the SM58, and the delay seems to suit singer Andy Paisley's voice. ❏

brief encounters

COOL SCHOOL INTERACTUS *CIRCLE OF DIGI VOLUME ONE* CD-ROM

I have to admit that I was intrigued when I first saw *Circle of Digi Vol 1* at a recent AES show, and the fact that the manufacturers were in the process of negotiating UK distribution gave me a good excuse to review a copy. Essentially, *Circle of Digi Vol 1* is a dual-format Mac/PC interactive CD-ROM designed to provide product familiarisation for Digidesign's Pro Tools, *Sound Designer*, Sample Cell and a number of third-party developer options. It all seems like a very good idea — if you need to find out about something, you can go straight to the topic of your choice, read a concise explanation and, in many instances, see a short movie clip.

Rather than present everything as boring menus and lists, *Circle of Digi Vol 1* presents you with a virtual world that looks not unlike a cubist representation of an atom, where the coloured spheres are different modules. There's also the 'Circle of Digi' itself, where the eight major topic modules are arranged as icons on the circumference of a circle, and clicking on one of them takes you to the relevant topic. Of course, this being a multimedia presentation, it doesn't simply hop from module to module, but instead does a kind of hyperspatial 'whoosh' thing, complete with sound effects.

Level 1 of each module provides you with a short list of headers: Introduction, Concepts, Basics and Troubleshooting. Selecting one of these calls up Level 2, comprising two text windows and a third window, which may hold text or a video clip. All the while, the presentation runs commentary, music and sound effects to try to keep your attention — it's all very slick. Level 3 gets you to full-screen flow charts and diagrams, as well as interactive examples. Subjects covered include the fundamentals of sound, computer basics, and MIDI and digital audio; there are also many photos and screenshots, some animated.

Finding your way around *Circle of Digi Vol 1* is pretty easy, but my main gripe with any product of this type is that your progress is slowed by having to wait for the beautiful but gratuitous graphics to draw themselves every time you move to a different location. What's more, when you finally do arrive, it's often to find that your ultimate reward is just a few paragraphs of text that don't go into nearly as much detail as you'd hoped. Some of the movie clips are helpful, and there's no doubt that a lot of thought has gone into making this a polished learning aid, but after just an hour or so with it, I felt an overwhelming urge to go back to the faster and more thorough owner's manual supplied with the relevant Digidesign products. My PC also crashed a couple of times while using the CD-ROM.

Circle of Digi Vol 1 is an entertaining and useful introduction to Pro Tools, *Sound Designer* and Sample Cell, but it isn't a substitute for a user manual, and I don't think that it has a great deal to offer to anyone already using the products in question. A great idea, but in many ways it's a triumph of presentation over content, and even on a Pentium PC running a quad-speed CD-ROM drive, it feels slow. *Paul White*

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Low-cost jack patchbays are available from a number of sources, but most tend to use the same type of budget jack socket to keep the price down. This is OK for a while, but in applications involving normalising, such as insert points, cheap jacks very quickly start to become intermittent, leaving you cursing as your signal comes and goes without warning.

Neutrik are manufacturers of high-quality connectors, so it's not surprising that the jack sockets used in this patchbay are far more elaborate, and somewhat more costly, than you'd find in a budget unit. Aside from having gold-plated contacts, the contacts themselves are largely enclosed by the plastic moulding, so there's less chance of dirt physically settling on them.

Whereas most jack bays seen in home studios are little deeper than the sockets they contain, this one is built into a fairly deep plated-steel chassis, the rear edge of which serves as a tie-down point to anchor cables. All the jacks are TRS stereo and terminate at the rear in spring-contact barrier strips, designed to accept bare wires without soldering. All you do is push back the little white lever above the appropriate orifice, poke in the wire and let go of the lever, and the wire is held fast. Similarly, the normalisation options are handled by small plastic 'jumpers' which plug into pairs of pin connectors on the main fibreglass motherboard. A separate link determines whether or not the socket grounds are connected to the chassis ground.

On the front of the patchbay, the sockets are surrounded by push-on plastic cosmetic trims; though black is pretty fashionable in studios, you can replace them with coloured trims if it helps you navigate. Furthermore, along the centre of the panel are two strips of card covered by clear acrylic strips for labelling the sockets.

This certainly isn't a cheap way to build a patchbay, and the cost comes to around three or four times what you'd pay for a budget model, but in critical applications, such as normalised insert points, I think that the peace of mind could well be worth the extra expense, especially if you own a serious project studio. It's also worth remembering that what you spend on the patchbay, you're going to save on not having to buy 48 jack plugs to go into the back end. You also save time by not having to solder, and, rather than having to buy a tie bar as an optional extra, you get one built into the basic product. Compared with a top-end studio bay, this one still works out relatively inexpensive,though it's a shame that there's nothing between the truly budget models and the cheapest high-quality units such as this one. But if you've already fallen foul of intermittent patchbay connections, you probably don't need me to persuade you of the advantages of a pedigree product. *Paul White*

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▼ Big Fish Audio (USA)

D-DAY
2CDs/145 minutes of all-new construction kits, drum loops and hits, synth & percussion effects and more. This is the second release in the Megaton Bomb series. Styles include: jungle, electro, west coast breaks, trip-hop, hard-hop, gabber- and of course, House & Trance (goa, hard trance, melodic trance, etc). Several of the contributors are internationally recognized and have toured and/or released records all over the world. The quality really shows. Also features drum machines, Junk yard and industrial clanks and scrapes, individual hits, loops and layers, vocoder sounds, west coast breaks, live drummers, Synth loops (TB-303's, MC-202-controlled synths, Nord Lead and lots more). Audio 2-CD set: £69.95

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Audio CD: £59.95, CD-ROM: £199

WRH

12-bit Hit

ROLAND S330 SAMPLER

DARIUS POCHA suggests that one of Roland's 'ones that got away' is an excellent second-hand buy: the S330 sampler. Additional material by DEREK JOHNSON.

Roland have something of a history of making novel products which are not highly regarded or fashionable when in production but later become cult objects — often with with wallet-frightening prices on the second-hand market. Look no further than the CR78, TR808 and 909 drum machines and that silver, boxy thing that makes the funny bass noises. This is probably because the instruments are initially cheap second-hand and are picked up by bedroom musicians who, in the words of Roland's Alan Townsend, "start playing with them creatively and realise they've got something". With reference to the above-mentioned silver box, he notes "It certainly helped our reputation..."

One Roland product that seems to have slipped through the net thus far is the 12-bit S330 sampler, almost the last of Roland's series of 12-bit sampling devices, which also included the QuickDisk-equipped MKS10 and S10. This is a shame or — depending upon how you look at it — an opportunity, because if you're in the market for a second-hand hardware sampler you should definitely give it a look.

ROLAND AROUND IN THAT STUFF

What's so special about it, then? Well, a number of things...

Firstly, it has a very powerful resonant filter and modulation stage, allowing you to mangle your samples in an entertainingly strange fashion, perfect for trip-hop, ambient and techno stylings, as well as for other creative applications. Ally this to eight individual polyphonic outputs (allowing you to effect and pan samples independently) and you have a very musical and creative machine. There's even a software sequencer which runs on the S330 (called *Director-S*), and it's still available from Roland.

Another feature which makes the S330 an excellent buy is its support for a monitor and mouse. I'm sure that the majority of hi-tech musicians are familiar with using screen-based editing and a mouse, if only through their knowledge of word-processors. The advantages conferred by working screen-and-mouse style are considerable for this type of machine. Navigating the operating system, editing waveforms and setting up keyboard maps is, if not exactly a doddle, a damn site easier than the pin-the-tail-on-the-donkey techniques necessitated by most hardware samplers with their tiny LCD screens. It's a testament to their original sampling concept, as seen in the S330, that Roland have carried it through to the current S760, albeit in a more powerful and sophisticated form.

You really do need the monitor to make the most of the S330, as its own LCD is cramped and only shows one parameter at a time. For this reason most second-hand S330s will have the original Philips monitor with them. If the unit you're looking at also has the optional mouse, this is worth having, although the mouse may still be available from Roland as a (rather pricey) spare. Standard PC or Mac mice *do not* work. The external control socket that usually hosts the mouse can also be used for plugging in the optional RC100 remote controller, which brings all the front-panel controls and an alpha dial together into a neat unit which can be used anywhere within a 5-metre radius of the sampler. Having an RC100 doesn't prevent you from using the mouse, however, as the RC100 itself features an external control socket, to which the mouse can be connected.

If you're lucky when buying an S330 second-hand, you might also get the vendor to throw in some of Roland's excellent library disks, which feature some of the best samples you'll find for a 12-bit instrument. The Harpsichord and Sitar disks are particularly worthy of mention, as are virtually all the orchestral sounds.

The S330, like the S760, occupies an economical 1U of rack space, although both units are very deep front to back, which can

> "...it has a very powerful resonant filter and modulation stage, allowing you to mangle your samples in an entertainingly strange fashion..."

make them fiddly to mount in a rack. The S330's spec is good, offering 8-part multitimbrality, 16-note polyphony, and MIDI In, Out and Thru.

THANKS FOR THE MEMORY

Two sample rates are available: 15kHz and 30kHz. Although they're not current industry-standard format (the S330 is, after all, a 12-bit sampler), S330 samples can be converted by some software if you want to load them into other equipment, or can be grabbed via MIDI Sample Dump by another sampler. Stefano Daino's shareware Mac sample editor, called ▶

▶ *D-SoundPro*, will let you load samples from S330/S550 disks, converting them to standard 16-bit sample data. This editor can be found, amongst loads of additional useful software, on the *Music & MIDI for the Macintosh* CD-ROM, available from Time & Space (01442 870681) for £29.95, or on the Internet from Shareware Music Machine (http://www.hit squad.com/smm/midi/mac.html

In practice, the 15kHz sampling rate is adequate for many applications, and is actually preferable in some cases, as it gives loops and samples a nice 'grainy' feel. Arguably Akai S900/950 samples have more 'character' to their sound but the S330 — in my view at least — more than makes up for this with its wild resonant filters and LFO. To my ears, it also has much better bass reproduction than the Akais.

The S330's sample memory is divided into two banks, each with a maximum capacity of 14.4 seconds at 15kHz or 7.2 seconds at 30kHz. Each bank can hold a mix of 15kHz and 30kHz samples. Oddly (and annoyingly) the two banks can't be joined together to allow one long sample. In fact, the limited sample memory is the only major fly in the S330 ointment, though, in mitigation, individual samples can be used to form the basis of several 'Tones', each of which can have drastically different settings. This really helps to stretch the available sample memory

— which is just as well, since the memory, sadly, can't be upgraded, and connection to a hard drive isn't possible. However, if you can live with this and don't mind swapping disks a bit, you'll be getting a fine sampler for your money. In fact, apart from time-stretching, the S330 does just about everything you need from a stand-alone sampler.

The S550, complete with monitor, RC100 remote control and mouse.

BAND IN A BOX: THE W30

Not everyone is aware that the sampling guts of the S330 were welded onto a keyboard and 8-track sequencer, to become the W30 sampling workstation, which rapidly found favour with itinerant DJs and dance musicians. Unlike the S330, the W30 has a SCSI option, though you're unlikely to find it separately now, so if you want it, you'll have to find a W30 with it already fitted. The W30 also features two banks of ROM samples, which are a bit of a mixed bag, but include basic drum sounds and staple timbres to get you going. Be aware that S550 and S330 library disks can be loaded by the W30, but W30 disks can't be used in either of the rackmount samplers unless they're first put through a conversion routine within the workstation.

W30s come up fairly regularly on the second-hand market; a reasonable price would be around £6-700.

ONE CAREFUL OWNER

S330s seem to be a good bet for reliability, being well built, with tough disk drives. The earlier version of the software can be a bit unstable, however, so if you get one with software lower than version 1.02 it's worth having a word with the nice people at Roland's Technical Department. If you send them a couple of single-sided DD disks they can give you a copy of the latest version.

Don't expect to find the classified ads stuffed with S330s. On average, you can hope to see about one or two a month in the *SOS* Free Ads. You could also try papers such as *Trade-It* or *Loot*, and the second-hand columns in the dealer ads are also worth a look. If you buy from a dealer, you may even get a short warranty.

Before handing over the money, make sure your unit comes complete with manual.

UPPING THE STAKES: THE S550

If you've got a few more quid to spare, it might be worth holding out for an S550, which was released before the S330 and is essentially two 330s in a box, with double the sample memory (though this is arranged as four banks of 7.2 seconds each at 30kHz, which, as with the 330, can't be joined together). Its greater capacity is reflected in its larger (2U high) and more professional-looking packaging. As with the W30, there's a SCSI option for adding a hard disk or CD-ROM — the entire Roland 12-bit sample library was available on CD-ROM. The S550, naturally, also lets you work with a monitor, and a second-hand one should set you back around £450-550 — perversely, not that much more than an S330.

Initially, at least, you'll definitely need it, but expect it to be a cause of severe frustration. In the time-honoured Roland tradition, it appears to have originally been written in less than lucid Japanese and then translated into equally inscrutable English.

CONCLUSION

Given its high specification, relative ease of use and creative potential, the S330 could be a very rewarding buy second-hand, with its powerful filters, eight individual outputs and full-on bass end allowing the creation of some very distinctive noises. In a couple of years it could be you saying to your envious friends "...yeah, got it off this bloke who was getting rid of all his old stuff. Only cost me 350 notes." [SOS]

THE ONLY REAL SALE turnkey THE ONLY REAL SALE

We've been clearing the shelves in readiness for the new year. If you're ever likely to need one of these items, then don't miss this once in a lifetime opportunity to pick up second-hand and ex-demo items at a fraction of the usual so-called discount prices offered by the other dealers!

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Alesis ADAT [needs attention] . . . . £ 1,169	JBL Control 8SR . . . . £ 329	Viscount RD70 . . . . £ 149
	JLCooper CS1 . . . . £ 269	Yamaha CBK-K1 . . . . £ 99
Alesis S4+ . . . . £ 469	KAM GM40 . . . . £ 69	Yamaha FX770 . . . . £ 199
Aphex Compeller . . . . £ 499	Langevin Dual Elec/Opt Limiter . . . £ 599	Yamaha MD4 . . . . £ 649
Aphex Expressor . . . . £ 299	Lexicon LXP15 MkII . . . . £ 549	Yamaha MFC06 . . . . £ 49
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Evolution MK 149 . . . . £ 79	Soundcraft Spirit 12:2 Mixer . £ 249	Yamaha RY30 . . . . £ 265
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Fostex DMT-8 . . . . £ 999	Steinberg ACI . . . . £ 206	Yamaha YSTM10 . . . . £ 49
Gemini PMX-15 . . . . £ 49	Tannoy PBM 6.5 [single] . . . . £ 29	
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Seasoned Professional

TC ELECTRONIC 1210 SPATIAL EXPANDER

Although still available new, TC Electronic's 1210 is actually older than much of the gear featured in Retrozone. However, it's not a synth, but an effects processor. NORMAN FAY explains his love of this veteran among effects.

The TC Electronic 1210 spatial expander and stereo chorus/flanger certainly is an odd one. It's lurked in TC's catalogue for over a decade, and this alone distinguishes it as peculiar, with hi-tech music products rarely staying in production for more than a few years. Also strange is the fact that the 1210 is an *analogue* time-domain processor. Although there's been a resurgence in the manufacture of analogue (particularly valve-based) units in recent years, these seem to be mainly compressors, EQs and mic preamps. Effects processors are still almost all digital, which leaves TC's 1U rackmounting oldie in a field of one, as far as I'm aware.

CONTROLS & USER INTERFACE

On either side of the 1210's rather ordinary front panel are identical sets of controls for its two channels, with a set of four buttons in the middle (with associated LEDs) to control the input and output configurations, which I'll come to in a moment. Each channel has an Input Level control with a 5-segment LED bargraph and a Bypass button. The actual effect controls for each channel are in two sections. The first contains an Intensity knob, which has a different effect depending on which of four modes is selected by the other control here, the Mode button. In mode 1, the Intensity knob controls the amount of direct signal, in mode 2 the amount of effect signal, in mode 3 the regen amount, and in mode 4, the amount of inverse feedback. It sounds confusing at first, but you soon work it out. The second section consists of three knobs. The first is for LFO speed — variable over a range of 0.1Hz to 10Hz, and with a flashing Rate LED. The second knob is for effect depth (or Width, as it is called

here), variable over a range of 0-100%, while the delay knob, which is next, varies the delay time from 0.65 to 12 milliseconds.

Two buttons each for the so-called Input mode and Link modes make up the aforementioned four buttons in the middle. The first Input mode button is marked Stereo Inputs, and allows you to choose whether the TC1210 operates as a mono-in/stereo-out affair, or as stereo throughout. The second Input mode button is marked Separate Inputs and, when selected, it divides the 1210 into two separate mono-in, stereo out units, and links the Bypass buttons, so you can switch both channels on or off at once.

The Link mode buttons, which come next, interact with the Separate Input button. The first, the Cross Off button, switches out cross-mixing of the two channels when the Separate Inputs button is off, but when this is on, Cross Off merely disables the link between the Bypass buttons — why, I don't know! The second Link mode button is marked Sweep Sync, and when the Separate Inputs button is off, the expected effect is produced — the two LFOs are synchronised. When the Separate Inputs button is on, Sweep Sync actually mixes the LFO signals together, with often interesting results.

As far as rear-panel connections go, the 1210 is uncommonly well specified. As well as the mains input and voltage selector (no 'wall warts' here), there are sockets for a Bypass footswitch and a Sweep Speed footpedal. As for signal inputs and outputs, the 1210 has two mono input jack sockets, two stereo output jacks, and balanced XLR duplicates of the inputs and outputs.

WHAT IT SOUNDS LIKE

Even 10 years on, the effect produced by the TC 1210 is still a real winner; the depth and richness of the effects it generates have to be heard to be believed. The effect is also deceptively subtle. After leaving it on for a while, you start to wonder if it's actually doing anything and switch it out, only to be horrified at the thin, lifeless sound that's left! When mixing down, I've often found that using the 1210 on backing parts and leaving the lead parts uneffected gives the lead part more space in which to be heard. I've used the 1210 on a wide variety of sources during the time I've had it, and have found that there isn't much that it can't improve. I especially

like to use it on synth pads, backing vocals, old string machines, and, best of all, acoustic guitar, on which it sounds lovely! It's also interesting to run complete tracks through the 1210, where you'll find it simple to reprocess a mono track into stereo, and to produce an effect similar to old-fashioned tape phasing.

SO WHERE'S THE CATCH?

Now, here's the bad news. The TC1210 may be lovely, but it's shockingly expensive — currently £1155 (*call UK distributors Raper & Wayman on 0181 800 8288 for details — Ed*). This is for a device which can only be called a multi-effects unit by virtue of the fact that it has two channels. On the face of it, it seems to be terrible value for money, but what you have to bear in mind here is that you're getting a professional, beautifully-constructed machine, with a genuinely different sound — and no, I'm not on a retainer from TC Electronic! When I bought my 1210, I could have got a lot of gear for the price, but I've never regretted buying it. If you can spare the cash, this is a strongly-recommended purchase. **SOS**

Ghost

Ghost of precisely these specifications as this month's incredible competition prize. Those who have already bought a Ghost needn't feel left out — they can win one of the new 24-channel expansion modules instead, worth £4894. Even if you don't win the big prize, you needn't be too despondent, as Soundcraft are offering Rugby shirts as runner-up prizes for the best 10 tie-breakers.

As this prize is such a biggie, we're asking a little more of you this time than in previous competitions. Of course, there's all the usual stuff — you still have to answer some pathetically easy questions and complete a tie-breaker in a manner guaranteed to have the SOS staff calling for a team of paramedics to attend to several bad cases of split sides — but this time we want a little *extra* commitment! Elsewhere on this page, you'll find a list of Ghost dealers located all around the UK, and in Eire. Each of these will be displaying a large Ghost serial number (different for each dealer) in their stores over the next couple of months. You need to include the name of one of the dealers and their Ghost serial number to validate your competition entry — and needless to say, incomplete forms will not be entered into the competition. If you can manage all that, we're certain you can ensure that you find a post box somewhere near you and send your entry so that it arrives before the closing date **(Friday April 4th, 1997)**. It's not really *that* much more to ask, is it?

Prizes kindly donated by Soundcraft (01707 665000). We regret that this competition is open to UK and Eire residents only.

Soundcraft's Ghost is a highly-versatile professional 8-buss 24- or 32-channel mixer, packed with facilities for all kinds of recording and mixing applications. Some of its features are normally only seen on far more expensive consoles, such as the stunning 4-band parametric EQ, the 10 (yes, 10) auxiliary busses (including 2 stereo pairs), and the MMC-compatible transport controls and MIDI muting (compatible with all industry-standard analogue and digital 8-track, hard disk, and 9-pin VT recorders). Of course, for the cost-conscious, the Ghost is also available as a cheaper, automation-less version, the Ghost Le. Now, the Ghost's already-vast capabilities can be enhanced still further with a 24-channel expansion module, which has just been released. The Ghost haunted David Mellor so much in his October '96 SOS review that he was moved to comment, "I doubt such sound and build quality, and well-directed facilities have ever been available before at such a canny price". It seems to have captivated plenty of others too — in the short time since it was launched, the Ghost has gained a number of high-profile users, including Virgin Radio, techno outfit Bandulu, and remixer Phil Kelsey.

But although the price of the Ghost is indeed keen when you consider the facilities it offers, it's still a sizeable hole in a bank account — for example, a 24-channel Ghost, with meterbridge and stand, would set you back a massive £5792.75! And that's why SOS have teamed up with Soundcraft to offer a

questions

1. What type of EQ does the Ghost offer?
a. HF and LF shelving filters only ☐
b. 3-band, with sweep mid ☐
c. 4-band, fully parametric ☐
d. Grimethorpe Colliery Band, MFI shelving ☐

2. Which model of Ghost does not offer the automation facilites found on the standard version?
a. Marley's Ghost ☐
b. Caspar The Friendly Ghost ☐
c. The Ghost Busters ☐
d. The Ghost Le ☐

3. How many aux busses are offered by the Ghost?
a. 2 ☐
b. 4 ☐
c. Oh... most of London Transport's reserve fleet ☐
d. 10 ☐

4. How many extra channels are offered by the Ghost's new expansion console?
a. 8 ☐
b. 16 ☐
c. 24 ☐
d. Blue! er... ☐

TIE BREAKER

Complete the following sentence in not more than 30 words: "I think Soundcraft's desks are the dBs because..."

....
....
....
....
....

Name ....

Address ....
....
.... Daytime tel. no ....

Ghost dealer visited ....

Ghost serial no. at above dealer ....

Post your entry to: SOS Ghost Competition, *Sound On Sound*, Media House, Trafalgar Way, Bar Hill, Cambridge CB3 8SQ.

Would you like to receive further information on Soundcraft products? If not, please tick this box. ☐

Ghost Competition

PARTICIPATING DEALERS

- CTI, Dublin (00353 1 454 5400).
- CTS, Nottingham (01623 721222).
- KGM, Wakefield (01924 371766).
- Marquee Audio, London (01932 566777).
- Raper & Wayman, London (0181 800 8288).
- Sound Control, Manchester (0161 877 6262).
- Sound Control, Glasgow (0141 204 0322).
- The Warehouse, Edinburgh (0131 555 6900).
- Turnkey, London (0171 379 5148).
- Way Out West, London (0181 744 1040).
- X Music, Bristol (0117 973 4734).

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OR WRITE YOUR NAME, ADDRESS AND TELEPHONE NUMBER ANYWHERE ON THIS PAGE, TEAR IT OUT, FOLD IT AND SEND IT TO US.

60 with case, £275; Yamaha TX802 with RAM cartridge, £325; EMT10, £100 All excellent. ☎ 01726 66715.

MOOG PRODIGY vgc, £200; Moog Rogue vgc recently serviced, £165 ARP Omni 2 vgc, £125; Korg Mini 700, £120 Emulator one with flightcase for spares, offers. ☎ 0116 283 5455.

NOVATION BASSTATION £250; Cheetah MS6, faulty edit mode but plays fine, £100; Sequential Circuits Prophet 5, rev 2, £775; wanted, Roland D70, MKS50, PG300. ☎ Brian 01752 550199 days.

PEAVEY DPM2 vgc, £450; Yamaha CS01, £60. ☎ Alan 01273 870179.

QUASIMIDI TECHNOX dance module, over 500 outstanding analogue sounds, classic 808/909 drums and 100 RAM/ROM performances, as new, £550. ☎ 01244 382815.

ROLAND CM32L sound module, £75; Yamaha MEP4 MIDI mixer effects unit, £95; Simmons SPM8 8-channel mixer MIDI control, £115. ☎ David 01283 791781.

ROLAND D10 multitimbral synth with hard case and ROM/RAM cards, £350; Roland U110 PCM sound module with 2 cards, £200. ☎ Mark 01332 571537.

ROLAND D10 with analogue sounds loaded, £265 ono. ☎ 01684 893698 after 7pm.

ROLAND D110 with D131 gold card and Valhala Analog card. £150 ono. ☎ 01293 427285 (Crawley).

ROLAND D50 perfect condition, manuals, £375; Roland Rhodes MKS80 great expressive rhodes and piano sounds, weighted keys, stand, manual, £800. ☎ Roger 01923 266664.

ROLAND D50 mint condition with sound cards and manuals, £375; Wanted, D50 RAM cards M256D, M256E. ☎ 01363 775214 (Devon).

ROLAND D50 two sound cards, flight case, £450; Alesis SR16 drum machine, £150. Both in excellent condition. ☎ 0973 956149.

ROLAND D110 multitimbral synth modules two for sale, Manuals, plus over 2000 sounds, £200 each. ☎ 01273 564286.

ROLAND MKS70 with one RAM and three ROM cartridges, manual, excellent condition, £450; Roland Juno 2, excellent condition, £350; Roland D50, £350. Swap for Akai S1000, S1100. ☎ 01302 835653.

ROLAND JD800 immaculate, boxed, manuals, £1350; Ezra X50 single manual drawbar organ, MIDI, overdrive, rotary, in own flightcase, B3 to cheesy, brilliant value, £250. ☎ Tim 01473 225503 eves.

ROLAND JD800 keyboard as new, £1100 ono. ☎ 01474 312644.

ROLAND JV1080 with memory card, £800; Roland D50 with stand, £400; Roland FP88 piano with flightcase, £750; Gem S2 workstation with case, £800. ☎ Robin 01223 363837.

ROLAND JV1080 mint condition, £825 ono; Yamaha SY22 mint condition, £250 ono. ☎ Ian 01484 646242.

ROLAND JX3P classic MIDI analogue polysynth, good condition, manual, £250; Chase Crumar Bit One rare synth, a la DX7, MIDI, good condition, manual, £220. ☎ Dave 01305 766600.

ROLAND JUNO 106 excellent condition, includes owners manual, £400. ☎ 01274 620004 (Bradford).

ROLAND JUNO 6 RE201 Space Echo, Emu Proformance, Fender Strat, Marshall JCM800, Roland GSG guitar processor, DM1000 delay, swap/px VS880, DMT8, MPC60II, Tascam 688, JV1080. ☎ Ian 01539 723631. .

ROLAND JUNO 2 vgc, £350; Roland MKS70, vgc, £450; Akai S1000, S1100 wanted. Must be in good condition, cash waiting. ☎ 01302 835653.

ROLAND JUPITER 6 £795; four Roland D50 ROMs for £30; 2.8-inch Quick Disks, £2 each; Yamaha QY10, £70. ☎ 01268 525347 (Essex).

ROLAND JV1080 boxed, as new, with manuals, £800. ☎ 0181 800 9944 days, 0181 800 1040 eves.

ROLAND JX10 recent service flightcase, £750; Roland D50, £450; Emu Emax, £450. ☎ 01494 713359 eves.

ROLAND JUPITER 6 with MIDI, 12 VCOs, split keyboard mode, very good condition, £750. ☎ 0410 458061 (Newcastle).

ROLAND MKS80 fat analogue monosynth with MPG80 programmer, £1650; Mackie CR1604 with rack ears, gooseneck light, excellent condition, boxed, £590. Don't call if you can't afford it. ☎ John 0956 127679.

ROLAND MT32 sound module, plus GM patches on PC disk, £95; MP40 PC MIDI card, £55. ☎ John 01606 331343.

ROLAND SH5 classic monosynth, excellent condition, includes owners manual, reluctant sale, £600. ☎ 01274 620004 (Bradford).

ROLAND SH101 three available: limited edition red with mod grip, £300; limited edition blue with mod grip, £300; Grey, £225. All have original PSUs and owners manuals, in excellent condition. ☎ 01274 620004 (Bradford).

ROLAND SH101 vgc, PSU and manual, £250, swap for Roland JX3P, R8, similar. ☎ Eric 0131 556 9600 (Edinburgh).

ROLAND U110 PCM sound module with user manual, £200. ☎ Darren 01328 856055 (Norfolk).

ROLAND JUNO 106 £350; JX3P with PG200 and flight case, £295; U220 with 2 cards, £250; Emu Proteus MPS, £395; Drawmer DS201 dual noise gate, £195. ☎ 01305 774367.

ROLAND XP10 multi-timbral keyboard, Roland sounds, arpeggio and stand, £265. ☎ Nick 01423 879168.

ROLAND XP10 brand new, 338 sounds, arpeggiator, superb all round synth and value, boxed, £390. ☎ 0831 771 491 (London).

ROLAND JV1080 plus dance card, £800, Roland MC303 plus PC200 MKII, £500 pair, all manuals. ☎ Rick 0161 856 0869 days, 0161 684 7195 eves.

ROLAND MKS70 JX10 rack synth analogue classic, pristine condition inc 3 ROMs, £650 or consider P/X TOA KD3; Atari Falcon HP600 printer; Roland PK5/XP50. ☎ Robin 01522 752458 eves.

ROLAND XP80 synthesizer new condition, £1400 ono. ☎ 01734 882079 (Reading).

ROLAND U20 keyboard with manual, excellent condition, £390. ☎ 01527 528930 (W Midlands).

ROLAND D10 synth, vgc, manuals and stand included, £300 ono. ☎

Matt 01705 756494 (Portsmouth).

ROLAND D70 super LA, resonance 76 keys, £499; Korg DW8000, £285; EMU MPS keyboard 32 Poly effects, £345; Juno 6, £175; Yamaha SK30, £175, will post. ☎ 01261 832898.

ROLAND XP10 synthesizer, 300+ sounds, arpeggiator, reverb, chorus, 16 drum kits, performance mode, GM/GS, etc as new with all the bits, £380. ☎ 01933 678608.

ROLAND JUPITER 6 absolutely mint condition MODI Pro flight case, best example in the country, £875. ☎ 0958 311 277.

ROLAND JV35 synth plus VEJV1 expansion card, excellent condition, £400. ☎ Martin 01223 360502.

ROLAND U220 with two sound cards, £250. Emu Proteus MPS keyboard, £395. Drawmer DS201 dual noise gate, £195. ☎ 01305 774367.

ROLAND D5 with manuals and memory card, excellent condition, £265. ☎ Colin 0181 337 9306 eves.

SEQUENTIAL PROPHET 5 rev3.3, £800; Pro One, £275; Roland SH2, two oscillator with sub-oscillators, £175; Roland SH5, £600; blue SH101 with grip, strap, case and PSU, £300. ☎ 01274 620004.

SEQUENTIAL PRO 1 synth, good condition, £330 ono, PX for Korg Prophecy. ☎ 01483 425733 or 561430 eves.

YAMAHA FB01 £40. ☎ Dave 01543 454253, 0973 540571.

YAMAHA VL7 physical modelling synth, £430; Yamaha V50 FM workstation, £270; Spirit Folio 12:2 mixer, £200, all boxed with manuals and in excellent condition. ☎ John 0973 132872.

YAMAHA SY85 mint condition, extra software, 64K RAM card, all manuals, original box, £650, will deliver UK. ☎ Steve 01624 676677 days.

YAMAHA DX7 with manuals and box, two ROM and three RAM cards and hundreds of sounds on disk for Atari, £195, no offers. ☎ 01255 431288 (Essex).

YAMAHA CLP122S Clavinova piano other voices, touch sensitive, 88 keys, MIDI, built in speakers, perfect as new condition, wood effect case plus stool, can deliver, £895. ☎ 0121 742 0330.

YAMAHA KX88 mother keyboard, home/church use only, £650; Roland P330 piano module, £250; Rack flight case, £50. ☎ 01609 775396.

YAMAHA SY22 vector synth leads and manual only, £250. ☎ Dave 01782 745065.

YAMAHA TG100 sound module on PC interface, £150; Studiomaster Diamond 12-channel mixer, all boxed manuals, home use only as new, £230. ☎ 01737 557792 (Surrey).

YAMAHA PFP100 digital piano 88, full-size weighted keys. 19 months old, vgc, excellent sound, £1050 ono. ☎ 01603 632736, 01494 529996.

YAMAHA SY55 £375; Roland D50, £375; Roland SF35 MIDI file player, £495, boxed, all with manuals, etc; 3-Tier keyboard stand, £70; Roland EV5 volume pedal, £25. ☎ 01229 466238.

YAMAHA SY99 FM/AWM workstation, excellent condition with 40Mb of top sounds and sequences on disk and manuals,

box and stand, £1100 ono. ☎ 01904 330695.

YAMAHA TX802 rackmount version of DX7 IID 8-part multitimbral, FM sounds, full MIDI spec, very good condition with manual, £290. ☎ 0161 440 8027.

YAMAHA TX81Z £150; Korg Poly 800, £120; Yamaha DD12 pads/drum machine, £80; Korg DDM20 percussion, £30; Echotec analogue delay, £20; Vesta Fire flanger, £20. All immaculate. ☎ Mark 01872 863966.

YAMAHA V50 workstation keyboard, 400 sounds, drum machine, sequencer, disk drive, excellent condition, £300. ☎ James 0115 937 2326.

YAMAHA W7 workstation synth, piano board, 3 sound disks, flightcase, £849, swap for Korg X3; Kawai K1M module, £139, or £925 the lot. ☎ Ben 01482 494737.

YAMAHA TG100 module, as new with original box, manuals, £120. ☎ Pete 01276 675678 day Mobile 0468 393875 Hampshire.

YAMAHA W7 new with dance and vintage boards plus sounds disks boxed. ☎ Ian 01207 509866.

YAMAHA W7 workstation, 4 months old including manuals semi flight case, £849 ono, swap/part exchange for Korg X3, Roland XP50, JV1000, Korg N364. ☎ Ben 01482 494737 (Hull).

RECORDING

AIWA HDS1 portable DAT, £250; Technics SV260 pro DAT with heavy duty, XLRS etc, slight fault thus £150. ☎ 0141 423 4824.

AKAI MG1214 multitrack upgraded version, mint condition and Studiomaster Mixdown Gold 24:8:16:2 with MIDI mutes, immaculate, both with manuals. Open to offers, PX for ADAT, Soundtracs Topaz. ☎ 0181 802 0748.

AKAI STEREO GRAPHIC £50; Tascam 644 MIDIstudio, perfect, £450; Casio CSM10P sampled piano module, £50; Monitor Audio R352 speakers, brilliant, £150. All vgc, manuals etc. ☎ 01256 468208.

AKAI DR4D hard disk recorder, MIDI clock card, £850. ☎ 01753 546374 (Slough).

AKG C414B ULS Mic, £700; Studiomaster P7 16-channel mixer, £950; Akai XE8 drum expander, £80; Tascam DX40 N/R unit, £150. ☎ 0171 385 3332.

AMPEX 797 1 inch professional wideband tape on 10.5 inch alloy spools, three at £20 each; two empty spools, £10 each. ☎ John 01822 854389.

APHEX TYPE C aural exciter, mono, half rack, never used, £60. ☎ Dave 01543 454253.

AUDIO TECHNICA Artist series mics, eight available, perfect for drums and pro studio, live use, boxed, mint condition, cost over, £1900 studio closure forces sale, £900 ono. ☎ Tom 0121 449 7229.

BOSS VT1 Voice transformer boxed as new, £290; JLC PPS100 MIDI to SMPTE converter, £90; TR707 drum machine, £75; Roland D110 synth module, £175. ☎ 0117 955 3668.

DBX II stereo noise compression expander unit, £50 ☎ Dave 01543 454253 or 0973 540571.

DIGITECH VOCALIST II barely

used, Boxed with manual, £325. ☎ 0181 575 5573.

FOSTEX A8 reel to reel recorder, just serviced, perfect condition, bargain, £395. ☎ 0181 224 2240 Surrey.

FOSTEX E16 £1600 ono; Seck 18:8:2 with loom, £600 ono. Drawmer LX20 compressor, £150 ono; JPL100 sequencer, £150 ono. All excellent condition. ☎ 01242 820965.

FOSTEX E16 with rackmount kit, excellent condition, few tapes, £1450 ono; Alesis Midiverb III, £195; Alesis Microverb III, £120. ☎ 01483 277708.

FOSTEX 380S boxed, manual as new, light home use, £450 ono. ☎ Andrew 016974 78069.

FOSTEX DMT8 digital multitrack, boxed as new, with manual, £1050. ☎ Paul 0121 475 7999.

FOSTEX DMT8 excellent condition, boxed with manual, £950, swap or P/X for sampler and sequencer. ☎ Ivan 01392 423130.

FOSTEX MODEL 80 superb condition, home use only, six reels of tape, £595; Ferrograph Super 7 2-track master, £150. PX E16 plus cash. ☎ Keith 01780 53900 (S Lincs).

FOSTEX R8 8-track reel to reel, remote extension, boxed, manual, light home use only, £650. ☎ 0121 359 5702.

FOSTEX R8 boxed with manuals, hardly used, £650; Yamaha REV 7, £180; Behringer multiband exciter, £100. ☎ 01273 833950..

FOSTEX R8 with looms, £600; Fostex 450 mixer, £200 vgc; Atari 1040 STE with monitor, £140. ☎ Gerry 0121 706 7770, 0976 712106.

FOSTEX R8 with MTC1 and three reels, limited home use, manuals, but no box, £700; analogue synths and weird drum machines wanted, will part ex. ☎ Carl 0113 282 2239 (Leeds).

FOSTEX R8 8-track, excellent condition, loads of reels, £800; XRI XR300 rackmounting SMPTE and MTC sync box, £120. ☎ Jeff 01242 527179 (Cheltenham).

FOSTEX X28H 4-track recorder, mint condition, hardly used, £220 ono; Midiman sync unit, connects 4-track to sequencer, £50 ono. ☎ Steve 0121 588 2350 (Birmingham).

FOSTEX X26 4-track machine, immaculate with box etc, very low use genuine sale, bargain, £150 ovno. ☎ Steve 0171 470 8822 days (London/Kent).

LA 4X4 compressor gate, studio use only, £160. ☎ John 01222 460757 (Cardiff).

TASCAM 424 MK2 portastudio, 8 months warranty, very high spec, fab 3 band EQ with mid sweep, MIDI sync, many features, boxed, manual, A1 condition, £295 ono. ☎ 0181 678 1304.

LEXICON ALEX reverb, mint, boxed, manual, PSU, £195; Casio VZ8M, boxed, vgc, all manuals, Atari editor, RAM card (doubles memory), multitimbral, different, £119. ☎ Dave 0191 372 2621(Durham).

MACKIE 32:8 8-buss, 8 months old, home use only, with stand, boxed as new, £3250; Mackie Ultramix system, boxed as new, £1900 no offers. ☎ 01293 552623.

MACKIE CR1604 mixer, ex con. with Rotopod, home studio use only, rack ears and bolts included, £565. ☎ 0113 263 7898.

MACKIE CR1604 £600; Ottomix

1604 automation, £400; Macintosh 7100 80, 24Mb, CD, Audiomedia II, keyboard, monitor, leads and lots of software, £1850; Mac 660 AV 22k/44k/48k recording, video in/out, keyboard, monitor, £750; Rocktron IICX, £120. ☎ Mick 0113 246 9254 after 4pm.

MACKIE CR1604 £545; Korg M1R, £450; Drawmer DL221 compressor, £235; Atari 1040 STFM, £85; Yamaha TX7, £99; Tannoy Stratfords, £85; Tascam M15 24:8:16 mixer, £945. ☎ 01903 755840.

MACKIE 1202 VLZ mint condition, boxed with manual and plenty of leads. Rack ears included if needed. Still under warranty, £300 ono. ☎ John 0181 524 0699.

MTR 16:2 mixer, £200; EMO 6-channel phantom power unit, £50; ADC 12-band stereo graphic, £20; Trantec frequency shifter, £15 perfect. Including manuals. ☎ 01634 301593.

PHILLIPS DCC730 18-bit digital recorder, remote, manual, boxed as new, £200; DCC134 portable player, boxed as new, both, £265. ☎ Adrian 01223 354298.

PHILLIPS DCC digital deck, very good condition, with remote control, manuals, as new, £150 ovno. ☎ 01924 257717 (W Yorks).

PHILLIPS DCC stereo digital mastering deck, good condition with manuals, tapes etc, £170; Boss Dr Synth sound module, £145. ☎ 01933 678608.

RAM 10:4:8:2 desk nice sound, £200 ☎ Roger 01450 870225.

ROLAND M12E mixer, rack-mounting, 12-channel, two aux sends with stereo returns, two band EQ, etc, £350. ☎ Andy 01732 762463 eves, 01622 694252 days.

ROLAND MC500 sequencer with hard case, £300; Yamaha EMT10 piano module, excellent sounds, £100. ☎ Mark 01332 571537.

ROLAND JX1 master keyboard 64 classic presets and 32 user settings, four real time controllers, excellent condition, £150. ☎ Tom 0115 950 2862, 0161 483 7230.

ROLAND VS-880 850Mb drive and VASF1 effects board, mint and hardly used, £1450. ☎ Matt 01253 731379 (Blackpool).

ROLAND DM80 pro spec, 4-track hard disk recorder, MIDI, video sync, MAC LCII 6/80 controlls it, £900 both; Studiomaster 16:4:2, flightcased, £400; Yamaha MT120 cassette multitracker, 5-band EQ, 2 speed, direct outs, £180; Phonic compressor, £150. ☎ Mark 01904 653573.

SECK 18:8:2 home use only, £600 ono. ☎ Lenis 01222 258390 (Cardiff).

SECK 18:8:2 MKII mixing desk, perfect condition, new power supply, home use only, £500. ☎ 0181 773 2894.

SENNHEISER MICS 2MD441 and 2MD427, £150 each, immaculate condition, quick sale, new price over £500 each. ☎ 0181 771 6045.

SONY DTC55ES DAT recorder parts serviced except heads need replacing, boxed with manual and remote control, £80. ☎ James 01978 362910.

SOUNDCRAFT 400B mixer, 24-channel, 8 monitor, 8 tape out good for live, studio use, £1350. ☎ 0181 875 9259.

SOUNDCRAFT SPIRIT FOLIO 10:2 mixing desk with manual, hardly used as new, offers. ☎ 01924 478155.

SOUNDCRAFT STUDIO 16:8:16:2, boxed unused with PSU, manuals, £900. ☎ Robert 0181 340 7503.

SOUNDCRAFT RAC PAC £420;Tascam 488 Portastudio, £550; Phonic PCL 3200 compressor, £120; Sony DTC750 DAT, £300; Korg AX30G, £180; Boss micro rack graphic EQ, £50, all excellent. ☎ Mark 0113 273 8841 (Leeds).

SOUNDTRACS MRX 24:16:8:2 mixing console, fabulous desk, £2000; Korg DS8 multitimbral keyboard/synth, £350; Emu Morpheus with dance card, £795; Yamaha QX3 sequencer, 48,000 events, £260; Roland D110 multitimbral synth, £220. ☎ Alan 0181 764 2022.

SPIRIT FOLIO 12:2 mixing desk, 3 band swept mid E.Q, 2 stereo inputs, A1 Condition, £295. ☎ Andy 01784 452253.

SPIRIT FOLIO SI mixer 18:2, 8 stereo inputs, 2 mono inputs with semi-parametric EQ, as new, £320 ono. ☎ Mark 0171 585 1814.

STARSOUND DYNAMIX 12:2 mixer, three band eq, 2 aux sends, £150 ☎ Robert 01707 326704 after 7.30pm.

STUDIO EQUIPMENT for sale. Mixers, effects, dynamics, mics, lots of bits. ☎ 01535 645233.

STUDIOMASTER 16:2 £270; Cheetah SX16 sampler, £275; MMT8 sequencer, £140: Roland PC200 MkII £110; Emagic Sound Diver/Surfer for Atari, £100 or, £820 for everything. ☎ 01733 371773.

STUDIOMASTER MIXDOWN GOLD 16:8:16:2 with MIDI muting, excellent condition, home use only, boxed with manuals, must sell, £750 ono. ☎ John 01822 854389.

TASCAM 22-2 2-track quarter-inch reel to reel, seven inch reels almost as new, £300 No Offers. ☎ 01827 261 059 after 6pm.

TASCAM 38 half-inch 8-track, flight case, remote, looms, degausser, test tape, some reels. Good condition, hardly used, £650 ono. ☎ Adrian 0973 509756.

TASCAM 48 half-inch 8-track, professional standard. Good condition, £550 ono. ☎ 0171 263 8891.

TASCAM 688 8-track MIDI portastudio, low hours, home use only, fully serviced by Teac, mint condition, studio involvement forces sale. ☎ Andy 01743 741261.

TASCAM DA88 8-track recorder, mint condition, with manual and SY88 card, private use, less than 200 hours. ☎ Chris 0181 961 4444 or 0181 908 5710.

TASCAM PORTA 07 boxed, mint condition, hardly used, upgrading to digital, hence, £200. ☎ David 01727 855321.

TASCAM PORTA 2 4-track, £150; Boss BX8 8-channel mixer, £110; Shure Prologue 16L condenser mic, £50; Technics RS-TR575 double deck, £200. All excellent condition. ☎ 0181 693 2398.

TASCAM 424 Portastudio, home use, mint condition, manual and boxed, £290. ☎ Nik 01482 844556.

TASCAM 464 Portastudio, 12-channel mixer section, inserts on 1&2, XLRs 1 to 4, two auxs, 3-band EQ, £400 ono; Trancendent 2000 synth, £140 ono. ☎ 01273 208099 (Brighton).

TASCAM 488 MKII 8-track portastudio, immaculate, boxed with manual and receipt of purchase, new august 96, £590. ☎ Graham 01423 771440.

TASCAM MIDIIZER multitrack sync controller, exc cond; Yamaha RY30 drum machine as new cond; 2KVA uninterruptable power supply, no break in power during short power failure; one-inch and two-inch virgin sealed tape. ☎ Michael 01753 860091.

TASCAM TSR8 £800; Mackie 1604, £500; Alesis Quadraverb+, £140; A3630 compressor, £150; HR16, £90; Yamaha MT3X, £260; PPS100 sync, £80; AKG C100 mic, £120 all boxed. ☎ 01342 313611.

TASCAM M2516 16:8:2 desk, MIDI muting, A1 condition, £750; Akai S950 fully expanded, £550; Boss BX16 16:2 2 mixer, effect sends, home use, £180. ☎ 01932 874682 eves (Surrey).

TL AUDIO EQ1 valve equaliser, dual channel, as new, boxed with manual, £500 ono. ☎ 0161 439 0963.

YAMAHA MD4 Mini Disk multitrack recorder, brand new, unused, £800 ono. ☎ 01483 562810.

YAMAHA MT3X multitrack, 6-channel mixer, 4-track recorder, leads, headphones, cassettes, Yamaha sync box included, all in mint condition, occasional home use only, boxed, £335 inc delivery. ☎ 01606 784394, 0161 499 4266.

YAMAHA MT120S 4-track, mint, boxed, manuals, hardly used and very well-kept, dbx, 4 tracks simultaneous recording, graphic EQ, twin speed, £179 ono. ☎ Dave 0191 372 2621.

YAMAHA PRO-1 digital mixer, boxed with manuals. mint, £1350 ono; Alesis datadisk manual, PSU, £150. ☎ Kevin 0113 252 3542 days, 01924 470384 eves.

YAMAHA SPX90 II multi-effect processor pro reverbs, £190; Digitech GSP27 pro with foot controller, mint, £350; Tascam Porta One Ministudio, hardly used, £190. All with manuals; the lot, £600. ☎ 0121 427 5754.

YAMAHA AM802 mixer, 6 mono, 2 stereo ins, 3 sends, 3 returns, EQ, pan, level fader, very quiet, immaculate, was, £389 accept, £220 ono P/X welcome. ☎ Lee 01472 310247.

YAMAHA PROMIX mixer 01, 6 months old, mint condition, warranty box + manuals, £1200 ono. ☎ Jim 0421 305035.

SAMPLERS

AKAI SO1 mono sampler with quite a few sample disks, offers? ☎ 01293 427285 (Crawley).

AKAI SO1 Sampler mint condition, boxed with manuals plus two hundred sample disks, £450. ☎ Graeme 0161 797 4305.

AKAI S900 Sampler only one owner, perfect condition with library sounds, £300; Kawai KIR, £120; Casio VZ10, £120; Yamaha TG55, £180; Alesis HR16, £60. ☎ Jim 01375 382597.

AKAI S950 sampler, manual vgc £530; Emu Morpheus Z-Plane synth, manual £580; Oberheim Matrix 1000 with extra sounds on Atari disk £240. ☎ Martin 01244 376446 eves.

AKAI S950 digital sampler, good condition, fully expanded to 2Mb, over 300 disk sound library, £700 ono; Alesis MMT8 8-track MIDI sequencer, £80 ono. ☎ Clifford 0181 951 0413.

AKAI S950 fully expanded with samples, hardly used, mint condition, £680 ono. ☎ Jeremy 01273 580323.

AKAI S950 fully expanded with manual, As new with lots of samples, £625. ☎ 0181 365 3318.

AKAI S950 fully upgraded over 60 seconds of sampling time with manual, £675 ovno; Proteus 1 module, £280. ☎ Jez 0151 728 7254.

AKAI S3000 stereo MIDI sampler 2Mb, manuals, excellent condition never gigged, £1400; MM 16:2 mixing desk, 4-band EQ, aux send, foldback, £150 ono. ☎ 0113 275 0235.

AKAI S1000 8Mb, digital in/out, SCSI, latest software, huge library, £1200 ono. ☎ Rob 01443 755425.

AKAI S1100 8Mb, SCSI digital in/out, effects, 10 outputs, boxed, 25 disks, £1100; Akai S1000PB, 8Mb, SCSI, 10 outputs, boxed, 25 disks, £600 or, £1500 both. ☎ 01553 617497.

AKAI S01 sampler, upgraded memory, very little use, many samples on disc, £400. ☎ Tony 01709 556274.

AKAI S1000 26Mb, digital I/0 excellent condition, home use only, £1350. ☎ Gordon 01225 761360.

AKAI S1000 sampler, 8Mb, digital I/0, SCSI, 11 outputs, V4.3 software, manual, £1350. ☎ James 01978 362910.

AKAI S1100 digital sampler 18Mb RAM, SCSI and SMPTE boards, in-built effects, immaculate condition, manual, boxed, £1600 ono. Consider swap for Roland VS880 hard drive and effects board of similar condition. ☎ 01384 410853.

CASIO FZ10 2Mb, rackmount sampler includes 100s of samples, £530. ☎ Brad 01633 865758 eves, 01633 873118 days.

CASIO FZ20M 16-bit sampler, 8 outs, SCSI port 30 secs at 36kHz but copies samples without using more memory, quick & easy graphical editing, boxed manual, £550. ☎ 0181 742 8913.

CASIO FZ10M sampler, 2Mb, includes large library, £495. ☎ Brad 01633 865758 eves/weekends, 01633 873118 days.

CASIO FZ10 sampler and extensive library, £600 ono; Roland D110 8-part multitimbral sound module and ROM card, £210 ono. ☎ Kevin 0171 381 0046, 0973 892 367.

CHEETAH SX16 16-bit stereo sampler, 8 outputs, 2Mb upgraded, quality samples, long sampling time, boxed as new, £450, no offers. ☎ 0151 228 0803.

CHEETAH SX16sampler 16-bit stereo inputs, reads Akai disks, excellent condition, with manual V3.0 software and library, £300. ☎ Simon 0161 737 4483 eves.

EMU EMAX II 16-bit stereo sampling keyboard 2Mb memory, excellent condition, with disk, £800 ono. ☎ Dave 0181 330 3329 after 6pm.

EMU EMAX II SAMPLER 8 outputs, SCSI, mint, £775; Roland D70 76 keys, mint, £650; Roland SPD8 drum trigger, internal sounds, 808/909 sounds, £280, offers considered. ☎ Dave 01249 713204 (Wilts).

EMU E4 sampler, 128Mb, Morpheus, Ultra Proteus, Vintage Plus, Orbit/Launchpad, Roland JV1080 with cards, Jupiter 8, D550/PG100, U220, TR909, JD800, SH2, Drumstation, Prodigy, Poly 6. Sensible offers. ☎ 01252 370550.

EMU ESI32 with 32Mb RAM, SCSI, leads, vgc, home use only, boxed with manuals and sound library, £1000. ☎ Andy 0181 524 0224.

EMU ESI32 With SCSI and manual, £750; Novation Bass Station rack with manual, £280; Alesis D4 drum module, £180, the lot, £1000. ☎ Paul 0191 290 1039.

ENSONIQ ASR10R sampler 10Mb internal effects etc, £1250; Waldorf Pulse analogue module, £425; Tropez soundcard, £100; CD ROM drive, £125. All as new. ☎ Steve 01273 776273.

ENSONIQ EPS16+ rackmount sampler, 24-bit effects and sequencer vgc, two available, £750 each ono; Roland S50 sampler with monitor, library, sequencer, vgc, £550 ono. ☎ 0115 962 4369.

ROLAND MS1 sampler in perfect condition with manual, PSU, etc, stereo samples, 18 to 68 seconds sampling time, £190. ☎ 01933 678608.

ROLAND W30 sampler workstation as used by Prodigy, Eat Static, JX, M Beat etc, excellent condition, boxed with manuals, 8 outputs, graphical editing and sequencer. ☎ Ben 01280 814385.

ROLAND S330 digital sampler, mint condition, 1U, fully expanded memory, £500. ☎ Guy 01903 211962.

ROLAND S760 sampler with OP760 video board plus SC-1224 Atari colour monitor, 2Mb memory, as new, 1 year old perfect working condition, £1550. ☎ 0181 981 4770.

ROLAND S760 10Mb boxed, as new, £965; Mackie CR1604 desk mint condition, £500; Roland W30 good condition, £650, all very cheap money. ☎ Martin 0976 403119, 0115 952 9114.

ROLAND S760 sampler, 16Mb of memory, CD ROM drive and five Roland CDs with all manuals and cables in mint condition, sensible offers please. ☎ 01924 478155.

YAMAHA SU10 sampler, new unwanted, price, £225. ☎ Tim 01905 426382 after 6pm (Midlands).

COMPUTERS/SOFTWARE

AMIGA A4000 with monitor, comes with bars and pipes, sequencer with 48 channel interface, plus full development software and manuals, £650. ☎ Andy 0117 975 4107.

486 PCI MOTHER BOARD 83MHz Pentium overdrive, two 4Mb SIMMs, ISA video card, £200, will split. ☎ Steve 01422 350281.

APPLE MAC SE30 all in one mac with monitor and clrisworks, £280; Notator logic music software, £150, both together for ideal mac music system, £400. ☎ 0181 302 5849.

APPLE MAC IICI 16Mb RAM, Audiomedia II card, SoundTools, ideal for digital editing, £750. ☎ Mark 01246 570781, 0976 771204.

ATARI 4MB STE Breakthru V2 sequencer, £200; Atari 520 STFM sequencer, Sequencer One, £50; Yamaha PSS 51 MIDI keyboard, £130; Yamaha RX8 drum machine, no manual, hence, £70. ☎ Robin 01305 785675 after 6pm.

ATARI 520ST 4Mb, 270Mb, SM124 hi-res monitor, Forget-Me-

Clock, basic ROM, Protext Protect, Lattice C++, Devpac, music software, 200+ discs, numerous books/mags, £500; HP520 inkjet b/w printer, £140. ☎ Jim 0191 536 2165.

ATARI 520 STFM, Sequencer One, £50; Yamaha DSS51 MIDI Keyboard, £130; Casio CZ230S, £50; Roland CA30 arranger for MT32 and CM modules, £40; Sansui MR6, needs attention, £100. ☎ Robin 01305 785675.

ATARI 520 STFM upgraded to 1Mb, can be expanded to 4Mb with ordinary SIMMs, with mouse, good condition, £130, Atari hi res mono monitor, £80. ☎ 01354 695239.

ATARI 520 STE with 1Mb RAM, can be easily upgraded to 4Mb with ordinary SIMMs, with mouse, good condition, £140; Atari hi res mono monitor, £80. ☎ 01354 695239.

ATARI 1040STE SM124 monitor, Creator and Band-In-A-Box, £250. ☎ Alan 01273 870179.

ATARI MEGA4 ST SM144 14-inch mono monitor, 500Mb hard drive, Digidesign Sound Tools system, boxed, manuals, excellent condition, genuine reason for sale, £695 ono. ☎ Matt 01629 636572.

ATARI STE 4Mb RAM hi-res monitor, 24 pin printer, Pro 24, £195. ☎ 01462 851169 (Beds).

ATARI STE studio sequencing setup, complete with leads accessories etc, 4Mb and high resolution monitor, excellent condition, was used with Notator editors M1 etc, £200. ☎ 01884 257487.

ATARI FALCON 030 240mb HD, external SCSI, Cubase 16 Audio, FDI, Time Lock, dongle, home use, £1200. ☎ Matt 01702 341983.

ATARI FALCON 4Mb, audio mods, £350; 270Mb SCSI-2 drive, £75; colour monitor, £75; Cubase Audio 16, £400; FDI, £80; Clarity 16 sample software, defrag, NVDI, etc. ☎ Duncan 0141 445 2415.

ATARI FALCON 14Mb RAM, 350Mb internal hard disk, £650; Cubase Audio 16, £450; FDI digital I/O, £150; 540Mb SCSI-2 hard drive, £175. Will consider P/X the lot, £1300. ☎ 01702 616961.

ATARI TT030 6Mb RAM, TOS 3.06, recently serviced, excellent condition, £500. ☎ Paul 01527 500873.

ATARI 1040STFM with 40Mb hard disk, hardware PC emulator and lots of software including Trackman and Sequencer One, £150 ono. ☎ 01883 347943 (Surrey).

C-LAB FALCON MK2 16Mb, 540Mb hard drive, Steinberg Cubase Audio 16, offers, swap for Akai ASR10 sampler. ☎ 01426 208798.

CUBASE AUDIO for Apple Macintosh, original manuals and discs, bargain at, £300 ono. ☎ Jay 0171 426 0340.

CUBASE AUDIO Mac version, mint easily upgrade for power- PC never been used, £400 ovno. ☎ 0161 721 4516, 0161 764 3041.

DIGIDESIGN SESSION 2.0 powerful multitrack recording/editing for audio media cards, power Macs automation EQ,SR convert etc, boxed as new, £95. ☎ 0161 439 0936.

DIGIDESIGN PRO TOOLS III eight balanced XLR inputs and outputs, AES/EBU and S/PDIF digital interfaces. Apple Power Mac

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7100/80AV, 32Mb RAM, CD, 17-inch monitor, Cubase Audio XT/VST Recycle, Sound Designer II, £5595. ☎ 0181 883 4329, 0956 339577.

OPCODE STUDIO 3 MIDI interface for Mac, 1U rackmount 2 in/6 out for 32 MIDI channel support. Also has SMPTE read/write if needed. All cables, disks etc. mint, £150. ☎ Andy 0181 524 0224.

PENTIUM 100 PC, Tropez soundcard, modem, £245; Roland D110, £245; Yamaha YFL2215 flute, £210 AKG D190E mic, £85. ☎ Edward 01707 850347 (Potters Bar).

REPLAY 16 SAMPLER for Atari ST, original disk, hardware and manuals 16-bit sampling on the cheap, £50. ☎ Mark 01772 888341.

ROLAND SMPU MIDI interface two in, two out, boxed, with Cakewalk V3.0, £75; Roland SCC1 soundcard, with Cakewalk, £115. ☎ Nik 01482 844556.

STEINBERG CUBASE V3 for Atari, registered includes manuals and dongle; NVDI 2.5; Kobold, TOS 2, £250; Yamaha REV 7, pro effects unit, £500; QED speaker switcher, £15; SIMMS, 60ns 72 pin, 2 x 16mb, £85 each, never used ☎ 0181 902 9784.

STEINBERG CUBASE AUDIO XT, £400; Band In A Box, £45; Sound Edit 16, £95. All for Mac, all ono. ☎ Steve 081 301 0777.

STEINBERG AVALON 16 bit D/A stereo converter plus steinberg avalon V2.1, universal sample editor software for Atari ST/STE computers with dongle, second manual, cost, £700 sell for, £295. ☎ 0181 883 4329, 0956 339577.

STEINBERG SYNTHWORKS Atari editor/librarian software for Roland D-series, £60; for Kawai K1/K1 MkII, £60, both for £100. Dongles and manuals included. ☎ 0181 883 4329, 0956 339577.

DRUM MACHINES

AKAI MPC60 sampling drum machine/sequencer, full memory, flight case, vgc, £995. ☎ Jon 01255 436065.

BOSS DR660 classic beatbox 255 sounds, reverb, chorus, 808/909 sounds etc, excellent condition with manual and PSU, £220. ☎ 01933 678608.

BOSS DR660 £260 ono; Akai VE8 module + 2 cards, £150 ono. Both vgc. ☎ Shaun 01273 690064 (Brighton).

CASIO RZ1 sampling drum machine, £100; Fostex X30 4-track, £100; Phillips CM8833 colour monitor with Atari lead, £100. All ono. ☎ Ian 01903 208883 days (W Sussex).

CHEETAH PAD TO MIDI converter, eight inputs, programmable, £100; C-lab Creator for Atari, manual, dongle etc, £200 ☎ Tony 0181 391 4201.

ROLAND TR626 rhythm composer, Pattern based drum machine with large LCD screen. Easy to use, excellent condition, almost as new, with manuals, £75. ☎ 01252 878565.

ROLAND TD7 percussion sound module,very good condition, manual, boxed, receipt, trigger happy, nine inputs, £450. ☎ Dave 01865 432712.

ROLAND OCTAPAD manual, PSU, APC66 clamp, mint, £225; Portman parallel 1x1 MIDI

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interface, £60; Studiomaster 16:2 Diamond mixer, £150; Goldstar GMK49 4 octave master keyboard. ☎ Paul 01527 853761.

SEQUENCERS

AKAI ASQ10 sequencer, ROM chips updated, will record all 16 tracks at once, best sequencer made, excellent condition, £525, complete with flight case and manual. ☎ Dave 01543 454253, 0973 540571.

ALESIS DATADISK two, with latest software £300 each; Fostex 160 4-track, recently serviced, £150. ☎ Jeff 01242 527179. .

ALESIS MMT8 sequencer, boxed, like new, manuals black colour, hardly used, £135. ☎ Felix 0181 668 6118.

KAWAI Q80 sequencer, 32-track and disk drive, plus K1 library, £200 ono; Casio FZ10M pro sampler plus disks, £450 ono. Both boxed, excellent condition. ☎ 01705 791701.

ROLAND MC50 MKII sequencer boxed with manuals, immaculate, £400. ☎ Mike 01367 240799 (Oxon).

ROLAND MC202 customised, control filter, etc, over MIDI, includes Kenton Pro 2, all original packaging, offers around £480, swaps; Fatar Studio 900 immaculate, £395. ☎ Paul 01482 443708.

ROLAND MC202 mint condition, manuals, £300; SH101 mod grip, £30; Technox rack, mint, boxed, £575, swap for JV1080 and will add cash, Novation bass station rack, boxed, £300, swap for drum station and will add cash. ☎ Phil 0113 263 1026.

ROLAND MC303 boxed as new, £500; Yamaha SY55 synth, hard flight case, internal sequencer, £350; Passport Mastertracks Pro V4.9, £50. ☎ Steve 0115 947 0384 eves.

ROLAND MC303 groovebox, absolutely wicked, speaks for itself, £500, consider part-ex for Emu orbit. ☎ Dave 01543 682095 after 2.30pm.

ROLAND MC303 boxed with manual, one month old, integrating gadget, this! offers around, £475. ☎ Paul 01732 455790, 0410 077759 (Sevenoaks).

ROLAND PMA5 music sequencer and arranger, £350 ono. ☎ 01734 882079 Reading.

ROLAND TB303 excellent condition in box carry case, manual, professionally fitted by Kenton, £800, will swap/PX for Akai MPC60 MKII, MPC3000; Ensoniq EPS with disks and manuals, £300. ☎ 0161 877 8096.

YAMAHA QY20 a sequencer/tone module/drum machine combined. Boxed with manuals and carry case, perfect condition, £220. ☎ John 01482 587432.

PERSONNEL

BAND REQUIRE lead and rhythm guitarists to perform all original material, style: modern pop/rock. Wilts/Hants/Dorset area. All own transport. ☎ Paula 01722 502100, 0850 762620.

DANCE/TECHNO PROGRAMMER looking for co-writer to release labels and maybe start a record label. Southampton area. ☎ Dustin

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01703 570412.

DRUMMY BASS BOYS producing tunes and looking for others to add. If you've got the talent then give us a ring. ☎ 0956 614808.

EXCEPTIONAL TECHNO producer seeks same for collaborations. Style: Steve Stoll, Robert Armani, please call only if you have produced a record in the above style. ☎ Baz 0181 539 2140.

FEMALE SINGER songwriter, keyboardist into EBTG/St Germain seeks innovative, rhythmical, Cubase, technical wizard for song collaboration to record/gig. ☎ 01483 825880.

GOTHAM CITY SOUNDTRACK Talented player/programmer wanted for project. Must like film music! other influences include Momus, Hannon, Walker. ☎ 01745 832134.

GOOD QUALITY new age style music for new label. Please send demos (all tapes returned) to Brian Marshall, SGM Music, Sutton Court, Tenbury Wells, Worcs, WR15 8RT or ☎ 01584 811653.

IS THERE ANYONE willing to collaborate with me, I have an 808, 101, MMT8, and Roland R8 MkII, I'm into trance. ☎ 01444 871260.

PROGRAMMER WRITER available for work, specialise in soul & r'n'b. Dedicated and hardworking. Feamal singer/songwriter also wanted, must be dedicated. ☎ Mark 01903 716580 eves, 0973 128652.

NEW LABEL looking for 'muscular' house/techno for 12-inch vinyl release, send tape/info to Sonia, PO Box 27, York YO2 1ZR.

PROGRAMMER collaborator possible business partner required by 2 musicians, own home studio currently underused, due to other commitments. Limited experience acceptable, enthusiasm essential. ☎ 0181 740 9663.

QUALITY DANCE hip hop/rock guitar crossover tracks required by library music publishing company. PRS members only. Demos to 101 Tavistock Drive, Nottingham NG3 5BE.

QUALITY HOUSE/TECHNO Tracks required for release in UK/Export. Serious potential for the right tunes. Send demos/whites to: MAGMA, PO Box HP120, Leeds, LS6 3TF.

TRAINEE SOUND ENGINEER or producer position required. Work experience includes remix work and own composition releases for up and coming dance label. ☎ Dan 01252 628694 (Hampshire).

TRANCEMISSION RECORDS requires innovative talent. All dance styles considered, submit your demo and info to: Trancemission Records, P O Box 4314, 6202 VA Maastricht, The Netherlands (Please enclose 1 international reply coupon for sure reply).

YAMAHA QY700 OWNER seeks another with work experience to help understand and possibly to collaborate composition. ☎ 01909 566695 after 6pm.

MISCELLANEOUS

EFFECTS PEDALS five, by Boss, Ibanez etc, £75; Casio VZ8M module with editor/librarian. All immaculate condition. ☎ John 01225 723513 (Bath).

HH IC100 AMP twin channel reverb, excellent condition, £170. ☎ Hatfield 01707 263734.

ROLAND GR707 guitar synth in very good condition, £400. ☎ Andy 01222 493528.

ROLAND GR707 guitar synth controller in bits, complete except for the two pickups, ideal for spares project etc with hard case, £120. ☎ Andy 01222 493528.

ROLAND JV/XP expansion boards, Vintage Orchestral, Dance, Session, £150 each ono. ☎ 01734 882079 (Reading).

YAMAHA ADP1 DX7 cartridge adaptor for DX7 MkII, £10; RAM 4 cartridge, £20 ☎ 0181 675 0651(Clapham).

WANTED

ALESIS QUADRAVERB+ will pay £150 cash. ☎ 01229 466238.

A1 APC01 guitar card; A2 SPC-07 Nashville card. ☎ Wayne 01782 205842 (Staffs).

AKAI MPC60II drum machine, cash waiting. ☎ 0181 883 4329.

BARS AND PIPES PRO V2 for Amiga 600, any other software to work with MIDI. ☎ Matthew 01482 352605.

ELKA RHAPSODY strings/piano in full working order up to, £100 dependent on condition; Roland TR707/TR727 required in good condition up to, £75 each. ☎ 01952 260064 (Telford).

EMU E64 SAMPLER wanted in immaculate condition, can pay around, £1600-ish. ☎ Ian 01432 269494 (Hereford).

HELP! if there is anyone out there who can teach a 14-year old student how to use digital equipment e.g mixers and similar stuff. ☎ 01484 513961.(W Yorks).

HELP! I need M-any platform. I'm desperate. ☎ Dan 01926 314235.

IS THERE ANYONE out there who owns a studio and is willing to let me have a day there so I can see what sound engineering involves? ☎ 01484 513961.

KURZWEIL K2000 wanted, can pay, £900-ish; SH101 wanted, can pay, £175. ☎ Ian 01432 269494.

LEXICON LXP1 reverb, cash waiting. ☎ Mark 0171 585 1814.

MOOG MINIMOOG circuit diagram so that I can attempt to repair my tight git friend's sick Mini, can you help?. ☎ Eric 01705 215980.

RACKMOUNT ears for Kurzweil K1000 series module. ☎ 01638 720090 eves.

ROLAND TR808 909, TB303, Yamaha DX100, No silly prices, dealers. ☎ 0113 263 1026.

ROLAND PK5 dynamic MIDI pedal board, Elka self-contained MIDI unit acceptable; Falcon 14Mb and internal H/Drive; TOA KD3 keyboard combo. ☎ Robin 01522 752458 eves.

TR606/TR707/TR727 £75; SH101, MC202; Wasp, £150; TR808, £300; TB303, £450; Oscar Jupiter, £500; TR909, Jupiter 8, Prophet 5, Minimoog, £600. ☎ Kevin 01353 663613.

WANTED SH101, MC202, Basstation, Matrix 1000, K1, D10, Quadraverb, SE50 can pay, £150; Juno 106, M1, Vintage Keys, £350; TB303, SY85, 1604, W30, S1000, £500 Cash waiting. ☎ 0973 123965/0181 699 6439.

WANTED: A COMPLETE STUDIO to buy, will take individual bits for home setup, Roland, Korg, any Sequential, Yamaha etc cash waiting for good offers. ☎ 01252 375025.

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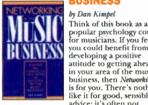
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By Rob Young
This book comes from Prentice Hall and aims to help you get the most out of your MIDI system, by providing loads of tips on programming realistic instrument and rhythm parts, using sequencers to create advanced effects. Basic MIDI concepts are covered, along with advice on setting up a MIDI-based home studio. More advanced discussions cover specific types of MIDI messages and what you can achieve with them. All the techniques in the book are illustrated on a supplied floppy disk.

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MAKING THE MOST OF MIDI

By Paul Overaa
This is a MIDI book unlike any other. It explains the fundamentals of MIDI and MIDI sequencing for the beginner, and it goes much further, dealing with issues that will be of help to more established users. Advice on MIDI fault finding, help for musicians using sequencers with live bands, MIDI messages and MIDI files, plus programming material. So if you are a keen first time user or a MIDI veteran, there will be something in this book to interest you.

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SEQUENCING

by Dave Clackett
Hands On MIDI Software's Dave Clackett has boiled his knowledge into a collection of tips for getting the best out of hardware or software sequencers. Dave covers the hardware and software you need to get sequencing, and introduces the basics of sequencing and notation, with particular emphasis placed on using General MIDI/GX/XG sound sources. Recommended.

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MIDI FOR THE PROFESSIONAL

by Paul D Lehrman and Tim Tully
Co-written by SOS contributor Paul D Lehrman, this substantial, exhaustive work covers pretty well any aspect of MIDI that you could think of. As an overview of what MIDI is and does in 1994, this 239 page book couldn't be better.

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by R A Penfold
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The new Musicians' Directory from Arc Publishing aims to provide all things to all musicians: whatever your style of music, the comprehensive listings offer you the contacts you need, whether it's labels, studios, venues, music shops or management - and that just scratches the surface of the dozens of topics covered by the book. Apart from being a directory proper, the Handbook also includes a selection of useful articles, including subjects such as how to be a session player (by SOS's very own Paul White), music publishing, releasing your own record, copyright protection, accounting, legal advice and more. The Musicians' Directory offers over 380 information-packed pages for little more the price of a new CD.

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SYNTHESIZERS

KEYFAX 5

by Julian Colbeck
The latest instalment in the intrepid Keyfax series offers potted specifications and concise mini-overviews of digital pianos, home keyboards, General MIDI modules, stage organs and computer sound cards. Look out for company backgrounders, contact details and cross-references to earlier Keyfaxes.

CODE B231 **£12.95**
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KORG X5 SYNTH

Korg's new X5 synthesizer, reviewed in SOS' January 1995 issue, now has its own video manual. Presented by Tim Walter in an entertaining and lighthearted manner, and explains all basic aspects of Korg's highly affordable instrument in very clear terms. If you're in any way daunted by your new purchase, or want to get a feel for the instument before laying out your cash, this is the video for you.

CODE V027   **£19.99**
Running time 55 minutes
Postage: UK £1.75, Europe £4.95, R.O.W. £6.95

PLAY BLUES NOW!

A complete guide to blues guitar playing that shows a variety of blues rhythm and soloing styles — for both beginners and advanced players alike.

Learn the blues style of Eric Clapton, Stevie Ray Vaughan, Peter Green, etc. The programme not only covers the usual blues shuffles and scales but using 4 complete songs, gives you the resources and techniques to create professional blues parts and solos in a wide variety of blues styles.

CODE V051   **£12.99**
Running time approx 50 minutes
Postage: UK £1.75, Europe £4.95, R.O.W. £6.95

ROLAND JV-90 SYNTH

Roland's JV-90 synthesizer gets the treatment in this informative Labyrinth production. Virtually every facet of the JV-90 is covered, from the basic functions to more advanced programming. Chris Allen guides you through the powerful features this instrument has to offer. Produced in association with Roland UK & USA to ensure that everything it contains is accurate, and relevant to new and more experienced users alike.

CODE V044   **£19.99**
Running time 70 minutes
Postage: UK £1.75, Europe £4.95, R.O.W. £6.95

THE EMAGIC LOGIC TUTORIAL VIDEO VOLUME 1

Produced by Emagic themselves, this video once again sees Tim Walter featured on the excellent Notator video) in the tutor's chair. There are plans for a series of videos which become ever more detailed and informative, but for now Volume 1 takes you through the first steps of getting the software up and running, and covers virtually all the controls you'll need. *This video is valid for all versions of Logic, whether being run on an ST, Mac or PC.* Presentation is rather intriguing, with a mobile camera that helps to hold the interest more than the average training video and some interesting graphics that aid comprehension and help to quickly find specific tutorials and bits inside tutorials. Contents as follows:
• Tutorial 1: The Arrange window and basic sequencing
• Tutorial 2: Playback parameters & Toolbox
• Tutorial 3: Controlling MIDI data flow
• Tutorial 4: Manipulating sequence data
• Tutorial 5: Moving around in Logic
• Tutorial 6: Windows and key commands
• Tutorial 7: More sequencing
• Tutorial 8: Score
• Tutorial 9: Looking at the Event List
• Tutorial 10: Editing the Event List
• Tutorial 11: Using the Environment
• Tutorial 12: The best of the rest (HyperEdit, Matrix Editor, Cycle Mode and more).
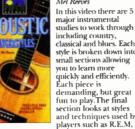
CODE V023   **£29.99**
Total running time 75 minutes Format: VHS (PAL)
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ACOUSTIC FINGERSTYLES

Presented by Mel Reeves
In this video there are 5 major instrumental studies to work through including country, classical and blues. Each style is broken down into small sections allowing you to learn more quickly and efficiently. Each piece is demanding, but great fun to play. The final section looks at styles and techniques used by players such as R.E.M, The Eagles, Sheryl Crow and Paul Simon, and provides an insight into creating new and interesting fingerstyles of your own.

CODE V059   **12.99**
Running time approx 90 minutes
Postage: UK £1.75, Europe £4.95, R.O.W. £6.95

KORG X3

This video offers an easy way in to this powerful workstation. It starts from absolute basics, including audio connections and volume levels, and the clear and helpful script covers all basic facilities of the X3. The video features a sophisticated use of picture in picture, with the X3's display cut into the main image to show you clearly what Phil is actually doing, and main points are reinforced with on screen text and graphics. The main section headings are:
• X3 Audio Connections • Getting Around the X3
• Factory Disk • Disk Drive Modes
• Selecting Sounds • Global Modes • Sequencer Mode • Quick Sound Editing
• Playing MIDI Files on the X3 • Using the X3 with an external sequencer
Note that these are loose headings, with each section also containing information on connected subjects in varying amounts of detail. A lot of ground is covered, and we can recommend the tape to any X3 owner.

CODE V018   **£19.99**
Running time: 55 minutes Format: VHS (PAL)
Postage: UK £1.75, Europe £4.95, R.O.W. £6.95

PLAY HEAVY METAL NOW!

A concise, easy to follow video guide to real heavy metal playing for guitarists of all levels.
A guide to hard rock/heavy metal techniques for guitarists of all levels. The programme covers RHYTHM, SOLOING, and TECHNIQUE. All of the techniques, solos, etc are clearly demonstrated. No need to read music. Free booklet contains tab, chord and scale diagrams.

CODE V052   **£12.99**
Running time approx 65 minutes
Postage: UK £1.75, Europe £4.95, R.O.W. £6.95

PLAY GUITAR NOW! ACOUSTIC & ELECTRIC

An easy-to-follow video to take you from the basics to a full rock solo.
The programme covers TUNING, CHORDS, FINGERSTYLE, 12 BAR BLUES, BARRE CHORDS and ROCK SOLOING for both acoustic and electric guitars. Includes free chord/tab booklet.
It is not necessary to be able to read music to learn from this video.

CODE V053   **£12.99**
Running time approx 51 minutes
Postage: UK £1.75, Europe £4.95, R.O.W. £6.95

ACOUSTIC BLUES GUITAR

Presented by Mel Reeves
Whether you can already play fingerstyle guitar, or are simply familiar with basic chords, this programme will show you all of the necessary techniques to play great blues guitar. All of the technical aspects are explained in the clearest possible way before getting down to the real business of learning 5 real blues that are great to listen to and terrific to play.

CODE V058   **12.99**
Running time approx 63 minutes
Postage: UK £1.75, Europe £4.95, R.O.W. £6.95

ANALOGUE HEAVEN

Remember our feature on the opening day of the Museum of Synthesizer Technology which we ran back in October 1994? Wish you'd actually been able to attend?
Well, now you can have the next best thing: the museum has released a fascinating 50 minute promo video that features footage shot on the day as well as a lightning tour of the facility. The programme is hosted by SOS contributor Julian Colbeck. The tape opens with synth pioneer Bob Moog making the inaugural speech and officially opening the museum, followed by a nostalgic run through one of his old modular systems. Analogue Heaven makes diverting viewing for anyone interested in vintage synths, represents good value, and offers the perfect companion to the Museum's guide book, also available from the SOS Bookshop.

CODE V026   **£12.95**
Running time: 50 minutes Format: VHS (PAL)
Postage: UK £1.75, Europe £4.95, R.O.W. £6.95

GETTING THE MOST OUT OF THE KORG M1

Written and presented by Julian Colbeck
The Korg M1 is a classic instrument. The range and quality of its sounds has made it one of the most successful synthesizers of all time. But because the M1 is so immediate, it's too easy just to scratch the surface, leaving many of its exciting capabilities undiscovered. This highly informative video enables M1 owners to unleash the full creative potential of this enormously powerful workstation. With plenty of musical examples, tips and inside information, the video offers clear, concise explanations of the instruments features and operating procedures, helping you to greatly expand your dexterity and creativity across he whole range of this world-beating instrument.

CODE V002   **£24.95**
Running time: 1 hour 10 minutes Format: VHS (PAL)
Postage: UK £1.75, Europe £4.95, R.O.W. £6.95

YAMAHA QY20 MUSIC SEQUENCER VIDEO MANUAL

by Tom Robinson
Yamaha's popular QY20 portable sequencer/ sound source is a deceptively simple device to use; if you think you might like some insight into getting that little bit extra out of the device, then look no further. Produced in co-operation with Yamaha, the video features mobile musician Tom Robinson showing you how to get the best from this portable marvel. Tom, who has used the QY20 extensively as a writing tool on the road, says: "I never leave home without it".

CODE V016   **£19.99**
Running time: 90 minutes Format: VHS (PAL)
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PLAY KEYBOARDS NOW!

Presented by Mel Reeves
An up-to-date video/booklet that gets to the heart of modern keyboard playing with no need to read music!
All you need to be able to create pop, rock and even jazz chord accompaniments using major, minor, seventh, sus 4th and added 9th chords. Learn arpeggios, random arpeggios, chord 'splitting', crushed tones, boogie blues, inversions and much more.
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COMPLETE ELECTRIC GUITAR

Presented by Mel Reeves
Learn how to create solos and rhythm parts in any style! The programme covers ROCK 'n' ROLL, BLUES, JAZZ, COUNTRY, ROCK and FUSION and much, much more. Plus the clearest possible guide to chord and scale construction, tonal centres and working out parts for yourself. Includes free construction booklet.

CODE V050   **£12.99**
Running time approx 65 minutes
Postage: UK £1.75, Europe £4.95, R.O.W. £6.95

COMPLETE ACOUSTIC GUITAR

Presented by Mel Reeves
Whether you only know a few chords, or are already proficient, this easy-to-follow programme will show you all of the main guitar styles, as performed on the acoustic guitar. As well as learning tips and 'tricks of the trade' Mel Reeves will show you 6 great instruments, step-by-step and take your playing to new heights. Plus, open tuning, the capo, high-strung guitar, reading music and more!
Styles include: Guns 'n' Roses, Bon Jovi, Hendrix Blues, Mississippi Blues, Django Style Jazz, Nuits En Moscow, Classical, Romance, Carter Style, Buffalo Gals and Bottleneck/Slide Style.

CODE V056   **£12.99**
Running time approx 78 minutes
Postage: UK £1.75, Europe £4.95, R.O.W. £6.95

PLAY KEYBOARDS NOW! MELODIES

Presented by Mel Reeves
This programme will not only teach you note reading, rhythm, rests, ties and how to move up and down the keyboard with ease, but will also teach you 6 complete melodies familiar to lovers of rock, pop and classical music.
In addition, there is a further look at improvising, making use of melody and scales combined.
Clear computer graphics and the booklet enclosed will mean you won't miss a thing.

CODE V057   **12.99**
Running time approx 55 minutes
Postage: UK £1.75, Europe £4.95, R.O.W. £6.95

START HOME RECORDING NOW!

This is the ideal volume one guide to the use of home recording equipment. From the set up of basic 4-track cassette machines to sub-mixing Mel Reeves uses straight-forward language to talk to fellow musicians about the methods used to get the sounds in their heads onto tape. Includes free instruction booklet.

CODE V055   **£12.99**
Running time approx 55 minutes
Postage: UK £1.75, Europe £4.95, R.O.W. £6.95

YAMAHA QY300 VIDEO MANUAL

Complete overview of Yamaha's new QY300 sequencer/sound source. Presented by Joe Ortiz of Heavenly Music, the video offers a hands-on tutorial on what is potentially a very powerful piece of hardware. Starts off with a runthrough of QY300 features, followed by a clear, step-by-step tutorial that introduces you to creating your own sequences and styles. As a bonus, the video comes with a free disk loaded with 6 new QY300 styles and 6 demo sequences

CODE V025   **£19.99**
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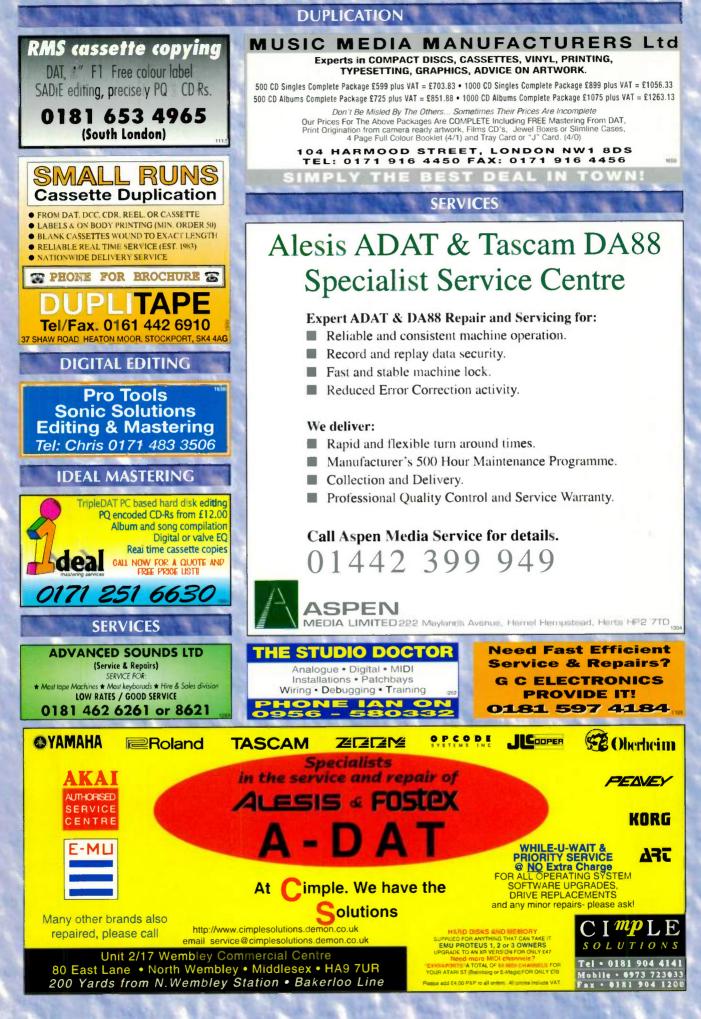
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Sounding Off

ALAN FISH-BARNES asks "What's in a name?" and concludes: "Quite a lot, actually..."

SX-WSA1R, N364, 01/W FD, VFX-SD2, K2500... If you're new to SOS and the world of hi-tech music-making, you'd be forgiven for wondering what this list of strange algebraic terms is doing on the pages of a magazine for musicians. More seasoned readers should be aware that this rather mathematical expression would be sufficient to describe a respectable collection of synths from recent years, even if they probably wouldn't be able to remember the correct positions of all the dashes and slashes if pushed. Yes, these are all synth names, and there are plenty more where they came from.

What do they evoke for you? Do they make you think of the spotlit centre-stage at a packed-out venue? The electric atmosphere of a pro studio in the white heat of creation? The hopes and dreams of the bedroom musician expressing his or her art through the sublime sounds issuing from their instrument? Or do they rather put you in mind of the model number emblazoned on the side of your microwave oven or the back of your TV? The top-secret designation for the latest CIA spy plane? A page torn out of a cryptographer's handbook?

Any reader older than about 35 should be able to remember a time when it was possible to own an entire multi-synth setup without having to remember a single arcane string of numerals. Instead of having a list of numbers disposed all over your three-tier A-frame, it could be graced by the likes of Jupiters, Odysseys, Tridents, Prophets and Alpha Junos — names which are a pleasure to say and hear, drawn from mythology and legends, with truly cosmic connotations. In 50 years, which would *you* remember best — the names or the numbers?

Recall the cheeky Rogue (named by Bob Moog so that people might get a clue about how to pronounce his unusual name — the two should rhyme). Remember the quirky Wasp monosynth, bravely named by British designers daring you to compare the sound of the two. What about its siblings, the Gnat and the Spider (the latter a simple analogue sequencer)? And let's not forget the unsurpassably cutely-named Octave Kat and Kitten, the grandly ambitious Moog Liberation, the colourfully monikered Rhodes Chroma, the aggressively hip ARP Axxe, or Korg's playfully designated Mono/Poly. It wasn't only synths which had names to conjure with: how about spicing up your sound with a Memory Man or an Electric Mistress (an early delay and a flanger pedal respectively!)? Today's number-laden labels keep us firmly earth-bound. They don't offer us any expectations beyond the spec sheet, unlike the magical names of old.

Before you all write in to point this out, it's true that synth manufacturers have always used numbers to designate their instruments, but at least they were often stylish (TR909, MC202, MS20) or instantly recalled their maker: everyone used to know that the prefix 'CS' meant a Yamaha synth, just as 'TR' meant a Roland drum machine. Now manufacturers are even running out of different alphanumeric designations: a few years ago an R8 was released simultaneously by two manufacturers — one a multitrack and one a drum machine — and a current DAT machine shares its name with a late '80s-vintage synth. A certain leading-edge synth rejoices in the same name as a tiny kiddy-keyboard released way back in 1981. Well, they do say we should recycle!

No doubt there are good reasons for the decline of the proper name: now that electronic music is a global business, names have to travel and work in all languages: what better than alphanumeric tags in this situation? And manufacturers probably don't have the manpower to devote someone to first inventing a witty or evocative name for each new product, then doing the necessary research to ensure that it can be used. Look at how much aggravation goes into naming a single new washing powder or hair conditioner: with their rapid development cycles and increasing number of products, can manufacturers really be expected to add to their costs by spending a disproportionate amount of time and effort on a name? Maybe not.

A few bright exceptions stand out at the moment: Korg's Prophecy and Trinity recall the majesty of the old days, Emu's Orbit synth and Launch Pad MIDI controller take the musician back into space with the Jupiter, and Quasimidi's Raven, Technox, Cyber 6 and forthcoming Rave-O-Lution leave you in no doubt as to what to expect from your new synth. And the tradition of inventive naming persists in software, with names like *Cubase*, *Logic*, *Cakewalk*, *Performer*, *Alchemy*, and *Vision*, amongst others. Let's see this healthy trend continue — they are not numbers: they are free instruments! **SOS**