First word

Greg Simmons

Lessons in coupling and isolation

"Yikes, those Questeds are making my ATCs sound bad!" I cried. Brad was somewhere behind the console, fossicking for test CDs in a dirty blue milk crate. "Ha!" he laughed, "I dare you to print that!". "No, no," I explained, "my ATCs don't sound right when those Questeds are sitting next to them... the low mids have gone all lumpy!"

Brad and I had just unpacked the Quested VS2108s and, noting their size, decided to sit them flush beside my ATCs for a quick and easy A/B comparison. But at 34cm wide, the Questeds' imposing baffles were playing havoc with the dispersion of the considerably slimmer ATCs, coupling the lower frequencies and making them sound dull and ordinary.

Interesting? Not really. It's simple physics, and it's something that PA operators deal with whenever they stack multiple boxes side by side. But it's been about 15 years since I stacked a PA system, and acoustic coupling was the last thing on my mind as we lined up the Questeds beside the ATCs. (I can see Jands' Nick Orsatti shaking his head in disbelief as he reads this…)

Switching between the ATCs and Questeds showed that neither pair were sounding worthy of their respective price tags. Taking the Questeds off the bench returned the familiar ATC sound. Likewise, removing the ATCs and listening to the Questeds in isolation revealed a much higher level of sound quality, certainly more in line with their reputation.

Lesson #1: Never compare monitors in any kind of side-by-side situation. It may seem like a good way to make a direct A/B comparison, but all you'll actually be testing is how badly each monitor is affected by the other one. Retailers, take note!

Later that week, during an AudioTechnology 'Spontaneous Human Consumption' event at Brad's place, Michael Stavrou spent a critical moment listening to the Questeds, rubbed his chin for another critical moment, then said, "You got any marbles?". "Dunno," said Brad, "take a look around." "How about washers?" Stav asked. "Ditto…"

You can dig up all kinds of interesting stuff while fossicking around Brad's place but his marbles and washers were too well hidden, so Stav returned with half a dozen beers. He carefully removed the top from each bottle, and placed three tops under each Quested in a triangular shape (one under each front corner, one half way across the back). The performance increase was obvious to all, and became a hot topic for the next half hour or so. Just long enough for the beers to go flat…

Lesson #2: Lifting a monitor's bottom off the surface it rests on minimises physical contact, thereby reducing the amount of sound energy being drained out of the monitor and into the surface. This 'draining' of energy out of the monitor causes a decrease in performance, but it gets worse: if the surface is not sufficiently well-damped, it will re-radiate that energy back into the room, causing an even further decrease in performance.

Brad's monitor bench is about six feet long, reasonably rigid, and supported at each end. But the Questeds have a large and squarish footprint which provides a good contact area with the bench, and they generate a lot of low frequency energy for their size. Combine these factors with their 22kg weight per box, and you've got a powerful source of low frequency energy with a large footprint and considerable mass pressing down onto the bench, allowing an even better draining of energy!

Interestingly, my ATCs are designed with three feet fitted in place for this very reason – and never suffered this problem when mounted on Brad's monitor bench. But why three feet? Why not four or more?

Lesson #3: Proper monitor performance requires stability. Powerful small monitors, such as the ATCs and Questeds, really need to be held stable. If the box wobbles or rocks in any way, it causes loss of output, blurring of the stereo image, and smearing of high frequency detail. JBL's Doug Button discussed this concept, which he calls 'inertial grounding', in my review of JBL's LSR32s [Vol. 1, Iss. 4.]

But why three feet? Three points defines a single plane, and therefore offers maximum stability – that's why microphone stands, camera tripods and my favourite 'non-rocking' café tables are all designed to stand on three feet. (Of course, 'tripod' literally translates to 'three feet'. Duh!). Increasing the number of feet beyond three increases the possibility of instability and wobbling – not a good thing for microphones, cameras, steaming hot cappuccinos or studio monitors!

While Stav's beer bottle tops demonstrated the benefits of isolation, they were only a temporary solution. Brad has since replaced them with height-adjustable brass cones designed specifically for decoupling speakers, which are available from your local hi-fi shop. Due to the squarish footprint and weight distribution of the Questeds, he's using four cones – one under each corner. Being height adjustable, he's able to fine tune them for maximum stability. His Questeds are now sounding better than ever.

So if your monitors are sitting flush on their bottoms, get some cones under them ASAP! You won't regret it. But make sure you put the cones the right way around – which is upside down! Their large flat end connects with the bottom of your monitor, while their small pointy end connects with the surface your monitor sits on. When done correctly, your monitors will look like they're standing on tip toes. (In fact, the first commercially available cones were called 'Tip Toes'!)

So how do the cones work? Physically, their pointed ends provide a solid connection between the monitor and the bench, which keeps the monitor from wobbling. But their small contact area with the bench creates a very high acoustic impedance, the sort of thing sound energy prefers not to travel through. With the weight of the monitors pressing down on them, the cones are able to firmly anchor the monitors to the bench while simultaneously providing acoustic isolation. Amazing, huh?

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Tutorials

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Competition: Win an Apogee Rosetta

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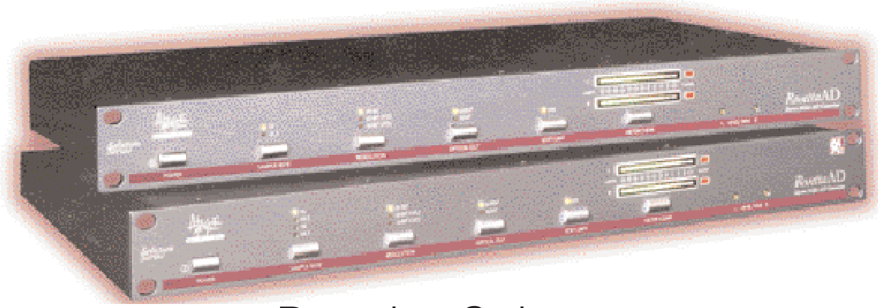
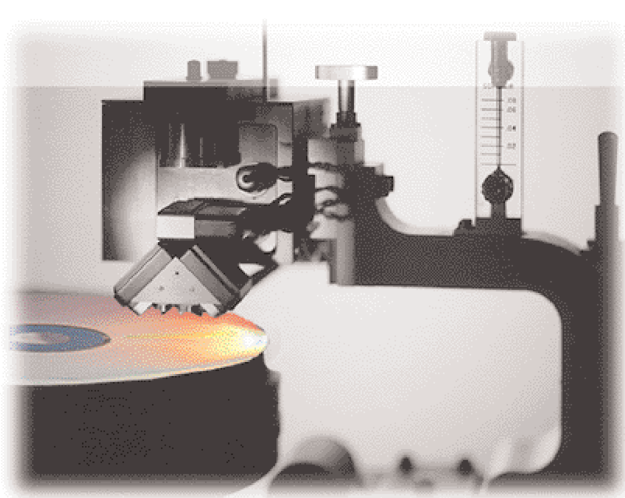


Equipment Tests

Regular Columns



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Second word

Christopher Holder

From ear to eternity

Recently I experienced one of life's great disarming moments, and it happened in the bathroom. Before your mind wanders where it shouldn't, let me elaborate. I was having a shave, as I am occasionally wont to do, and as my Gillette blade glided effortlessly over the last remaining stubble like a Lambourghini down an alpine road, it happened. Rather than slap on the aftershave, admire my demi-god looks, point at my reflection and growl, 'go get 'em Tiger', I stopped… and looked properly. In that one moment of sudden insight I noticed something that I'd never noticed before. My right ear was lower than my left! "No, never, can't be, my sideburns are different lengths," and so the desperate, and ultimately futile, explanations ensued. But, I couldn't escape the hideous truth, (and I know for anyone who knows me this will be hard to accept), I wasn't perfect.

That's a nice (if slightly unnerving) story, and here's another. I'm involved in the launch of a new independent record label. It's called SPL Records and we're currently releasing a compilation of some of the best Australian and UK big beat and electro tunes around, called *Impact* (that's not a shameless plug, I'm rather embarrassed actually). Initially we decided to scout for talent, and tap into the rich seam of undiscovered Aussie producers. After all, everyone knows there's a bedroom producer tapping away in every bedroom – word is, Jeff Kennett lost the Victorian election because he spent too much time noodling away on his new national anthem (Waltzing Jeff, I think he called it). After placing a few carefully worded advertisements Australia wide I was amazed at the response. Wretched. Most of the applicants were obviously very serious about their craft. All had CD burners and colour photo copiers, and many had decently equipped studios. Furthermore, many actually had their demo albums professionally mastered, a number from Australia's top mastering houses. The results (despite the best efforts of the top class mastering) were diabolical. Forget about the lack of imagination or the 'colouring by numbers' approach to using heavy distortion on vocals and drums, the mixing and recording was hopeless.

I'm not sure why this surprised me so much. We've got to the point where high quality home recording is well and truly possible – that much is obvious. There is an arguement for saying there is still another significant leap to be made, technologically, to get the polish and quality of a truly commercial product – that's fair enough. But it is possible. Josh Abrahams did it [read my interview on page 34], and I think two ARIAs this year is a legitimate indication of his success. Equally, I'm sure I'd be pleasantly surprised to hear the excellent results of many other home producers who have no desire for their songs to be heard outside their living room or circle of friends. But, I think my experience demonstrates a basic tenet that can often be overlooked in the perpetuation of the home studio dream – for professional results you need skill. It sounds obvious, but too often the equipment is portrayed to be the only means to an end. It isn't if you haven't the skill to use it. Which leads me to my next story.

Unhappy with the quality of the flow of happy hardcore, and dillusioned by the deluge of 'smiling death metal' (no, honestly) that I was receiving, SPL Records decided to approach the proven Australian performers, who could supply us with top quality tracks for the compilation on a licensed basis. One of the first ports of call was of course Thunk Recordings, who have over a number of years garnered a formidible reputation for world class dance music. I soon found myself in the home studio of Thunk impressario and regular AudioTechnology contributor, Brett Mitchell. He was running through a number of likely tracks and after one particularly good example of techno influenced break beat I inquired whether it was mixed in his studio on his Soundcraft Ghost. The answer flabbergasted me. Brett informed me that the track I just heard was from two years ago and mixed on a tiny 12:2 Samson console. Skill. During the bringing together of the other tracks for the album the skill factor became even more evident – and Australia was dripping with it. For example, there was one brilliant track that was entirely produced on nothing but an Akai sampler and a tiny mixer. But, if I had to pick one moment that shook my personal home studio world, it was that moment at Thunk Central. I went back to my studio and reassessed my approach. Which brings me back to my first story.

I've been looking at my face in the bathroom mirror every morning since I was big enough to jump onto the vanity unit, and never once did I ever question or notice the cock-eyed alignment of my ears, it took a moment of insight – rather than look, I truly observed. After my epiphany at Thunk, I realised that I'd grown lazy in my own recording endeavours and hadn't striven to push my own particular envelope for some time. I hadn't listened to a recent well produced album in my studio and tried to learn from it, I hadn't for some time tried to emulate an interesting effects process that I'd heard – the sorts of things that present me with a challenge that I can learn from.

Having the requisite skill takes time, and having the best equipment in the world is no replacement for that skill, but we can all improve our work. The best way to improve is by using our ears. Listen for where our own work is deficient and listen for where our favourite producer's work isn't – and I mean really listen. Meanwhile I've just realised why my sun glasses don't sit perfectly straight on my face.

Your word

Price ripoff

After reading glowing reviews in various magazines for the Focusrite Voice Factory and Drawmer MX-60 Voice Channel, and noting their low prices overseas (375 pounds for MX-60, 399 pounds for the Focusrite in the UK) I decided that a new purchase was in order. When I arrived at my local dealer, to my surprise I found the prices were $A1795 for the MX-60, and $1595 for the Focusrite Voice Factory. After doing a quick calculation and taking into account transport fees, duties, taxes… I determined that the MX-60 should be below $1199 and the Focusrite VF $1299. At least, that's what it would cost me to import one from the UK. Why are we being ripped off?

Name withheld

I remember the first time I got my hands on a US recording magazine and looked at the prices. Yes, they're very cheap and to compare them with Aussie prices is very disheartening. It would be easy and very gratifying to surmise that there are a few local fatcats making loads of money at our expense. Unfortunately it's more complicated that that.

It's worth noting that the two products you cite as examples are both UK companies, and in the UK their pricing is direct from the manufacturer to the dealer – there's no distributor (or 'importer') margin involved. But then, while I was working in the UK I can recall being thoroughly disgruntled with US product imported to the UK for the same reason – it always seemed to be far more expensive than it had any right to be. Australia isn't a big manufacturer of high tech recording gear and as such, we import the bulk of what we buy. And to import gear we need distributors who take their cut which inflates the price somewhat. Why do we need distributors? Well we don't… until we need some technical help or repair work done, that is. It is frustrating seeing what the US and Europe pay for their gear, but with their economies of scale and strong manufacturing sector it's understandable. In short we've just got to live with the prices, just like we have to live with the prices of imported cars, hi-fi, parma ham… Good new is that with the removal of the sales tax and imposition of the GST in Australia next year prices should drop. – CH

Studio makeover

Firstly, I would like to congratulate your team on a great publication. I've bought every issue so far and always enjoy reading the articles. I'm fairly new to the music scene, and that is why I'm so interested in reading your articles. They help me to better understand the concepts that go with working in this great industry.

At the moment I am still studying, but during my free time I pursue my interests in music and music recording. Being a musician I have already recorded my own CD (Majestic Worlds) in my project studio. After my studies I wish to open a proper recording studio, and I've already begun its design.

Because of this, my favourite article so far has been from Issue 3, 'Studio Makeover'. Reading about topics such as ways to reduce sound leakage and general studio setup is great! It really makes it easier for myself to become aware of how I'll one day be setting up my own studio.

Another aspect of your magazine I am always interested in is the vast amount of recording gear you review. Thanks for not cutting the product reviews too short, and thanks for generally displaying the prices – I can't stand it when I find a product I really like and it doesn't tell me how much it's going to cost. I look forward to reading your next issue, its great to see an all Australian magazine, keep up the great work.

Marly Luske
Canungra, QLD

Gee Thanks

I am currently setting up a Midi home studio on the Gold Coast, so I was pleasantly surprised to see AudioTechnology on the shelf of my news agent – something that certainly lives up to the standards of today's international magazines. It is not only a great magazine filled with all the info readers die to get there hands on, but it's also Australian which is beneficial to local readers like myself. My main subscription has been with Future Music which is another great magazine from overseas, which includes a CD as well (saying that I pay $16.95 for the airmailed copy over here). AudioTechnology is the only other magazine I have come across that is really what the readers want to see and is hard to beat as far as the profiles, reviews, and price. You certainly can't get a magazine of this quality for under 10 bucks these days! A cover mount CD would be appreciated though.

Damian Newell, Gold Coast, QLD

Thanks Damian for your kind words. Of course, we have looked at doing a CD. Although I don't see AudioTechnology having a CD on the cover of every issue, there are a couple of special one-offs planned. We want to be extra careful to offer something geniunely useful rather than a bi-monthly beer coaster! – CH

Got something to say? Want a rant?

Send your comments, observations, and criticisms to Your Word, AudioTechnology magazine, Suite 33, 84 Dee Why Parade, Dee Why NSW 2099.

As an added incentive, the best letter printed in each issue wins some equipment from M Audio courtesy of Electric Factory. This issue's winner is Marly Luske, who receives some great M Audio gear like you see pictured left. Future issue prizes include the Flying Calf 20-bit A/D convertor, and Mac and PC Midi/SMPTE interfaces.

Tascam MX-2424

Tascam has released a new heavyweight contender in the hard disk multitrack market. The MX-2424 features 24-track, 24-bit recording for under $10,000. The MX-2424 comes as standard with a 9GB internal hard drive, offering 50 minutes of 24-track, 24-bit recording. The I/O configuration is user defined with a variety of combinations of TDIF, ADAT optical or AES/EBU and analogue. The front panel 5.5-inch drive bay can be fitted with another hard drive, a removable Kingston drive, or backup DVD. The MX-2424 comes with an optional remote and Viewnet MX graphic user interface. Looks to be a very tasty bit of kit at the price.

Tascam has also updated their DA series of 8-track DTRS machines. The DA-78HR is effectively a 24-bit DA88 machine, while, similarly, the DA-98HR is effectively a 24-bit DA98 machine. The new gear is due out early 2000.

TEAC Australia: (03) 9644 2442

Digi go to square 001

ProTools v5.0 recently went out on the road spreading the Digidesign gospel. The new software had the crowds ooo'ing and aaah'ing as many lingering niggles were ironed out and new features shed new light on accepted ways of working. What caused most eyebrows to arch was the inclusion of Midi sequencing in the package. The Midi section is by no means comprehensive, and neither is it meant to be. The Midi integration is intended to be stripped down, offering everything that is essential and none of the arcane complexities often seen in the latest generation Midi+Audio sequencer packages, and to make it a 'natural extension of ProTools' audio features'.

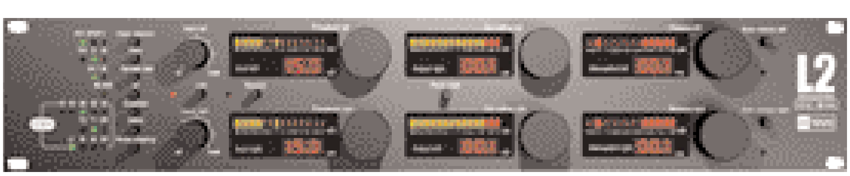
Also on tour was the entry level Digi 001. The Digi 001 is 'a complete audio/Midi, hardware/software solution in one box'. On the breakout box you'll find eight analogue inputs, eight analogue outputs, eight channels of ADAT optical I/O and two S/PDIF I/O channels – all at 24-bit resolution. The package comes with ProTools LE v5.0 Audio+Midi software and a PCI card. As you'd expect with Digidesign, there are plenty of third party plug-ins available. Digi 001 is retailing at $1699.

Digidesign: (03) 5428 7780

Waves L2 Master

Wave's first hardware unit is based on Waves L1 software, the 'most popular digital limiter in the world'. The L2 takes everything further, to 48-bit resolution, 96k support, ninth-order noise shaping IDR (increased digital resolution) dithering, and the world's cleanest brickwall limiter. Add to this Wave's 24-bit, 96 k A/D and D/A converters, and the L2 is an obvious choice for mixing, mastering and more. Hot levels, highest resolution, clean sound. 2U 19-inch rackmount. The L2 should be priced in the $4000 to $5000 region.

Sound Devices: (02) 9283 2077



Alesis Masterlink ML9600

More news on the hotly anticipated Masterlink ML9600 mixdown and mastering system from Alesis. it combines hard disk recording and editing, digital signal processing and CD creation in a single unit. Masterlink ML 9600 stores, delivers and plays stereo 24-bit/96k audio on standard recordable compact discs. The unit can also produce and play back conventional 16-bit/44.1k Red Book format CDs. Masterlink features a 3.2GB internal hard drive with editing, digital signal processing and mastering functions, a 4x CD-R drive, and 24-bit A/D and D/A converters. The Masterlink should retail in Australia for around $4000.

Electric Factory: (03) 9480 6708

Soundscape & Hyperprism

Arboretum Systems Inc. and Soundscape Digital Technology have unveiled the new Hyperprism digital effects package for the Soundscape SSHDR1, Mixtreme and R.Ed digital audio workstations.

Hyperprism is the first full-featured multieffect package for the Soundscape platform. Hyperprism for Soundscape is fully compatible with the SSHDR1-Plus DAW and V2 software, the popular Mixtreme audio card and the new 32-track R.Ed digital recording and editing system. The pack will retail for $499.00. Purchasers of the 1.0 edition will receive the next upgrade – including further new effects – free of charge.

Soundscape Australia: (02) 9356 1955

Yamaha Nearfields

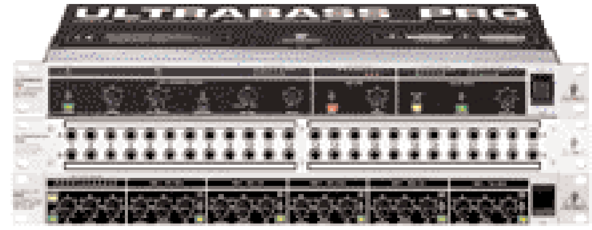
Yamaha has added to their range of powered nearfield monitors with the MSP10 powered monitor and comparable SW10 powered subwoofer system.

The MSP10 features an 8-inch long-throw woofer housed in a bass reflex design cabinet, providing a deep and tight low end. Its one-inch pure titanium dome tweeter with wide dispersion waveguide horn provides a 'smooth, high frequency response to beyond 40kHz'. It's internally powered with a 120W amp for low frequencies and 65W amp for high frequencies.

Yamaha Music Australia: 1800 805 413

Akai DR16 Pro

Akai has released the DR16 Pro hard disk recorder. Here's some of the features: 16, 20 and 24-bit recording, up to 96k sample rates in two-channel mode, 24-bit A/D converters on input, 10 track simultaneous record 16 track playback, five track simultaneous record eight-track playback at 96k. 16-channel digital mixer, ProTools file import and export. *The DR16 retails at $10,995.*



Behringer Ultra Gear

The new Behringer Ultra-Q Pro PEQ2200 is a five-band parametric equalizer – designed for broadcast, stage work, or in the sound or TV studio. The PEQ220 Ultra-Q retails for $399.

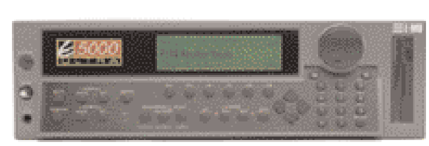
The new Multigate Pro XR4400 is a four-channel expander/gate which packs four frequency selective expander/noise gates in a compact single rack space chassis. The Multigate Pro combines simple operation with flexible control functions. The Multigate Pro is particularly suitable for handling the difficult separation of microphone signals when recording drum sets. RRP $599.

The Behringer Ultra-DI DI100 'puts an end to hum and impedance problems'. Any source of sound can be routed to the mixing console clean and balanced. You can even pick up a guitar amp (or power amp) directly at the loudspeaker output – on systems up to 3,000 Watts. RRP $129.

The Behringer Ultrabass Pro EX1200 is a bass enhancement system for live or studio application. With this digital subharmonic synthesizer you can add bass power to your sound system, installation, club or studio. RRP $399.

The Behringer Ultrapatch Pro PX2000 is a patchbay which helps you keep things organized in your studio or rehearsal room – whether you're dealing with complex patchbay cabling or just want a solution for smaller setups. RRP $199.

MusicLink Australia: (03) 97656530



E-mu E5000

The E-mu E5000 Ultra is the newest member of E4 Ultra sampler family and 'the most affordable Emulator ever'. As an E4 Ultra sampler, the E5000 provides you with the same professional features as the rest of the range: the intuitive Emulator Operating System, the new fast Ultra processor, access to the most comprehensive sound library in the world, and proprietary tools like Beat Munging and Z-Plane Filters, all at a competitive price. Some of the specs include: 64 Voice polyphony, 4MB RAM (expandable to 128MB), four balanced analog outputs (expandandable to 12) and ROM slots. RRP: $3495.

Music Technology: (02) 9369 4990

NEWS & NEW PRODUCTS

Cool Edit goes MP3

Syntrillium Software has announced the release of its new MP3 plug-in for Cool Edit Pro. Now you can import and export MP3 files right in Cool Edit Pro using a high-quality process based on the Fraunhofer's MP3 codec. The plug-in supports bit rates from 20kbps to 320kbps. The MP3 support means you can put your music on the web, send e-songs [yippee!], or even store up to 10 hours of music on a single CD, all at a sound fidelity that's generally recognised as pretty darn good for the rate of compression. There's a free, functional demonstration version of the MP3 plug-in available via download from Syntrillium's web site, or you can pay $49 for the real thing.

Major Music Wholesale (02) 9545 3540

VST drums

Steinberg Cubase VST v3.7/v4.0 can now fully integrate software synths into the VST environment. No suprise then that Steinberg are bringing on some new plug-in synths. First up is the LM-4 plug-in drum machine. The LM-4's six outputs (stereo mix plus four individuals) can be routed directly into the VST host's internal audio mixer ready for adding EQ and effects. LM-4 is supplied with 20 sound sets, each containing at least 18 drum sounds. Users can also build their own kits from any standard 16- or 24-bit AIFF or WAV files, even with velocity layers. Having the LM-4 as part of the VST environment also results in much greater timing accuracy over an external Midi device, a big plus in the realms of rhythm. RRP $295.

Get It On CD is the latest addition to Steinberg's Creative Tools product line. Built on Wavelab 1.6 code, it includes all basic audio editing functionality and CD burning in a package priced at $195.

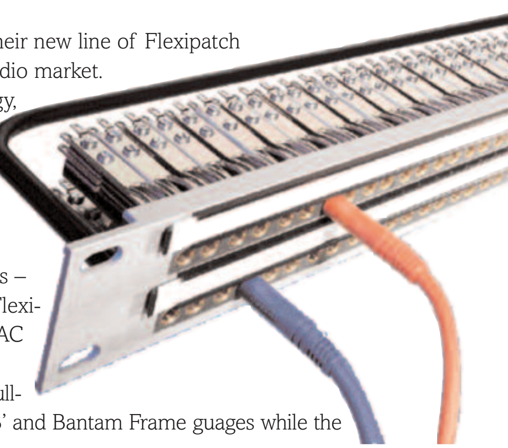
Music Technology: (02) 9369 4990

Budget TC

TC Electronic are at it again, this time addressing the studio on a budget with TC sound quality and operating system. The result is the M.ONE, and the D.TWO. The M.ONE is a dual effect processor where you can run two full blown effects without 'compromising the sound'. The D.TWO is a digital delay. Both units featurebalanced jack I/O and S/PDIF digital I/O. Otherwise we're a little sketchy on the specs as we went to press but we do know that each unit will be retailing for $1400, which is sure to bring TC to a new market.

Amber Technology: (02) 9976 1211

M&M patchfields.



Mosses & Mitchell have released their new line of Flexipatch patchbays for the professional audio market.

By utilising modern PCB technology, M&M have effectively removed the usual problems associated with the hard wiring of jackfields. Flexipatch is available in Long Frame 'B' guage and Bantam Frame gauge with either plain brass or nickel-plated brass jacks – both reliable and hard wearing. The Flexipatch is terminated using 56-way EDAC connectors.

A selection of half-normalled and full-normalled 'pods' is available in both 'B' and Bantam Frame guages while the 'B' Frame, a digital version is offered.

Control Devices: (02) 9356 1943

Antares Mic Modeler

The AMM-1 studio DSP processor allows any reasonably full-range mic to sound like virtually any other microphone – well that's the promise of Antares anyway. Using patented Spectral Shaping Tool (SST) technology, Antares created 'precise digital models' of a wide variety of mics, from historical classics to modern exotics, as well as a selection of industry standard workhorses. The user simply tells the AMM-1 what microphone they are using and what mic they would like it to sound like. The AMM-1 references the stored models of both the source and target mics and processes the input to create the sound of the desired mic. Antares reckon they've modelled over $300,000 worth of mics thus far. Sound too good to be true? AudioTechnology will let you know next issue.

Sound Devices: (02) 9283 2077

Bruce Evans passes away

It is with great sadness I write that Bruce Evans, the designer and maker of the wonderful ETR mic preamps, died of cancer in his home town of Newcastle, on Friday, September 10th, 1999.

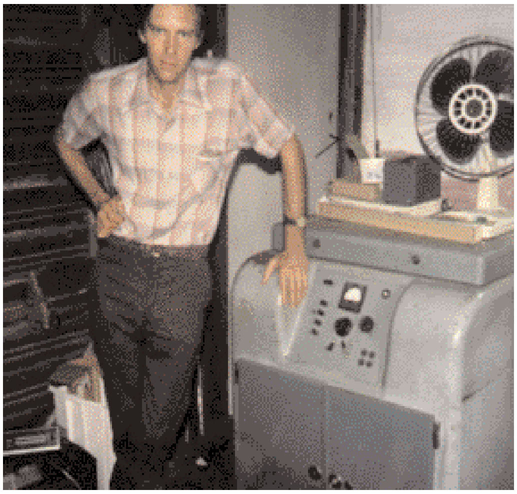
Bruce had been diagnosed with cancer only a few months before and, with his typical analysis and optimism, had fought bravely to the end. He was a man of unusual and fearless intellect, completely self-taught, who fastidiously stuck to his guns, and who would believe or support a theory only if and when it made complete sense to him. And who can argue with that?

The ETR mic preamps were only one part of Bruce's interests. He was also busy working, until the week he passed away, on numerous projects including his earth-shaking bass subwoofer (which I've heard and can testify to its impressiveness), compressors, limiters, hand-made ribbon mics, magnetic cartridge pick-ups, a small high-end (as always) monitor amp, and the next ETR mic preamp.

Bruce's funeral was held in Newcastle, and was very well attended. His eldest daughter, Kate, gave a great eulogy, and, at Bruce's request, we all applauded as the coffin disappeared through the curtain into the crematorium.

He will be very much missed, and I am grateful to have known him.

- Bob Spencer



Bruce was admired and respected by all of us at AudioTechnology. He was an innovator and an educator, and had many interesting audio designs up his sleeve. The wonderful ETR mic preamp was just one of a few that became a commercial reality before his untimely passing. I believe the Australian audio industry has suffered a greater loss than we can imagine.

- Greg Simmons

Neutrik Stage Boxes

Neutrik's latest NSB Series stage box systems combine quality and flexibility with modest pricing in a range of 12 models to suit a variety of live sound and studio applications.

The series features three different styles ranging in size from 8-In/4-out to 32-In/4-Out. All models feature high quality Neutrik XLR connectors with a solderless IDC design to ensure rapid, simple termination. The fully modular construction also includes a robust strain relief suitable for multicore cables up to 20mm diameter. The stage boxes are very competitively priced, ranging from only $96 to $169 recommended retail.

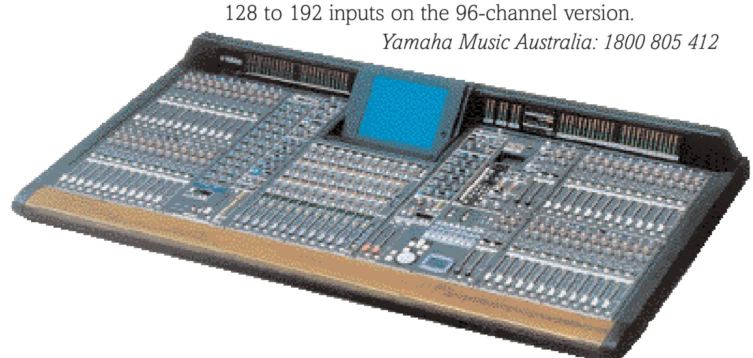
Amber Technology: (02) 9975 1211

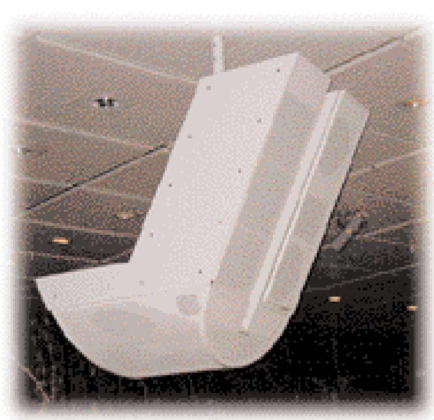
Yamaha's Live Digital Board

Amid longtime industry-wide speculation that the company would be releasing a new digital board at the 1999 AES, Yamaha finally confirmed the rumours by unveiling the PM1D, their first large-format digital mixing console designed specifically for live and installed sound applications.

The PM1D mixing console is a totally digital, computer-based system which performs all mixing and audio processing functions completely in the digital domain with 32-bit internal processing for superior audio quality. The CS1D Control Surface operates the system DSP1D digital audio engine, which is contained in a variety of stage racks which can be located with the control surface or remoted to the stage or other convenient location. (where all audio is processed). Configurable in both 48- and 96-channel versions with 48 mix busses, 24 matrixes and 12 DCAs (Digitally Controlled Amplifiers), the board system features top-quality 28-bit A/D and 27-bit D/A conversion. Dual inputs on each channel provide 96 inputs for the 48-channel system and 128 to 192 inputs on the 96-channel version.

Yamaha Music Australia: 1800 805 412





JBL sits in Parliament

Elecoustics has just finished commissioning a permanent JBL sound system for the Great Hall at Parliament House, Canberra. The space is quite reverberent, however, with strong echoes off the back walls, the speaker positions were constrained by a large hanging tapestry, and lighting and maintenance requirements. The solution was to create a moderately sized array for the main system with a constant directivity from 100Hz to 15kHz. "This would ensure the best possible tonal balance throughout the room and minimise the amount of sound radiated under the speakers, allowing for a second smaller system for down fill which could be tailored to minimise feedback," Elecoustics' Glenn Leembruggen explained.

Jands Electronics: (02) 9582 0909

Yorkville AP6040

Yorkville's latest addition to the Audiopro amplifier line is the AP6040, delivering 2,000 watts per channel (at 4 ohms), making it the perfect solution for subwoofer applications such as a pair of two to four large format subwoofers per amplifier.

The CR5 amplifier has been specifically designed for the budget-conscious consumer, and will be a competitive alternative for commercial and rental applications. Featuring 250 watts per side at 4 ohms in a rugged three rackspace package, the quiet, passively cooled CR5 includes separate rearmount channel gain controls, stereo/mono/bridge switch, power, clip and protect LEDs, a defeatable limiter and circuit breaker.

Dynamic Music: (02) 9939 1299

Grace Knight's 301 comeback

While the new Studios 301 were in a state of construction, house engineer and programmer Jochen Muller's work was very much underway. Set up temporarily in the studios' board room Jochen has been finishing preproduction on a Grace Knight album. Jochen was given an offer he couldn't refuse by 301 boss, Tom Misner, leaving his studio work in Europe and hitting the ground running in Sydney. AudioTechnology sat in as Jochen demo'ed some of the new tracks.

AudioTechnology: Tell us a bit about the Grace Knight project.

Jochen Muller: Grace Knight is obviously an accomplished jazz singer, but she wants a pop comeback. She's an incredibly versatile vocalist and easy to work with. We decided to use her jazziness, but bring it into an up to date context.

AT: I can see what you mean. There's a real swing to the groove, and there's a real ad lib quality to her vocals. How are you getting these tracks together?

JM: I recorded the vocal straight into Logic on an audio track. I gave Grace a click and asked her to sing the song the way she felt, with only a guitar as accompaniment – there was no backbeat from me. I found it was a good way of going about the recording – there wasn't any beat or groove to disturb her feeling for the song, so she flowed more easily. The downside is that it's been quite difficult to put her in time because the groove has a hard swing.

AT: I notice you're using the new Logic Audio v4.0. How are you finding it?

JM: Version 4.0 is brilliant. The parametric EQ sounds really good, the compressors are usable, the other plugins are very handy as well, the best being the tape delay and Autofilter.

In preproduction you need something like Logic Audio. It's no longer a case of just being good with a sampler or good on keyboards, preproduction is turning into the bulk of the process and you need to be able to do the lot as a programmer. I mean, even 80% of the effects are done in preproduction using outboard and plug-ins. Logic's a great tool for getting it to that end stage.

AT: You've also got the Mackie digital 8-bus.

JM: Yeah, I like the sound. I know from the internet that initially there were loads of problems in the States with it but I've never had a single problem. I think the console's plugins sound good, the automation is easy to use, and the assignable V-pots are a great idea. It takes a little getting used to if you're coming from an analogue console, but digital is here to stay.

When the Mackie goes into the programming suite upstairs in the new studio complex, it'll talk directly to the SSL console in the main studio. All the fader movements will be replicated by the SSL, which is great when it comes down to the mix.

AT: I notice you've got a Yamaha VL1, and a Roland JV1080, but what's your 'desert island' synth choice.

JM: The Axis Virus, I can't get enough of it. I'd have three if I could afford them. The new software update has ironed out most of the bugs of the original and it's great to have all the parameters there on pots. I've recently bought the Novation Supernova for the same reason. Its very important in my job not to waste time reading manuals. I need to be able to have a new synth up and running straight out of the box. The first time I saw the Supernova I had it working in minutes. There are so many new technologies hitting the market these days, and somebody in my job needs to see and operate something new virtually each month – it's not like the old days where you set your studio up and it can remain unaltered for years. Those new things need to be user friendly and up and running on the first day otherwise its just time wasted.

The Grace Knight album will be out early next year and will also be available as a 5.1 surround DVD, the first to be mixed in Australia.

For more information on Studios 301: (02) 9698 5888

Recording Red Hot Chili Peppers

The Californication album signalled the welcome return of the Red Hot Chili Peppers to the world stage. Paul Tingen discusses the recording with the album's engineer, Jim Scott.

The 'Californication' of the world is in full swing. Of course, it's been going on for decades, mainly via the influence of the likes of Hollywood and Disneyland. But recently there's been a major accessory to the global takeover, in the form of the latest album from the rejuvenated Red Hot Chili Peppers. It's their most successful album yet, topping the hit parades all over the world and turning platinum in Australia, New Zealand, Canada, Japan and the USA, and gold in almost all European countries.

Apparently the title came out of a visit by singer Anthony Kiedis to Borneo. He was surprised to see the local culture saturated with images from the US, and California in particular, whether it was Coca-Cola, Marlboro, and Hollywood movies, or T-shirts, posters, CDs and cassettes from Californian bands (including the Red Hot Chili Peppers themselves). *Californication* is therefore a wonderfully appropriate and prescient title. There's also the sleazy double entendre there, fitting for a band who are famous for hitting the stage virtually naked with each band member's member sheathed in a gym sock!

Genuine worldwide fame found the band in 1991 when the *Blood Sugar Sex Magik* album lifted them onto the international stage. It was also their first collaboration with the legendary Rick Rubin, founder of the rock and rap label Def Jam Records in the mid '80s, and now head of American Recordings. Rick came on board again for Californication, an album that many suspected would never happen.

The trouble began when their guitarist John Frusciante descended into a journey of drug-induced self-destruction. Four years later several near miracles have happened. Frusciante is off drugs, has a new set of teeth, some skin grafts on his arms and clearly still knows how to play the guitar. He has also joined singer Anthony Kiedis, bassist Flea and drummer Chad Smith to resurrect the Chili Peppers, and together they have created an engaging album. Most of the tracks have a high-energy rock vibe, but they're interspersed with many touching ballads, of which Scar Tissue became a monster hit.

Californication's engineer Jim Scott has an impressively long list of credits to his name, including Natalie Merchant, Lucinda Williams, Counting Crows, Tom Petty, Neil Young, Rolling Stones, Finn Brothers, Robbie Robertson, Sting, Santana, Seal, and Jewel. Scott was responsible for capturing the sound and energy of the Chili Peppers onto tape. He has done a remarkable job, resulting in a hard-hitting and gritty sounding album. It's also an album that largely in mono, and without reverb, but more on that later…

PT: Jim, could you take the story of recording* Californication *from the beginning?

JS: After the return of John Frusciante there was a long period of song writing and time for the band to reacquaint themselves with their songs. The band rehearsed most of the 1998 summer in Flea's garage, and when they came to the recording studio they were very well prepared, and in extremely good shape. They actually started work with another recording engineer, but after a week in the studio they felt things weren't happening. As luck would have it, I happened to be working in an adjacent studio in Ocean Way (now Cello Recording) in Los Angeles. I've worked with Rick Rubin before, so he asked me to step in.

After I took over, Rick and the Peppers became happy with the sound, but all I really did was capture the sound of the band in the room. On the first day we recorded maybe three or four tracks, the next day twice as many, and the next day another four or five. In all, we recorded 30 songs in about a week, which is a lot. It was a lot of tape and a lot of performances, but they were playing great, so all we had to do was get it down.

"Don't turn each session into a recording school, it's more important to get the ideas down while they're hot"

PT: Can you give us more detail about the setup in the recording studio, and what you recorded? Where they all in one room, for example?

JS: We recorded all four of them at the same time, which basically amounts to the sound of the album. There weren't many overdubs. John did some guitar overdubs on maybe two or three songs (the slide guitar on Scar Tissue is overdubbed, for example), but in today's world of overdubs that's not a lot. We also overdubbed some AC/DC-type piano power chords during the mix, but it is not like there's layers and layers of overdubs. The sound of the record is what happened during that first week of recording, and those 'live' drums, bass and guitar went on every song.

We set up in Studio Two at Cello Recording, which is a rectangular, medium-sized room. The drums were on a riser in the middle of the room, and there was one large iso-booth where we put John and Flea's amplifiers – just to keep them out of the drum room. We built a little doghouse around the bass speakers, to protect them from leakage from the guitar amp – but in my experience leakage is not an issue as long as you make at least some attempt to achieve separation.

There was also a small, separate vocal booth, where Anthony sang. Flea, John and Chad were about 10 feet away from each other in a circle, and Anthony was just a few feet away in this iso-booth, separated by glass. They could see each other all the time. There were no baffles around the drums, so they were just sounding out loud in the room. I set up two room microphones, but I didn't end up using them. The sound of the drums on the album is pretty close, and that's from only using the close mics. Chad has good sounding drums, and you can hear the dynamics and the details in his grooves, so it was just a matter of getting that on tape. The same with Flea, who has great bass technique, and even when he plays fast you can still hear all the notes. Because they all played so well, and because of the way I miked them, the overall sound was dry and punchy. Everything was clear and loud.

PT: How did you mic up the drums?

JS: To get that tight 'up front' sound you have to put the mics really close to the drums – the room sound did not come into it. On some records all you want is the room sound, and you get that great Led Zeppelin drum sound, but Rick prefers things to sound really loud and right in your face. He doesn't want the mic 30 feet away if it can be just one foot away.

The microphone setup was a basic rock'n'roll approach that I learnt in The Record Plant in the '80s, nothing too fancy. I used a Neumann U47 on the kick, putting it right inside. There were two Sennheiser 421 mics on the toms, two Neumann U87 mics as overheads, and a Shure SM57 on the hi-hat. The snare was picked up by two SM57s – one above and one underneath – plus a Neumann KM84 on top, which gave me a good-mic/bad-mic setup. The three snare mics all ended up on one track. The SM57 underneath the snare gave me more of the rattle.

We had a second set of drums in the room which were tuned completely differently, kind of mismatched and oversized. We used it on the track Porcelain. It had more rattle on the snare drum, a fluffier sound on the kick, plus a sizzle cymbal. I used leftover microphones for that kit. I think there was an SM57 on the snare, an Electrovoice RE20 on the kick, and two RCA77 ribbon microphones as a general drum balance. I also had a pair of Neumann M50 mics in the room for ambience on both drum kits. Although they went to tape, I didn't use them in the mix.

Waxing lyrical: Jim Scott (sporting the white shirt) in the studio

PT: Did any of the vocals recorded during that first week make it to the mix?

JS: We probably retained some of the 'live' vocals, but Anthony went back and redid most things. In many cases this was because of changes to the lyrics. We recorded his vocal overdubs in the small iso-booth as well. He came in every day around three o'clock to get ready for the vocal overdubs, which normally started at four. He had his vocal teacher down every day, and took his time to warm up. The dedication of Anthony and the band to get this record the best they could was awesome. Anthony sang about three or four songs every day, and then we would spend time picking and choosing the best pieces. He sang great – I think it's the best he's ever sung.

The vocal overdubs took about two weeks. The whole recording period took about five weeks, after which the mixing took a few weeks, largely because Rick wasn't always available to listen to the mixes.

PT: Can you run through the other mics you used on the album?

JS: The bass went DI into the desk, and I also miked up the bass amps with a Neumann U47 tube microphone. I usually mix a 50/50 blend of DI and speaker mic to one track. I find that it's important to have the sound of moving air on the bass. Moreover, sometimes there's distortion on the bass that comes from the pick-up, but it sounds much nicer when it's gone through an amp and a speaker.

For the guitar I used two Shure SM57s and two Neumann U87s, one of each on each cabinet. Again, it's a good mic/bad mic combination. All four went down on one track, and together they made a nice big guitar sound. Anthony always used a Shure SM57 for lead vocals. We put the mic on a stand, but I'm sure he held it in his hand, and leant on it and swallowed it – that's how he gets his sound. But, it meant that it was important to compress him, in order to protect the tape.

The backing vocals were almost entirely done by John, although Flea and Anthony sang a few parts as well. I usually used a Neumann U87, but sometimes, on the spur of the moment, if John had an idea that he wanted to try out quickly, I would have him sing into Anthony's SM57. Over the years I have discovered that you gain little from auditioning 25 microphones when you know you already have a good sound – just record the good sound and get it over with. Don't turn each session into a recording school, it's more important to get the ideas down while they're hot.

PT: What was the console you were using for the session?

JS: As I said, we recorded the album at Ocean Way Studios, which is now Cello Recording. I like to work there because it has the best rock'n'roll microphone collection in the world, plus many vintage Neve modules and vintage analogue tape recorders. The album was recorded using a '70s Neve 8038 desk, on an Ampex 124 24-track, of which there are hardly any left in the world. We didn't use any Dolby. We like hiss, hiss is our friend. Listen to any record from The Who, and it's full of hiss. The Neve has excellent mic preamps, so I used them. All other mic preamps are trying to be Neves in my opinion, so it was good to have the real thing.

This particular desk had the Neve 1073 EQs, and I used quite a bit of EQ while recording. There's a huge difference between what somebody hears in a recording room – where a 500w guitar amp is blowing his hair back – and what he hears in the control room, where the same sound comes out of a speaker only six inches high. So you have to put everything in its place, and make sure it feels and sounds right. The way to do that is by using EQ and compression, or anything else you can lay your hands on. For this reason I added mostly mid-range to the electric guitar, between 1k and 2k, as well as a lot of low end – the Neve EQ gives you 56Hz and 100Hz settings, so it was probably around 100Hz that I was doing the boosting. It is good to add low end to guitars, because under the disguise of sounding exciting, they can easily get small and thin. It may jump out of the speakers, but it's not big any more.

I compressed John's guitar with a Urei 1176 as I recorded it. I didn't do much else to it, because most of the sound comes from his fingers. John uses a few pedals, but he's not really a pedal guy. He starts with a guitar, a cable and an amp – we were never waiting for him to set up his sound. He and Flea played fairly loud, which of course created amplifier compression. John has amazing control of his own dynamics. On a track like Get On Top his guitar sound would go from

huge to tiny in one instant. That's all pretty much done live, and it's nothing to do with me. It's what made this record really easy to record, they were really ready to go. Flea was a case in point. That distorted bass sound that opens the album comes simply from the way he hit the bass guitar. I didn't do anything to that. Flea never went to his amp to change anything, and used the same bass for the whole album, apart from on *Road Trippin'*, where he played an acoustic bass. I compressed his DI and his amp bass sound with Teletronix LA2A tube compressors. I don't think I added much EQ, apart from maybe a little bottom end, around 56Hz. I don't remember adding any mid or upper mid range for clarity, because he sounded clear. I really want to stress that these guys are great musicians. I have seen them go through highs and lows, but right now they're really good. I think they're the best band around right now.

PT: What about the processing on the vocals and the drums?

JS: The compressor I used on Anthony's vocals was a Urei 1176. Everybody uses that one. Even though my records can sound fairly radical, I don't overdo things with the settings. I used a pretty standard setting: 4:1 ratio, with fast attack and slow release, just enough compression to do the job. I printed his vocal with compression, it was simply part of the sound.

There was a lot of EQ as I recorded the drums, to that classic old rock'n'roll vibe. Personally, I think you can find the sound of rock drums on the faceplate of an API EQ – add a lot of 5k and 100Hz, and there's your drum sound! I also used the Neve EQ, but it was the same concept. I added a lot of low end to the kick drum and the toms, probably around 50Hz, and some top to the snare and cymbals, just enough to make it pretty.

PT: There's also an acoustic guitar on* This Velvet Glove*, and on* Road Trippin'*, plus some keyboards on some tracks.

JS: My usual set up for acoustic guitar is to use a Neumann U87 and an AKG 452, once again the good mic/bad mic idea. Between the two you can find the ideal sound, and you can get brightness and fullness. You don't want the mics to be too close because the sound will get boomy, so I place them a couple of feet away, pointing just above or just below the sound hole. Unless you want a Jumpin' Jack Flash sound, acoustic guitars just need to be pretty. I combined these two mics on one track. We recorded the acoustic piano during the mix at The Village in LA. We went for a Beatles-style *A Day In The Life* kind of thing: using one mic that's compressed real hard, to make the sound cut through. I used one U87, not too close to the piano or the hammers. We wanted the sound to have a lot of ring, and got a lot of attack from using EQ and a compressor – in this case a Urei 1176, with a lot of input gain and a very slow release time, so the sound almost gets louder as it sustains.

PT: How many tracks did you eventually end up with?

JS: I ran two synchronised Ampex 124 24-track machines. One of the machines was really a vocal slave. On the main machine I had around 10 tracks of drums: kick, snare, stereo toms, stereo overheads, hi-hat, stereo room, and then I'd usually print a compressed drum track. I took a little bit of all the drums and ran them into a compressor, compressed it firmly and then printed it on a track. It makes the rough mixes sound a little bit more exciting. Often there was also some percussion, like tambourines and so on. The second 24-track enabled us to put down lots and lots of lead and backing vocals. We then transferred these tracks to ProTools, to experiment and edit and slide things around and archive them. It's much faster to do vocal composites in ProTools than on analogue tape. And when we wanted to run the changes past Rick, it's easier to do it in digital, all the while keeping a map where parts came from, so you can later retrieve it from the analogue tape.

Cello Recording's Studio 2, where the recording of Californication took place. The console is a custom Neve 80-series Class A, 24-bus model, with 40 inputs and 1073 EQ/channel amplifiers with GML moving fader automation.

PT: So apart from vocal comping, you're not a big fan of digital?

JS: I do prefer analogue, but I'll do my work any way I can. I'll use any tool. We were able to make decisions about where the good stuff was more quickly by using ProTools, but I do think you are shooting yourself in the foot by using digital all the way.

PT: So what can you tell us about the mix?

JS: I mixed on a Neve as well, using Flying Fader automation. I actually ran the two 24-tracks during the mix, to be able to get back to the original source tracks. By the end we had bounced the best bits down to single tracks. In terms of effects, the mix was totally dry. I didn't add any reverb, and very few other effects. We used a human compressor on the vocal, i.e. we kept our finger on the fader, making sure Anthony stayed in the mix, which the automation really helped with. But the tape sounded good, and had sounded good since the first day, so there was no reason to change that. It wasn't like there was anything to fix in the mix. All I had to do was balance it correctly and make it really loud. There wasn't much re-EQing either. We had an exciting drum sound, so why fiddle with it?

My main concern while mixing was to build the tracks in the right way, so that each chorus would get successively louder and the dynamic of the song would get more intense as the song progressed. I also added a bit of bottom end and a little top to the overall stereo mix, just to make it more hi-fi. You can ruin records by trying to make them sound too big. If you add a lot of low end it may just slow the record down, and it will start to wallow in its own size – Chili Peppers records need to be spritely and funky. I used quite a lot of Neve 33609 compression on the stereo mix, to add punch. We mixed to various digital formats (Apogee DAT, regular DAT, 96k DA88) and analogue two-track. We compared the results, and ended up using the 30ips, no Dolby analogue two-track.

PT: Apart from the last track,* Road Trippin'*, and the odd panned guitar overdub and toms, most of the record sounds rather mono. Why?

JS: Yeah, it's all dry and mono – quite a daring record really! Mono helps to keep things loud. I think Rick decided that he didn't suddenly want something blasting from the left channel on this record. We just wanted to hear the songs. We tried panning the guitars left and right, but whenever we did, it was like, 'it sounds better in the middle, let's just leave it in the middle'. When there's some really great guitar going on, it can be right behind the vocal. The record is bass, drums, guitar and a singer – it's not that complicated. It sounded best in one big mono.

Josh **Abrahams**

Josh Abrahams' name has been inextricably linked to Addicted To Bass, but Christopher Holder discovers there's a lot more on his plate.

In 1999 few people have been more influential in furthering the cause of dance music in the Australian mainstream than Josh Abrahams. [Josh will thank me for saying that when he's being pursued by a bunch of 40-year-old rock dinosaurs with greying pony tails, hitting him over the head with their Led Zeppelin box sets and shouting, 'So it was you, ya bastard!']

Songs like *Addicted To Bass* and *Thrillseeker* finally jumped the Triple J hurdle and, via mainstream FM, found their way into every office, car, and kitchen in the country. *Addicted To Bass* went top 20 and his album, *Sweet Distorted Holiday*, has been doing a brisk trade ever since. Then there's the 'small' matter of his double ARIA award winning triumph last month, which has done little to harm his mainstream marketability.

Josh Abrahams production skills were honed during his time with the ARIA Award winning Future Sound Of Melbourne. Josh's star rose higher when DJ heavy-weight, Carl Cox signed him up for a worldwide techno release on his Ultimatum label. The result was *The Satyricon*, an uncompromising dancefloor manifesto. Now, since the release of *Sweet Distorted Holiday*, Josh has been signed by Sony, contracted by Baz Luhrmann to work on his latest film, and has moved his Fishtank Studio to the commodious environs of the Festival building in Sydney.

I met Josh at Fishtank to discuss gear, Baz and his rare incurable bass addiction.

Christopher Holder: I imagine when you set up shop here at Festival you were able to get everything 'just so'. Is this your dream studio?

Josh Abrahams: It's certainly a room where I've got access to almost anything that I want. The only thing that's lacking is a latest generation digital synth, like a Korg Triton. I've got a whole wall of analogue keyboards, analogue drum machines, old analogue effects units, and what's great is that everything can be accessed from the patchbay now. So, if I want to take a vocal sample and run it through the filter on the MiniMoog I can. It's great being able to think of a weird connection and then being able to make it.

But I'm aiming for the studio to build up what I hope is the best of everything. We've got some great old mics like the AKG C12 and a Neumann U87, and a new

Neumann KM184 which we're going to use for drum overheads and brass. We've also got the Manley VoxBox as a preamp, which is very warm and smooth.

CH: Anything here that you can see will need changing?

JA: I think the desk is the only thing. I like the Mackie 8-bus, it's really workable and I'm really on top of it, but for some reason it's just a bit lame in the bottom end. The Mackie has a 'sound' to it and anyone who's familiar with the Prodigy album will know that sound. It's a very direct sound, very in your face, but now it's a little too techno or electronic for my liking. All the machines that I have are so electronic anyway that I tend to want to soften them with the desk. But for the price, it's unbelievably musical and versatile.

CH: What did the Mackie 8-bus replace when you bought it?

JA: I moved from a Tascam 8-bus, which was a desk that I really enjoyed. Being into hard techno at the time, I really liked overdriving the gain, and it had a great bottom end. I also loved the Midi muting on the Tascam. I was able to do a lot of weird stuff by muting the effects returns. The Mackie console doesn't have Midi muting, which I really want at times. To make up for that I've got a Midi-controllable eight channel Midi gate, called a Niche Audio Control Module. It's just a box of channel faders really, and you put it across the inserts. That way, the things which don't receive Midi volume messages, like the analogue synths, can still be faded in and out via Midi.

> "I **think** making **baby noises** in **the** middle of a **chorus** really **works**"

CH: Talk us through your wall of analogue synths.

JA: I recently bought a Roland SH7, which I've fallen in love with. The SH7 completes the set of four Roland SH series synths I wanted. The SH7 and the SH5 are both huge beasts, there are all sorts of weird routing possibilities on the filter section and they both have ring modulators, which can also receive an external signal. For example, I can run a drum machine through the SH ring modulator against one of the synth's oscillators, which normally produces something interesting. Then I've got the Roland SH9 and the SH2. They're all great for fat bass sounds, just plug them in and you sound like Devo without trying.

Next up is the MG1, which was actually made by Realistic and sold at Tandy stores. But it's a design licensed from Moog – its actually a two oscillator MicroMoog in a Tandy package. I bought it from a guy in Melbourne's west who does these bizarre modifications, and he put a couple of pots on the side for me. The first pot is a cross modulation pot, forcing Oscillator 1 through Oscillator 2, which gives you this really bizarre screaming cross mod sound. The second pot is a filter modulation which forces Oscillator 1 through the frequency of the cut-off filter, which results in some insanely hard sounds.

Then there's my Roland TR909 techno drum machine, which also has mods on it by the same guy. There's a pot for the kick drum pitch, so I can tune it really high or low, and there's a switch for the pitch decay envelope on the kick – you can lengthen the decay on the kick which can make it sound much more like a Simmons tom. There's a long release on the white noise of the snare drum, so you can get a really long decay. The snare drum is composed of two different pitch oscillators, and a white noise generator, and I can pitch the different oscillator independently. There's a pitch control on the hi-hat, and there's a mix pot on the handclap which balances the white noise aspect and the click part of the clap. They're great mods. He has done other mods which are so extreme that it's very hard to get the machine to sound like a 909 any more, but this is more of an enhanced 909, which I love.

CH: I see some Korg synths over there…

JA: The Poly 6 is a lovely pad machine and has a lovely built-in phaser/flanger unit – a really great '80s string sound. The Korg Monopoly, which is really whacky. It has four oscillators, and you can force them through each other in different cross mod and sync type ways – you can really get that machine to scream.

CH: And the rest?

JA: There's the Sequential Circuits Pro One, which sounds a little like anything by Depeche Mode – I actually suspect that this Pro One may have belonged to Pseudo Echo at some stage in its life. Similarly the Jupiter 8 has a very specialised sound – the minute you

plug it in you're sounding like the Eurogliders, or Icehouse – but again, if you push it you can get it to sound really extreme. The Roland TB303, which sits atop its own Mutron digital delay unit. The Mutron has a nice analogue front end so I actually use it as an overdrive unit, specifically for the 303 which sounds so much more tougher if you put it through a little overdrive.

Some of the analogue synths, including the Roland TR909 with mods, Korg Polysix, and Roland Jupiter 8. Top left is the Tandy MG1.

The MiniMoog I bought quite recently. I don't use it a great deal because it's such a specialised sound – I sample it more than I actually play it. I'm slowly getting into sampling my analogue synths rather than playing them directly. They're all hooked up via two Kenton Pro4 Midi to CV converters, so often I'll use them like any other sound module – like the Roland SH machines, which I tend to run directly via Midi the whole time and they're quite stable. But the Moog is quite unstable. You might program a sound and then lose it after a while, that's why I tend to sample it.

The Roland TR808, which is Midi'ed, I use that quite a bit depending on the style of music I'm doing. I love the sound of the 808. Also the Roland TR626, which is bit of a cross between the TR707 and the TR727 – a drum machine and a percussion machine full of 8-bit Brazilian samples. These drum machines are quite specialised, so the 808 may only be used in a hip hop track, or maybe the hi-hat might end up on a dance track, while the 626 is such an early house sound, it's too specialised to use extensively.

I enjoy these synths for their fatness, or their particular speciality. They all sound very different from one other. For example the Pro One might look quite similar to the SH9, but it's an incredibly different sounding machine. The SH9 is a single oscillator synth but it's so fat. The Pro One is a two oscillator synth and it's quite thin sounding, but it's filter routing makes it very powerful.

CH: You talk about sampling your synths. Has sampling played a bigger part in your production in general?

JA: It has, yes. Currently I'm producing a solo album for Amiel, who sang on *Addicted To Bass*. Being a teen of the '90s, she hears these old synths and complains that they sound "…too '80s". So it's quite a tough job for me to use these machines and pull them into a more modern context. I find that I often need to sample them into the Kurzweil K2500, which has loads of weird filter and distortion algorithms, and make the sample sound more current. The '80s were about exploiting the inherent character of synths, the challenge now is to use them as powerful raw source material, put them into the digital domain and change them.

On my current album, *Sweet Distorted Holiday*, the sampler and the turntable were probably the most prominent players. Meanwhile my previous album, Satyricon, was much more of a techno record and I really used the wall of analogue to its logical extreme. I've got heavily into the sampler, going to the second hand record stores and buying old RCA records, and Readers Digest albums.

CH: But for a man who cut his teeth on Detroit techno, borrowing sounds with a sampler is bit of a cheat isn't it?

JA: Maybe, but I found there's also a lot of cheating in the techno scene now. For example, if you buy the right synth, set a standard kick and hi-hat pattern running and hold two notes on a string machine you've got a techno trance song. Sampling is cheating if you just take a chunk of someone else's music and stick the words, 'you can't touch this' over the top. But that's not art, that's not writing music. I like to think that I take a much more sophisticated approach.

CH: So are you a second hand LP man or a sample CD man?

JA: I use both. I have quite a good collec-

More analogue madness, including (top right) the Roland SH2 responsible for the Addicted To Bass bassline.

tion of sample CDs, for beats particularly. A vocabulary of classic breakbeat loops has been developed which people now know and get excited about on the dance floor. So I have those standard loops on CD and often use them. I rely on old vinyl for melodic material, or rhythm sounds – scratch it a bit, put it through an effects unit and then sample that.

But I've become increasingly obsessed with Steinberg's ReCycle program. Just record a phrase into your sampler, then suck it into ReCycle on your computer, and it'll chop your phrase into its constituent parts, then it spews it back into your sampler complete with key groups and programs. Suddenly you not only have your favourite drum loop, but the drum kit that it was played on to put together your own variations. ReCycle is probably my instrument of the moment, because it just opens up a whole new world. It used to be so laborious sampling the same drum loop five times and editing it down, to get the kick, the snare, and the hi-hat, etc.

CH: You're currently doing some work for Baz Luhrmann. What's that about?

JA: It's for his next film, which is a musical. It sounds a bit odd but musicals are in the air and you'll find that many big film directors are making musicals. I think even Martin Scorcese is working on one now.

CH: I can't wait to see Bobby DeNiro 'singing in the rain'. So what's your role?

JA: Being a musical, it has required a big emphasis on music from the beginning. I was in charge of the music pre-production, which meant that while Baz and his co-writer Ray Pearce were writing the script I was in his office drafting musical ideas. They'd give me an idea and I'd knock something up, and put it onto CD so there would be something to play while they wrote the script, or wrote new lyrics, or whatever was the case. It was quite challenging having to write a lot of stylistic stuff which might not have been exactly what I'm into.

Since then Baz has got me involved in the production process along with people like Marius DeVries, who did the music for Romeo+Juliet. [See our interview with Marius in AudioTechnology Vol. 1, Iss. 3.]

CH: Producer/programmers don't come much bigger than Marius. He must be an interesting guy to work with.

JA: Marius is unbelievable, it's like working with the master, I've just learnt a lot. Everyday in conversation I learn another thing from him, which I know will save me hours. Techniques in Logic especially – he's the Logic king.

There's one technique in particular which we've been using that's made me rethink my approach to studio work. Marius has an assistant in the UK who will start a draft of a song idea, and will send it to Marius in Sydney. Marius will add some ideas, stick some further ideas on Midi, and because we're all using ProTools 24 with Logic Audio, we all share a standard Midi song format and audio format. Marius will send it to me, and, being the analogue man, I'll add in some ReCyled beats and some fat synth sounds. I'll lay them down to audio within his song structure, and send it back to Marius. It's great to share this high quality sound file format that can travel around the world. It's really changed the way I work.

Totally Addicted

***CH: Tell us about the writing of* Addicted To Bass.**

JA: Addicted To Bass was my first real foray into pop with ReCycle – and I loved it! I knew that drum & bass was all about using ReCycle and, this song being a drum & bass track, I knew that it was going to be a ReCycle track.

There's two drum loops used in the first verse 'gogo' section which came from the *Cuckooland* series of sample CDs. Then I got my main loops from a drum & bass sample CD called *Jungle Warfare* and chopped them up in ReCycle.

It was also the first time I'd used the Matrix Editor in Logic. ReCycle makes a Midi file of the loop in its original form which you can import into Logic. If you open up the Matrix Editor you can just drag the individual edits around, writing your own version of the loop, doing variations and fills.

At that time I'd just bought an ElectroHarmonix Microsynth, which I used for the first time on Addicted To Bass as well. The Microsynth takes the source then adds a voice an octave below and an octave above, and then adds a square wave which is meant to track the pitch of the input. It's intended as a guitar pedal but I sent part of the drum loop into it. I brought it back on another channel on the desk, and when the drums do a solo in the chorus I'd de-mute that channel. The effect is a wah distortion and it adds a new dimension to the drums at that point. I've heard that the Chemical Brothers have used it quite a bit on their drums – it was a bit of fun and pretty effective I think.

Because it was a drum & bass track I was using a few of the drum & bass tricks. For example, I wouldn't normally pitch bend my drums but one drum & bass technique is to have a drum fill composed of a bar's worth of 32nd or 64th notes of one drum sound pitch bending down. Also there's some reversing of the drums in places – all the tricks that you hear in the underground, but I layed a pop song over those tricks.

Also the kick and snare is very up front and uneffected, which is less likely to happen in drum & bass, and that's more of a rock element I suppose. I wanted people to hear the beat clearly, while all the drum & bass rhythmic stuff happening in the background gives it that drum & bass feel without the beats taking over.

CH: Where did the main bass sound in the choruses come from?

JA: That came from the Roland SH2. It's a two oscillator monophonic analogue synth and I used two saw tooth waveforms slightly detuned from each other. The filter is wide open, making it really buzzing and fat, and I put it through a phaser. For me it was a nod to the old techno sound. There was a lot of old school techno that had these huge fat and phased bass lines. So you get this growling, threatening sound, which is what I was after.

CH: And later on a sub bass sine wave sound follows that bass line?

JA: Yeah, which was a sound I made on the Roland JD990 – so it's a digital bass, but it's really very rich and subby. I EQ'd out a lot of the mid content so you would just feel the subs – I didn't want the sub bass and the SH2 bass to be competing for space.

Actually, I made a point of leaving out a lot of bottom end until Amiel the vocalist says 'totally addicted to bass' for the first time. A lot of people have said, 'for a song about bass there really isn't much', and until the sub bass drops that's quite true. But that's intentional, so when she says the words, the bass really slams in.

CH: There's also the bass sound in the bridge which is almost like a descending tom roll.

JA: That's an old favourite Yamaha FM bass sound. It's called Nasty Bass on the TX81Z. So many electronic musicians have fallen back on that bass sound, it's tight and clean and you hear it all over the place.

CH: What about the other instrumental sounds?

JA: The strings that come in on the second verse and the last chorus are an attempt at a Duran Duran-style '80s string sound. It's a Selina string machine emulation. Selina was one of the first string machines around, and I happened to have one – it was my first keyboard. But I didn't use it, there's a better Selina string pad on the 990!

There's also a weird harpsichord stab [sounds a little like a car horn], which came off the Jungle Warfare CD. I just wanted some sort of detuned stab as a counterpoint to the vocals in the chorus.

CH: And compression?

JA: Drum & bass producers love their compressors. There's heavy compression on most of the drums and the sub bass in particular.

A mate of mine, David Carbone from Future Sound of Melbourne has got heavily into drum & bass, so much so that he moved to Bristol [drum & bass's spiritual home]. He spent ages discovering exactly how they got those sub bass sounds. He ended up with a Roland SH5 which makes the right fat bass sounds, but he couldn't quite get it doing what he was hearing on the records. He finally found if he compressed the bass sound to the max, sampled it, then recompressed it and resampled it, and did all these ridiculous resampling techniques he could get the sound. Just playing the SH5 wasn't enough.

CH: What about the vocals? They're nicely distorted…

JA: I used the Tech21 SansAmp for that. I spent loads of time and probably did five or six passes to DAT recording different versions of the level between the vocal and the vocal distortion. I wanted the vocals to be tough but the words needed to be intelligible. In some other songs the vocals were nicely distorted but it was quite hard to understand the words. Luckily, the words are really clear on this one, and it turned out to be a good mix.

CH: If one thing sticks in the listener's head it's the nonsense vocal line 'a-wah-ow-wah-ow'. How did that eventuate?

JA: When I was originally writing the song I was imagining the overall sound would be a lot less hardcore and more like trip hop or a light jazzy drum & bass sound. Then I got sick of jazzy drum & bass and thought a more hardcore vibe would be fresher. But initially, that vocal sound was going to be a double bass riff. But then I started wandering around humming it and thinking that it wasn't such a bad vocal line. Some of the '80s songs that were iconic to me used weird vocal noises. Like the 'ooh-wa-ooh' of *Video Killed The Radio Star* from the Buggles. Everyone remembers that. So I think making baby noises in the middle of the a chorus really works! In the end, it really was the idea that made it a hit song.

CH: Amazing how it can pivot on…

JA: …one bloody idea? Yeah, totally. And to me it shows how much a hit song is about an idea and not entirely about production. You get to a point where you expect a certain level of production from everybody, and now it's not all about how well you produced it – I mean this song wasn't that well produced, it was done in my living room back in Melbourne. My vocal booth was three mattresses set up to look like Stonehenge, with the singer standing in the middle! Reading about the Moby album in your magazine [AudioTechnology Vol. 1, Iss. 6], I couldn't believe the crap that he made that album on. But then they stick one of the songs on an advertisement for the Star City casino, and it sounds so lush and rich. Sure, it might not sound like Madonna's *Ray of Light* but ultimately it's about the tunes and ideas.

Stadium Australia

Stadium Australia will be the focus of entire world when the Olympics come to Sydney in 2000, and thankfully the sound for the venue was given high priority. Christopher Holder spoke to The PA People's Chris Dodds about his $3.5 million sound installation.

Over 107,000 people packed into Stadium Australia for the recent Bledisloe Cup rugby match between Australia and New Zealand. There are few sporting events that evoke more passion. Forget about England v Scotland in the World Cup soccer semi finals (too many tartan skirts and exposed beer bellies), forget about the Superbowl (too many pom poms and hotdogs), forget about the America's Cup (too many winged keels and blazers) the Bledisloe Cup has a gladiatorial rawness that makes any other contest look like a CWA bridge club meeting.

It was a monumental night in many respects. A record crowd was in attendance and Australia rubbed the All Blacks' noses in the turf with a record victory.

The evening also marked another significant step in the Olympic stadium's coming of age. The eyes of the nation (and a good part of the rugby playing world!) were on Stadium Australia, and it came up trumps.

Getting in and out of the venue was painless, the view from all seats was good and unobstructed, the beer was nicely chilled, and the sound? The sound was something else. I think we all take for granted that in large stadium events the sound is all about compromises. Coverage, intelligibility, power, are all compromised by the inherently difficult nature of the large space. My praise of the system that night is almost without qualification, the PA was highly intelligible, there was virtually no huge delay problems (all the sound I heard came from the two nearest speakers and nothing arrived half a second later from the other side of the park), and the coverage was very uniform as I wondered around the stands. All the compromises that we to take for granted as being part of the stadium experience seemed eerily absent. I say eerie because the bi-product of the clarity is a strange isolation. A try is scored, 100,000 people rise as one in rowdy homage to the scorer, but you find that the only cheering you hear is from the portion of the stand that you see around you. This takes some getting used to, but clearly represents the way of the future for stadiums as a modern entertainment venue.

Christopher Holder: Chris, I was quite impressed by the lack of messy delays and reflections in the audio while at the rugby. What's that due to?

Chris Dodds: Acoustically it's a very controlled space compared to many other stadia. Much of that is due to the roof that concaves up in the East and West Stands. An open roof propagates the sound out, while here the concave roof allows the sound to go up the back of the stands and dissipate within that space. The roof goes up a long way and a lot of the energy is trapped. Anything that does go up into the roof and reflects, will reflect back into the stand, it doesn't reflect out on to the ground – whether it's crowd noise or the PA.

CH: So there aren't any clever construction tricks, or space age materials, it's all down to the architecture?

CD: That's mostly right. But as far as materials go, the roof is a polycarbonate sandwich, basically a honeycomb polycarbonate, which is reasonably transmissive of low frequencies, which helps, but quite reflective at high frequencies. But ultimately it's more about the geometry than acoustic treatments.

CH: But the controlled nature of the space must have given you great encouragement when you were doing your modelling of the system?

CD: It's a remarkably forgiving acoustic space, and it's very well designed architecturally too – I don't know how much of that was good luck or good management, but, acoustically it's a remarkably good space. I mean, if you sit in the East Stand with that stand's sound on only, then turn the West Stand's system on, you hardly hear it, nothing. It meant that with careful design we were able to achieve great coverage – the result was a variation of less than 2dB over the entire stadium. As far as intelligibility goes, there were a few small 'worst case' pockets which were down to an STI [Speech Transmission Index] figure of around 0.47, but the vast majority of seating was between 0.55 to 0.63. In a space like this, those figures are unheard of, it's just mind bogglingly good. I should stress that those figures are measured, not predicted.

CH: Okay, let's get down to nitty gritty, what speakers have you put in here?

CD: The speakers are all Bose. In back-of-house [all the function rooms, public concourses, corporate boxes, etc.] there's Model 8 and Model 32 ceiling speakers. There's Model 25 small cab speakers, and there are Model 502A, which is commonly called the Banana Box, in large acoustic volume paging spaces, and they also serve as under balcony fills on the East and West stands. In all there's 1600 speakers in back-of-house. For front-of-house (inside the main stadium area) there's Bose 402s around the balcony and on the North and South stands. The rest of the cabs are LT9702, LT4402 and LT3202 which are all horn loaded mid/high cabs, and the 502BE bass cabs. All of the bass cabs are in directional arrays. [See diagram for a typical cluster configuration.]

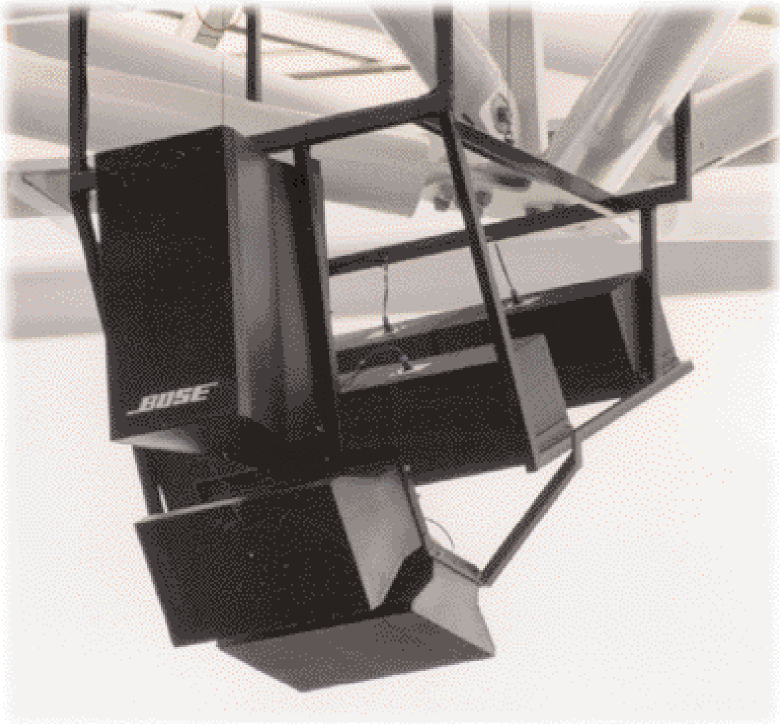
CH: Bass directional arrays? How does that work?

CD: I think the only way I can answer that is, yes it does work. We're not letting all the tricks out of the bag, but if you physically look at them, they are arrays which are digitally controlled, and it achieves the goal. So all the bass cabs are beam steered, they're all pointing at seats with something like a 12dB front to back ratio at 125Hz. So it's just as directional, if not more directional than a horn loaded cabinet.

CH: Why did you spec Bose?

CD: The primary reason why The PA People corporately specify Bose is, a) it's a good product, and b) the design development tools Bose has brought to their package, give us a level of certainty in design unmatched by any other manufacturer. Bose's Modeller, (the acoustic modelling tool), and Auditioner, (their auralisation tool), in our opinion are a significant level above any comparable tools – particularly in their ability to predict the reverberant field, which after all is the real field. Sure, other people make good speakers, but they're a lot harder to use than the Bose products and a lot harder to actually know what result you're going to achieve. For example, the correlation between predicted and actual perfor-

The PA People's Chris Dodds at the control room's Soundcraft K3 console.



A look at a typical seven-speaker cluster in the East and West Stands. Refer to the CAD diagram opposite for the cluster's specifications. Photo courtesy of Bose's Brian Chilcott.

mance for this system showed that we exceeded our coverage predictions – which shows Bose are relatively conservative – and as far as intelligibility, we were pretty much on the money.

CH: What have you got powering the Bose speakers?

CD: We're using 56 Crown CT Series amps for back-of-house, and 94 MA series amps for front-of-house. There are four different amps in the CT range and they're purely selected by the number of speakers that they're powering. Same applies to the MA range, there's three amps in that range, and we just match them to the right size cab. The amp racks are located in four points around the ground.

CH: Those are the new Crown amps with the DSP card slots aren't they?

CD: Yes, the new USP2 modules. Each module provides eight bands of filter, as well as delay, compression, and load monitoring within the amplifier.

CH: So what do you see as being the principal advantages of having that functionality in the amp rather than in the control room?

CD: The dimensions of the system is one benefit. If you've got 160 amplifiers, like we do here, and you needed separate processors on every channel, then you'd have a system that generates 320 outputs. As it is we have a system that generates 60 outputs, so that's a significant benefit. Also the way we've implemented the system using Cobranet, we have several levels of functionality that you wouldn't get – or would be very expensive to get – using a centrally located process system.

CH: Cobranet is a fairly recent technology, what exactly is it about?

CD: Cobranet is a proprietary digital audio protocol from a company called Peak Audio in the US, who are licensing it to a number of vendors, including both Peavey and Crown. It's becoming a standard for distributing audio via ethernet, where we can distribute 64 channels of 20kbit audio over a standard 100baseT ethernet connection. Rane are about to release some Cobranet stuff, QSC have got it. It's a pretty exciting mechanism.

We can use Cobranet to easily reconfigure the system amp by amp, zone by zone. It allows us to have a virtual network, which is a significant benefit. But to use the protocol in the exact manner that we wanted, we needed the back-of-house amps to have Cobranet going straight into them. So, in co-operation with Crown, we developed our own interface to do that.

CH: We're heading into system control territory here. Perhaps we should discuss the control issues?

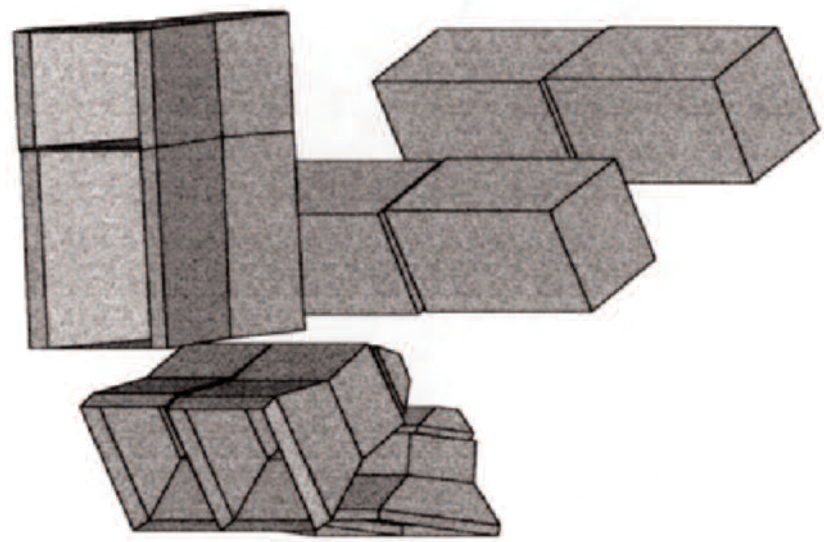
CD: From a control perspective, the system is split into two halves. The first half is a fairly standard but comprehensive analogue front end, the second half is where we believe the smarts are – the control system is what makes the thing tick.

Looking at the analogue side, there are copper tie lines from eight major positions around the field – four corners of the stadium, plus four in-field pits under the turf. Each of those has 12 mic lines on a multi-pin connector and six XLR lines, they all come up and terminate at the patchbay in the control room. Then there are other dedicated positions – like the media boxes, television boxes, press split positions, the OB control room, TV screen control room, Telstra termination room – where there are also cables. They either come up as undedicated tie lines, or they come up as dedicated feeds. The patchbay is comprehensive, with all of the control room's Soundcraft K3 I/Os on there, and all the insertable effects units. The system is designed so that if you strip the patchbay, the system will still work. We've also got almost every conceivable playback device in here, and a good array of processors and effects – you should be able to come in here and not have to bring anything.

CH: Sounds like you've got the front end covered extremely well…

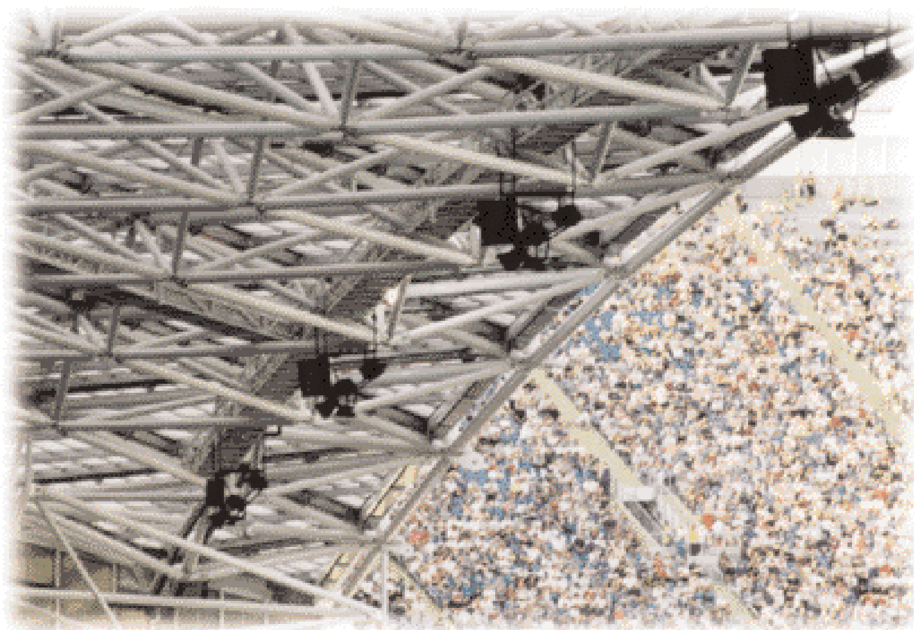
CD: Yes, which brings us to the control system. It takes crowd mics, paging mics, background music sources from the desk, and so on, and routes and manages them throughout the site.

The actual audio processing box is a Peavey Media Matrix, while the amp control system is Crown's IQ. Both



There are twenty clusters distributed along the main front gantry of the East and West Stands. A typical cluster comprises a steel frame loaded with a Bose LT3202 mid/high cabinet, a Bose LT4402 mid/high cabinet, a Bose LT9702 mid/high cabinet and four Bose 502BE bass cabinets. The cluster is supported by two Demag DS1 wire rope winches.

systems have their own dedicated PC interface, and we have our own software which controls those two systems plus the paging stations and all the ancillary hardware. So, from an engineering setup perspective you're using the relevant native application – whether it be ours for the paging stations, Media Matrix for the routing, or Crown IQ for the amps – but from an operational perspective you have a single user interface which addresses all three as separate pieces of hardware. In our opinion, this was the clue to making the system uniquely versatile and friendly.



The leading edge of the roof of both the East and West Stands is above the front row of the seating, which provides an ideal position for the seven-speaker clusters. Photo: Brian Chilcott.

CH: Is there much of an overlap between the roles of the Crown IQ control software and Media Matrix?

CD: The Media Matrix's primary function is as a router, with some basic dynamics and ducking functions. All the processing which relates to where the speakers are and the delays, that's done in the amps via the USP2 modules.

CH: So going back to Cobranet and your interface modifications, how did all this fit in?

CD: The latest version of Media Matrix has Cobranet as an option. So the actual Media Matrix box itself has a number of RJ45 connectors on the back and not the traditional BOB (breakout box) interface. Then Peavey has their Cobranet to Audio box, which they call CAB, which we use to get the inputs into the system and the outputs to the analogue amps. As well as that we go directly to the back-of-house amplifiers with a Cobranet interface in the actual amp we were talking about earlier. This modification is basically a daughterboard for the Crown USP module. It's fair to say that these amps here are the first in the world to have direct Cobranet interfacing, the only ones at this stage…

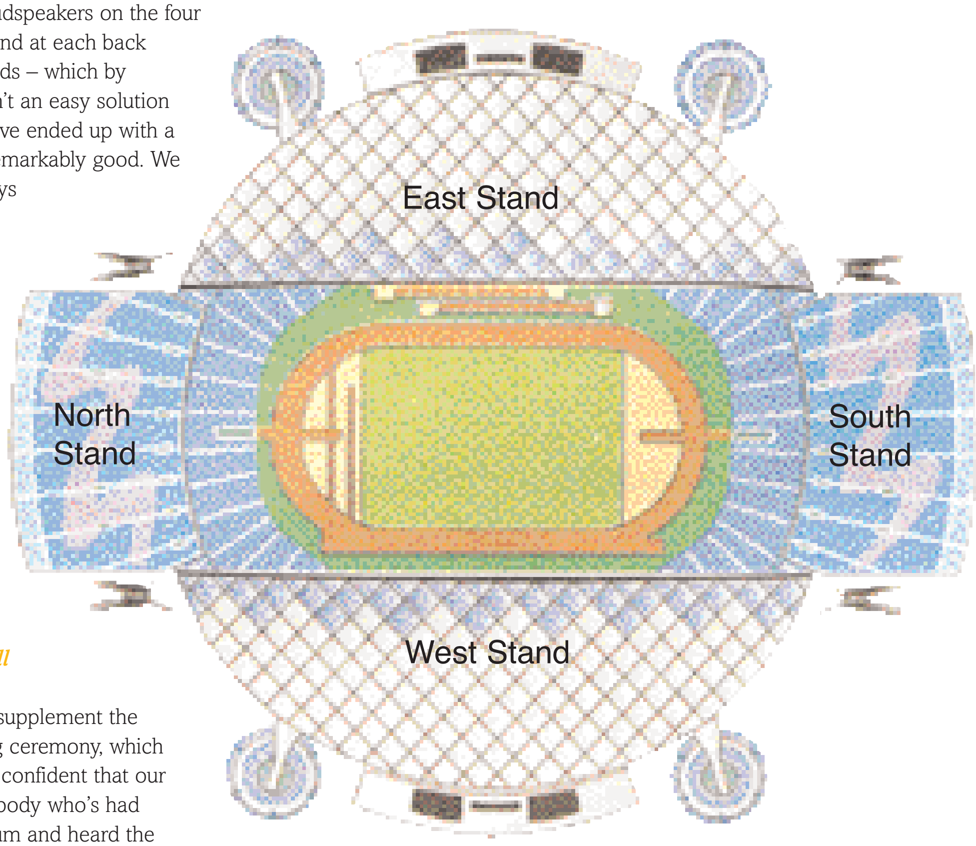
CH: Can you tell me about any innovative solutions you had to make with the installation of the cabinets into the stadium, or did the architecture and construction make life easy for you?

CD: There are certainly some innovative solutions in the coverage of the end zones. The East and West Stands were easier, they have roofs with leading edges which are basically above the front row of seats, and as such were a perfect location to put loudspeakers in those stands. The roof concaves upwards, which means that those speakers which are on the leading edge can't 'see' the back rows of seats, so there's a supplementary row of delay speakers half way up the roof. But primarily the East and West Stands were good spaces to work with. The end zones, however, are temporary structures without roofs and caused us a few headaches. We were instructed that there should be no obstruction in front of those stands which put an end to our initial proposal of stringing a catenary cable in front of each stand and having a centrally located point source of speakers – which sonically would have been a superb solution.

So after evaluating and modelling many options, we put the loudspeakers on the four lighting towers that stand at each back corner of the two stands – which by anyone's analysis wasn't an easy solution to implement. But, we've ended up with a solution that sounds remarkably good. We use beam steered arrays of low frequency cabs, which keeps the low energy where we want it, and propagating down onto the people and not washing across them, so to speak. The mid/high cabs are in the same pattern. It's a weird looking array of loudspeakers.

CH: So how do you think the Olympic opening ceremony will sound?

CD: They're going to supplement the system for the opening ceremony, which is appropriate, but I'm confident that our part will perform. Anybody who's had cause to use the stadium and heard the system before and during an event, have all been very happy. The sound guys from the Denver Broncos when they were in town couldn't believe the quality of the sound. We are confident that what's here is as good as anything in the world.



Mastering with George Graves

In the first of a series of interviews with prominent international mastering engineers, Tony Mantz talks with George Graves of Lacquer Channel.

Veteran mastering engineer George Graves is definitely of the old school, and takes his craft very seriously. Originally from California, he now resides in Toronto and works at Lacquer Channel, one of Canada's leading mastering facilities (www.lacquerchannel.com). As the name implies, Lacquer Channel is fully equipped for mastering to vinyl – something George does on a weekly basis.

His career includes stints at RCA (Hollywood) and Doug Sax's famous Mastering Lab, and contains a diversity of clients ranging from Peter Gabriel, U2 and The Guess Who to Killjoys and jazz diva Holly Cole. In the following interview, George talks about the art of mastering for vinyl, the impact of digital technology, and the directions mastering may take in the future.

Tony Mantz: How did you get started in mastering, George?

George Graves: I was looking for a summer job before going back to college, and there was a vinyl pressing plant in my home town. Unfortunately no job was available but the chief engineer had a studio in his garage, so I hung around and learned how to cut a disk. From there he offered me a job at the pressing plant, and history started. That was in 1963.

TM: Do you approach mastering as a science or an art form?

GG: Both. I started out mastering lacquers [for vinyl], where you have set parameters because of the lathe and you need to know what problems can be caused by level and EQ. After learning this, it stays in the back of your head and you don't seem to think about it. Now that I'm mastering CDs I use the same starting point and go from there, although levels and EQ don't cause as much of a problem for CD as they do with lacquer cutting.

TM: I notice you have a Neumann VMS70 cutting lathe. How much vinyl are you mastering these days?

GG: Our lathe cuts a full 10 hour day once a week. That's about seven jobs…

TM: Is there any different approach you utilise in either EQ or dynamics processing when committing to a lacquer?

GG: Yes, I don't put as much bass or highs on for lacquers. You must remember that the lathe is a mechanical device, it's not dealing with numbers like a digital system. Lacquers have limitations because they are cut with the relationship of program amplitude and side duration in mind. In other words, you have to consider the effect that the overall level will have on the recording time available on each side of the disk. Loud program material requires wider grooves, and that results in less physical space on the lacquer to work with, and therefore less recording time per side.

On a digital format, like CD, extreme volume or heavy bass content won't shorten the available recording time because there isn't the same physical/mechanical relationship. The digital world is based on 0's and 1's, and adding bass won't make those 0's or 1's any wider!

So there are limitations with lacquers that you have to be aware of, but there are many more limitations on playback systems. Some of those are beyond the control of the mastering engineer. Most people do not know that the weak link for vinyl is the playback system.

TM: Could you explain those playback limitations?

GG: Records have a tendency to skip when the record is cut too hot in level, or when low frequency sounds – such as bass or kick drum – are too high in the mix. Out of phase information doesn't help, either. The playback stylus is like a bob sled travelling down a mountain – when the G forces are too high the sled leaves the ground, as does the playback stylus when the record is cut too loud. The mastering engineer has control of these things.

Beyond that, we have to consider the differences between the cutting lathe and the playback system. The cutting lathe's stylus moves across the disk in a straight line, from outside to inside, as it cuts the groove. But on most consumer turntables the playback stylus is pivoted at one point, and so it moves across the disk in an arc.

This can cause intermodulation distortion during playback – especially at high levels, like in dance music. That's why it is important to keep the modulation of the record to the outside of the disk as much as possible (i.e. keep the duration of each side short).

Proper stylus weight is another factor in playback distortion. If it is too heavy the stylus bottoms out when playing. If it is too light, it floats and creates distortion. You also have to make sure the anti-skating is set up properly – this will help prevent the playback stylus from skipping.

There are many more factors but the best place to find the answers is in the Audio Engineering Society journal 'Disk Recording', Volumes 1 to 28. It was published from 1953 to 1980.

TM: You recently did a reggae project for Polygram where you cut the program to 14-inch lacquers before transferring it to CD. You said it gave the CD a really punchy sound. Did you cut the lacquers before or after final mastering, and how did that approach come to you?

GG: First I EQ'd for the lacquer. I added a little EQ and level adjustment and then played the lacquer into our Sonic Solutions system. This was done on a track by track basis. The idea came from Gordie Johnson of Big Sugar. He remembered the great bass sound from his old reggae records, so we tried it and he loved it.

TM: When listening to a program to be mastered, do you have a strong idea of where you want to take the track, or do you find you'll try a couple of options?

GG: If the client has a specific direction that they want the project to go, I ask them to bring in a CD for reference. That way I have something for a starting point. If the client wants me to do my thing, I ask which track they think has the best sound and I start with that one.

Sometimes the first track is a ballad. When that happens I leave it and move along until I find an up-beat track or a rocker, and start there. That way I don't make the ballad louder than the rocker! There are times when I'll have to go back to the first song and re-EQ it, because my perspective changes as I work through the project.

My problem is that I like a wider dynamic range than most of my clients. This makes me do a lot more dynamic control (limiting, etc.) than I'd like to.

TM: Do you master to CDR or 1630?

GG: We can master to both. Some clients still like their masters on the 1630 media because, traditionally, it forces the CD plant to make their glass master in real-time. Unfortunately, a lot of CD plants nowadays transfer the 1630 to DDP, and write their glass master from that. This way they can write the glass master at a faster speed, which sort of defeats the purpose. We have found that a DDP written at real time (x1) sounds the best.

TM: While the audio chain is only as strong as its weakest link, is there any particular stage you would deem the most critical?

GG: Line amplifiers. They're at the beginning and end of each piece of audio equipment. They make it or break it.

TM: What gear resides in your audio chain?

GG: Our basic board is a modified Neve mastering console, originally designed for RCA in the '70s. All the line amps have been de-ironed [transformers removed]. Our EQs are Sontec (Parametric Disk Mastering EQ), a pair of Pultec EQP-1As, a Dolby 740 Spectral Processor and, of course, the Neve EQs. We've also got a stereo Manley Variable Mu compressor/limiter, Urei LA-4As, and some handmade mastering compressors from Tubetronix (a local fellow who makes LA-2 based compressors with a difference!). Digitally, there's a Lexicon 300L and a Sony SDP1000 digital mastering board. We use Apogee converters and Sonic Solutions workstations.

TM: Any particular favourites?

GG: Sontec and Pultec. Pultecs have that nice tube sound that's needed for so many digital recordings. The top end is sweet and it rounds the bottom to help remove some harshness. Sontecs are very nice sounding parametric EQs, good for digging out or putting in those special frequencies that are needed to make the recording cook.

TM: Is there anything in the digital realm that you've found appealing?

GG: The Weiss bw102 limiter. It has a clean sound that doesn't get too woofy when you use it in hard limiting. It can also be a good sounding de-esser. But the problem here is that you need two sets of limiters if you want to limit and de-ess at the same time. This can get you up into the big dollars for equipment.

TM: How do you manage to keep up with the latest gear?

GG: I use my assistant, Phil, to keep up on all the new toys. He tries them and, if something really stands out, then we both listen to it to make sure we're in agreement.

TM: How often would you audition or shoot-out equipment?

GG: Depends if we think we need something that will help out. I don't buy equipment just because my competitors have it. I feel the equipment we have is top notch. Analog equipment will always hold its value more than digital.

TM: Do you ever see analogue being phased out as digital technology and algorithms get better?

GG: Yes, but not in my life time!

TM: Do you think recordings have improved over the years? Is technology making things better, or is it covering for a lack of engineering skills?

GG: Yes and no. It depends on the style of music you are referring to. Technology has improved the quality of sound. Unfortunately engineers have not improved on the way this technology is used. As a mastering engineer I hear all kinds of bad recording techniques. It is like a painting – some art is traditional and you can recognise it, while other styles of art are like an abstract. It's hard to know what is correct, but the engineer who recorded it thinks it's the cat's meow. Because there are no rules in recording and everybody has their own opinion, everything is correct.

TM: It seems the bar is always being raised (or lowered) to produce the world's 'loudest' CD. How do you feel about this? Is it a demand of your clients?

GG: Yes, my clients are always saying "I want my CD to be louder than #@! &%". When I hear something that is loud, my first reaction is like everybody else – the hype factor is there, but then I get tired and my ears start closing down.

I think the client should produce two masters, one for the consumer and one for the radio stations. Let the consumer have some dynamics and let the radio have its hype and lack of dynamics.

TM: You sometimes give your time to young bands and take them through the mastering process, because it's still a misunderstood concept. Has this 'client education' helped your business?

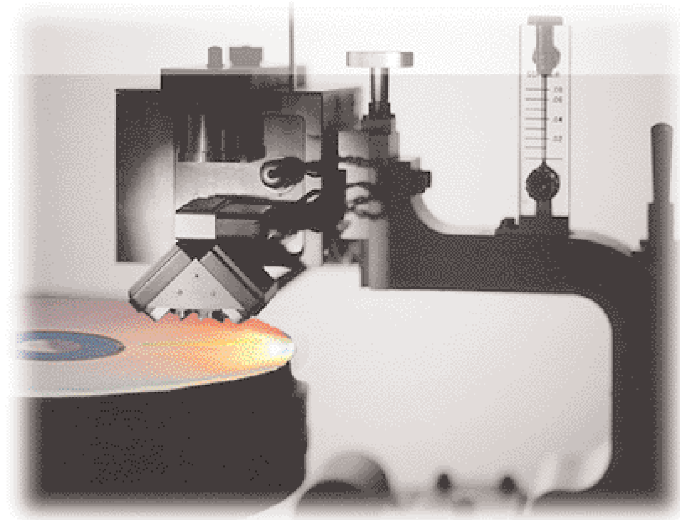
GG: Yes! The more your client understands the mastering process, the more it helps to network your service.

TM: Rightly or wrongly, many Australian bands believe the quality of mastering in the USA is better than they can get here in Australia. Being so close to the USA, does that mentality exist with your clients?

GG: Yes, especially considering that New York City is only an hour's plane flight from here. The good thing about having young bands and low budget projects is that they can't afford the prices that NYC mastering houses charge. But, once they've obtained the bigger budgets, they're gone. The grass always seems greener on the other side of the fence.

TM: What has been the most significant change or development in mastering over the years you've been involved?

GG: Computers…



TM: Where do see mastering going in the future?

GG: The basement! The cost factor is the big item, and the small client cannot afford the professional facilities. So, they settle for the low end and they get low quality.

TM: I read an article where you talked about backyard 'mastering' people who run on presets and cheap PC-based set-ups, and how they give professionals a bad name. You said that DVD and surround mastering will sort these people out. What did you mean?

GG: Because DVD and surround mastering is so costly, it will be a while before the basement establishments have the equipment to do it. When the equipment comes down to the price where the basement establishments can afford it, the quality will come down with it.

TM: What moves have you made towards DVD and surround mastering?

GG: None. Canada is a large warehouse for US product. We don't see an immediate need to move in that direction.

TM: I really enjoy watching DVD movies in surround sound, but still haven't found surround music as enjoyable an experience. Do you think it's just an infancy thing, or am I being too narrow in my approach?

GG: You are right and you are not being too narrow. Humans are used to hearing things from the front, and when sound comes from behind or from the extreme sides we become uneasy – we naturally start thinking that something is wrong or something is going to happen to us. Have you ever noticed how people like to sit in a cafe with their back to a wall, or so that they are facing the door? For this reason, I think the best sounds to put in the rear or extreme sides are ambient sounds.

TM: What do you do to preserve your hearing? Do you monitor at a particular SPL?

GG: The monitor level goes up and down. The norm is between 85 and 90dB SPL. About every two hours a short break is taken. That way I can get another coffee and visit the water closet…

TM: Are you a fan of nearfield monitoring when mastering?

GG: No. I like the sound of large speakers in a great room. Nearfield monitors remind me of headphones – you can hear detail, but they are not reality.

TM: Do you enjoy music as a recreational pastime, or is it difficult to get out of critical listening mode?

GG: It is hard to get out of critical listening mode. My darling wife refuses to go to concerts with me because of this!

Preparing Audio for the Internet

In Part IV of our ongoing series, Scott Christie uses Apple's QuickTime to deliver audio on the web.

QuickTime was introduced by Apple Computer way back in 1991. Most people's initial encounter with QuickTime was when they inserted their first multimedia CD-ROM into their computer. As the images, movies and sounds came to life, it was most likely QuickTime that enabled the synchronisation and delivery of these multimedia elements. Through the '90s QuickTime became the defacto industry standard for CD-ROM multimedia delivery. Although slow off the mark to adapt QuickTime to the internet, Apple has recently gone all out to make QuickTime technology a major player in delivering multimedia content on the web.

Unlike most other types of software you may encounter within internet audio production, QuickTime isn't an application or set of applications but rather an 'enabling technology' that is actually an extension of your computer's operating system. Once installed into your system, QuickTime's multimedia functionality can be called up by numerous programs, including word processors, multimedia applications and, of course, web browsers (via the QuickTime Browser Plug-in).

QuickTime stores all of its media elements in what are known as 'movies', which are recognisable by their '.mov' file name extension. A QuickTime movie is basically a container that provides the playback and synchronisation of a wide variety of media types, including video, audio, text, timecode, music/Midi, sprite/animation, tween, MPEG, QuickTime VR and 3D. In all, the latest version of QuickTime (QuickTime 4), provides access to 35 media formats. Sound file formats supported include .WAV, .AIFC, AIFF, AU, AVI, Mac Sound Resource, OMF, Sound Designer II, LAW, and MP3 – as well as being able to playback Midi files.

At this point it's important to note that a QuickTime Movie can contain all or, in fact, just one of these media elements. Hence the term 'audio-only QuickTime movie' refers to a file with the .mov extension which contains a single .WAV or .AIFF audio file, for example. The 'audio-only QuickTime movie' is the particular flavour of the .mov file that we'll be looking at in this tutorial. For the latest version of QuickTime Version 4.0x head for www.apple.com/QuickTime. There is, however, one catch… the Standard version of QuickTime 4 enables the playback of all the file formats mentioned above but doesn't allow for comprehensive conversion, editing and encoding. To get into the world of QuickTime multimedia authoring you'll need to activate the Pro version, which will set you back US$29.95 and serves as the basis for the following tutorial. For my money, the Pro version is well worth the cash if you have the remotest aspirations for multimedia production.

The Player

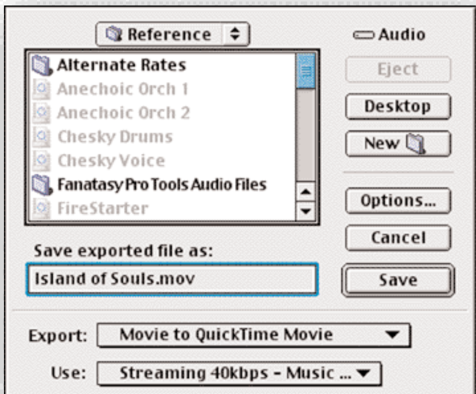
Once you've installed and registered your version of QuickTime Pro, head for the rather underachievingly named QuickTime Player application located inside the QuickTime folder. Behind this mild mannered facade of a few menus lurks a surprisingly powerful multimedia editing program capable of viewing, editing, encoding, combining and manipulating a host of media types. The reason for the rather bland front-end is that the QuickTime Player application (previously known as MoviePlayer) was initially developed as an 'in-house only' program by Apple engineers to test QuickTime's latest functionality, and only later was it released as a multimedia authoring tool.

The QuickTime Player application can serve as both the encoding application and the playback application for QuickTime multimedia. A downside of this is that QuickTime's limited menus and 'layers of dialogue boxes' work environment makes for a pretty clunky interface for encoding. Other applications such as Peak, MovieCleaner – or, in fact, any application that supports the standard QuickTime 4 export interface – can also be used for encoding and may be better suited to large scale internet audio production.

And so onto the actual production of QuickTime audio over the internet. As with the production of RealAudio (as covered in Vol. 1, Iss. 6) the process remains basically constant regardless of the technology: capture, encode, embed, upload, test… get paid! Or so goes the theory.

Capture

First up you'll need an audio file to encode. Even here, QuickTime can help. If you've got a song you'd like to capture from a garden variety audio CD, insert it in your computer's CD-ROM drive, launch the QuickTime Player application, head for the File>Import menu command, and navigate to the audio CD. Select an audio track by clicking on it and then click on the Convert button in the first dialogue box. This will bring up a second dialogue box where you hit the Options button. Select your settings, i.e. 44.1k, 16-bit, mono, start and end times, and then name and save the audio file to your computer's hard drive. The QuickTime Player now imports a direct digital copy of the CD song onto your computer – similar to other programs such as CD Stripper, CD Ripper etc.



You should now have the shiny platinum QuickTime Player transport controller on your screen. To audition your song file simply hit the big button marked Play. If your chosen audio file was already on your computer, use either the File>Open Movie or File>Import menu commands. Both will convert your audio file into a QuickTime Player-friendly version, as long as the file type is a .WAV, .AIFC, AIFF, AU, Mac Sound Resource, Sound Designer II or MP3. At this point you can also use the QuickTime Players' File>Save As command to convert audio files into any of the other recognised file formats – except, unfortunately, MP3 – making the QuickTime Player a handy utility for basic audio file conversions.

Code X

As the File>Import command unlocked QuickTime's capture secrets, the File>Export command is the key to QuickTime's encoding abilities. Selecting this command takes you through a series of four (yikes!) dialogue boxes which allow you to adjust the sample rate, bit depth, stereo/mono, codec type, bit rate for internet audio encoding as well as internet streaming options (covered in the 'Delivery' section of this tutorial, Vol. 1, Iss. 5). The actual audio codecs employed were integrated into QuickTime after Apple went on a shopping spree for some well developed internet-savvy audio codec technology. These came in the form of the Qualcomm PureVoice speech codec and the QDesign Music codec. The bad news is that the latest version of the music codec – QDesign Music 2 which comes with QuickTime 4 – is unfortunately not backwards compatible with QuickTime 3, so only computers with QuickTime 4 installed can decode the files. The good news is that the QDesign Music 2 codec sounds quite amazing considering its meagre file size – particularly good for creating stereo audio files. The blurb from QDesign claims "better than MP3 quality at all bit rates up to 128 kbits" (the defacto standard rate for stereo MP3 files over the web). Other improvements in the new QDesign codec include a higher resilience to network errors and faster encoding speeds which allow for the 'live' streaming of audio data from a server, thus providing web broadcast functionality. A Pro version known as the QDesign Music Pro 2 codec is also available directly from www.qdesign.com. The Pro version provides a number of further tweaks and delivers even speedier encoding times. However, this will set you back US$399.

The QuickTime Encoding Bit

There are two ways you can approach the encoding of audio files using QuickTime 4. The first option is to use one of QuickTime's preconfigured codec settings as found in the 'Use' pop up menu located in the first dialogue box you encounter after the File>Export command. These presets consist of 'Streaming 20kbps Music', 'Streaming 20kbps Voice', 'Streaming 40kbps Music' and 'Streaming 40kbps Voice'. Logically, 20kbps codecs are applicable to 28.8k modem connections and the 40kbps codecs are for a 56k connection. QuickTime lacks an intelligently scalable codec technology such as RealAudio's SureStream, so it's important to note that you'll have to encode a separate file for each of the connection speeds you want to accommodate, i.e. 28k and 56k, and then embed/upload each file accordingly. Using a separate QuickTime utility application called MakeRefMovie – available from www.apple/QuickTime/developer/tools – a single link can be placed on a web page that accesses only the encoded file that suits your connection speed, but more on this in a later tutorial!

The second method is to build the encoding parameters from the ground up rather than deal with the preconfigured ones. QuickTime, unlike RealProducer, allows you to customise the sample rate, bit depth and bit rate for encoding, along with selecting the actual type of codec. These parameters are located and tweaked along the File>Export>Options>Settings>Options dialogue box path. If you've got the time, the second method is a good place to start as it gives you a feel for the differences between 22.050k versus 32k sampling rates, 16-bit versus 8-bit audio and 20kbps and 40kbps encoding, for example. Also be aware that some of these encoding settings can take a reasonably long time for a computer to process. Three minute 44.1k/stereo files can easily tie up a 300MHz CPU for ten minutes or so. You can speed up the encoding time by selecting 'Speed' in the Optimisation settings (found in the QDesign Music Encoder dialogue box), however, 'speed' comes at the expense of quality and has no effect on the speed of decoding the audio file from the internet.

Auditioning your encoded files is as simple as using the QuickTime Player's File>Open Movie command and selecting your newly encoded file. A couple of tips for general encoding include: always use the highest bit rate possible relative to the intended connection speed; previously encoded files and 8-bit files are usually unsuitable for re-encoding (i.e. garbage in = garbage out). Finally, regardless of which encoding method you use, you should select the 'Movie to QuickTime Movie' option in the Export pop-up menu. This saves the file with the all important .mov extension which is essential for browsers to recognise QuickTime files and process them appropriately.

Delivery

QuickTime 4, like RealAudio, enables you to use two types of streaming – HTTP streaming (referred to by Apple as 'progressive download') and RTSP streaming ('true streaming'). In HTTP streaming, the audience experiences the audio as the data gets progressively downloaded to the their hard drive. To encode your audio file for HTTP streaming using the QuickTime Player's

File>Export command, make sure you select the 'Fast Start – Compress Movie Header' option in the Movie Settings dialogue box. Having selected this option, QuickTime will begin playing the audio file when it detects that enough data exists for the entire file to be played through to the end without interruption. This method of streaming has the advantage of requiring no dedicated media server, so you can stream files from server space usually allocated to you free of charge by your ISP. Other advantages include: users wishing to replay the data don't have to reconnect to the web since it's already sitting there on their hard drive, and media can be streamed at a higher quality or data rate than what the network throughput might support. This form of streaming is ideal for short amounts of data and is thus particularly suitable for delivering song excerpts on the web for promotional purposes. HTTP streaming was also used for delivering the trailer for 'Star Wars: Episode I The Phantom Menace' – claimed to be the biggest download event in the history of the web.

On the other hand, RTSP streaming requires a server that handles the delivery of media in real-time, whether that media is captured live or is stored on disk. The server provides a constant stream of media at a data rate that is consistent with the actual throughput of the network. Digital data is transferred and displayed – and discarded once you've seen it. Although a three to 10 second cache of data is stored to compensate for occasional network 'burps' that might otherwise compromise quality, at no point is the entire movie stored on your computer.

RTSP streaming is most suitable for live broadcast events, lengthy programs and large scale delivery as well as having the advantage of allowing the audience some level of control over the playback of the content. RTSP streaming is limited by the fact that a specific Macintosh G3/G4 server is required (with Mac OS X server software installed) in order to deliver your audio content. The good news is that Apple are keen to make inroads into this market and are providing the server software for free.

We've managed to cover capturing and encoding in this tutorial. Hopefully this is enough to get you started using the powerful QuickTime Player application. Next tutorial we'll cover embedding QuickTime Movies on web pages.

By The Way…

I noted with interest that online music e-tailer ChaosMusic has teamed with global independent record company V2 Records Australia to release the first track of the posthumous Michael Hutchence solo album to radio stations across the country via secure downloading on a special website for the occasion – http://www.hutchencesingle.com.

ChaosMusic used Liquid Audio technology, which allowed the secure downloading of the track 'A Straight Line' to radio stations with a log on and password to the special Internet site. The song was delivered instantly at 7am on September 13 for twelve hours only, two weeks prior to the single being released in retail stores. The song was downloaded by regional stations who recorded it and could put it to air within a matter of minutes.

Interesting stuff and probably a real pointer to the future of global dissemination of audio for broadcast.

ARTIST	PROJECT	ENGINEER	PRODUCER	MEDIA	SESSION DETAILS
Sony Studio (02) 9383 6461					
Vince Jones	Album	Ross Ahern	Vince Jones	Protools 24	Mixing Live Album
Elsewhere Fine	EP	Tod Deeley	Elsewhere Fine	Protools/Quantegy 499	Mixing & Tracking
Human Nature	Demos	Simon Tonx	Human Nature	Protools 24	AFL Grand Final
Storm & Serenity	EP	Louise Taylor	Storm & Serenity	Protools/Quantegy 499	Recording & Mixing
Passionflowers	Album	Tod Deeley	Lee Cutelle	Protools 24	Mixing
Trackdown Digital (02) 9568 5607					
Cut	Film	Simon Leadley	Guy Cross	Protools	Full music underscore
Flipper	Animated TV series	T.Ryan/T.Lista/D.C	Yoram-Gross EMTV	Protools	Full music & Post audio
Iva Davies	Single/LP	Simon Leadley	Iva Davies	Protools	Full Orchestral Mix
Skippy	CD Rom	Damian Canduso	Forest Interactive	Protools	Voiceover & Mix
Asian Civilisation	Talking Audio Guide	Torey Lista	Narrowcasters	Protools	Voiceover, Edit & Mix
Velvet Sound Recording Studios (02) 9267 2915					
The Cool of Me	Album	Tony Wall	Tony/Band	Quantegy GP-9	Album Track & Mix
Cryogenic	Album	Tony Jarrett	Adrian Grigorieff	Quantegy GP-9	Album Track & Mix
Belle's Pocket	Album	Steve James	Steve James	Quantegy 499	Album Tracking
Montana	Album	John Haeny	John Haeny	Quantegy 499	Album Tracking
Psikore	EP	Lacklan Mitchell	Nick Tropiano	Quantegy 499	EP Mixing
Wombat Rd Recording Studio (03) 5145 4204					
Mental Disaray	Demo CD	Barrie Clissold	Band/Clissold	Quantegy GP-9	Tracking/Mixing/Mastering
Flatten Before Disposal	Demo CD	Barrie Clissold	Band/Clissold	Quantegy GP-9	Tracking/Mixing/Mastering
Rick Hodge	CD Single	Barrie Clissold	Barrie Clissold	ADAT	CD Production
East Gipsland Tafe	Demo Tape	Barrie Clissold	Barrie Clossold	Hard Disc	Tracking/Mixing/Mastering
Young Voices of Sale	Mini Album (CD)	Barrie Clissold	J Ward	Quantegy DAT Tape	Recording & Mastering

STUDIO UPDATE

JMF Recording Studio Has a new Website address ***jmf.mainpage.net***. Four recording formats are available for recording, ranging from 2-inch analogue, x850 1-inch digital, ADAT to CBX D5 hard disk recording. JMF recording cater for the needs of their clients in a comfortable studio atmosphere. Large live room, vocal booth and drum booth. Visual contact with all musicians. Sony Studio: Melissah Kochel has been appointed as the new studio coordinator 9383 6461. Trackdown Digital has bought a new Digidesign universal slave driver and has appointed Mike Duffy as their new studio manager. Velvet Sound has played host to international producer Charles Fisher, who has booked Studio B for fifteen months to work on three major projects. The first was a new album for Australian vocalist Glen Bidmead, followed up by an album for Scandinavian artist 'Yeska', and finally, Canadian group 'Indecision' – which he co-produced with fellow international engineer/producer Femi Jiya. The album was mixed at Pacifique Studios, Hollywood by legends Chris Lord-Alge and Tim Palmer. The project, for Universal Music Canada, is destined for release this Christmas. On the new gear front Velvet Sound has taken on two Avalon 737-SPs, Event 20/20 powered monitors, a new ProTools24 MixPlus v5.0 system, which features 24-bit resolution, over 64 audio tracks, and is fully SMPTE linked to Logic Audio Platinum with no 'lag time'.

Dex Audio, Melbournes first CD Mastering and audio restoration facility has several unique additions to their existing custom analogue mastering gear. Modified 1950s vintage valve compressor/limiter/line amps, named the 'Gedo Zen' now complement the existing modified AD/DA converters, the industry standard Sonic Solutions with No Noise and 52-bit DSP Sony digital effects systems. The Gedo Zens were redesigned by Co-Director and analogue guru, Daniel Desiere and feature beefy +34dBm output stages and greater than 100dB dynamic range. Unlike modern 'tube' processors, most of which are primarily solid state designs, these are the real deal, with super quiet, extended likfe valve circuits, transformer coupled I/O stages, stereo linked VCAs and enough headroom to blow the meters off many top name brands of pro audio gear! Used on recnet projects for The Push and Melbourne bands Dish, Second Honeymoon and the Dancehall Racketeers, they are proving as popularl as the original Leak Varislope valve preamp/EQs, which were drastically cleaned up in terms of S/N and dynamic range.

Suite 1 at Dex Audio

Sample CDs

Killer Horns 2

The first two sample CDs reviewed here are from Bestservice. These guys distribute a plethora of titles covering a wide range of styles and genre. The first CD off the rank is Killer Horns 2 by Albie Donnelly. For those who don't know, Albie Donnelly was the sax player and bandleader for the English '70s group, Supercharge. Supercharge toured Europe and Australia over the years as a support act for Queen, (viva la Freddie Mercury! [can you say that about a dead bloke? – CH]). Albie has also worked with the likes of Bob Geldof, The Boomtown Rats and Graham Parker. Albie has recorded this CD for anyone who's tried to capture a bold and brassy feeling, given up and called for session musicians.

The CD opens with sax section chords: major and minor 7ths, 9ths, etc. Then it goes into full brass section chords and harmony riffs in the four predetermined keys of B flat, C#, E and G. Then there's a tenor sax playing diminished scale riffs, solo trumpet, trombone effects, soprano sax riffs at 174bpm (these are quite cool), another bunch of full section riffs, trumpet falls, rises and riffs, another batch of solos by various horns, and then all the instruments in a multisample format.

Albie points out in the sleeve notes that the instruments were recorded at Parr Studios in Liverpool using a Neumann U87 on trombone, a Neumann U87 for trumpet and a U47 for the saxophones. He also outlines the method used to record these ensembles. Directly from the mics to a 20-bit A/D converter into a Soundscape editing system, so the recordings have never seen tape. He's also left the recordings unprocessed or EQ'd, so you know you're dealing with the original performances. The recorded sections are also positioned from left to right as trombone, trumpet, trumpet, sax so you can get the vibe of a true brass section pumping away up the back of stage.

It's really quite a good collection – well recorded and edited – and guaranteed to give your horn stabs and wails the sort of spice and authenticity required. *Brad Watts*

East West Sounds: (02) 9698 8466
Formats: Audio, all others
Price: $149.95 (Audio); $299.95 (CD-Rom)

Orient Odyssee

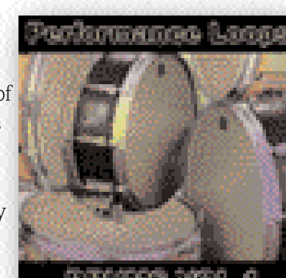
The second CD from Bestservice here is again in the audio format and is dubbed (and doubtfully spelt) Orient Odyssee. This is a collection of riffs, grooves, vocals and atmospheres with a Turkish flavour. The instruments used in the recording are listed on the sleeve but, quite frankly, are of not much use in their identification – I haven't heard of any of them. Tell a lie, there are bongos, but the rest of the sounds are delivered by the likes of darbukas, emrekanunds and curas, for those who have any idea what I'm talking about.

The CD is sensibly laid out, starting with percussion grooves at tempos ranging from 83bpm through to 140bpm. If you've ever tried to use traditional middle eastern rhythms in contemporary anglo music (i.e. 4/4 time signatures), you'll be glad to hear these loops all have that eastern feel but in 'four on the floor' time signatures. Very useful indeed.

The CD then moves into plucked instruments like tanburs and sazes [now you're just making it up – CH] then into string licks and phrases, oriental woodwinds and flutes, and then a succession of voices and choirs in both male and female persuasions. The disc finishes with a collection of Turkish and Byzantine ambience recordings – marketplaces, baths, of course, and a teahouse. Fabulous! I found Orient Odyssee to be a very useful CD, and you will too, if indeed these are the sorts of sounds you're after (and probably even if you're not). In fact, I lent this disc to an older musician friend of mine. This chap played throughout the '50s and '60s with the cream [should that be the tzaziki? – CH] of Egypt's Arabic musicians and I can't get it back off him. It must be authentic! *Brad Watts*

East West Sounds: (02) 9698 8466
Format: Audio
Price: $149.95

Performance Loops Drums



The concept behind this two CD set from Big Fish Audio is the creation of loops from a series of performances by session drummers recorded at studios in Seattle and LA. There are over 1000 loops and fills arranged by tempo, and consecutive loops are pulled from a single performance, giving several variations and fills for each groove. The CD is further divided into three sections named Artist One, Artist Two and Artist Three representing different recording processes. Artist One is a collection of different drummers on different kits and provides the most variety of style and tone in this collection. Artists Two and Three are each a single drummer on one kit playing many variations of fairly similar beats, with neither separation of parts nor single drum hits – which is a strange ommission. The result is a clean and consistent sound, ready for treatment with filters and effects, however, if the raw kick or snare sound isn't suitable for your production your only option would be triggering another sound or some time-consuming editing. I can see Performance Loops Drums' sounds and the grooves would suit those looking for clean, consistent beats for pop, jazz and latin styles.
Brad Watts

Sample Soundhouse: (02) 9973 2832
Formats: Audio, Akai, Roland and Samplecell
$169.95 (Audio); $249.95 (CD-Rom)

PC Audio

The Millennium Bug – major disaster or damp squib? Martin Walker examines the prospects.

The mainstream media have devoted countless column inches to calamitous events that may or may not occur due to the Millennium bug. In the process they have generated so much misinformation that the average PC owner has either shut off completely and ignored the whole thing, or is already camping in a cave with stocks of canned soup and rice pudding! So, is there really anything to worry about? Well, yes and no. (You knew I was going to say that, didn't you?)

The main reason for the bug is that the RTC (Real Time Clock) used inside all PCs since the days of the 286 processor was simply never designed to cope with changing centuries. The year information is stored as two digits, so that, for example, 1948 becomes 48, and 1999 becomes 99. When midnight strikes on the 31st December 1999 most RTC chips will happily roll over to 00, so they think the new date is 1st January 1900.

Thankfully the tiny amount of memory used by the BIOS (Basic Input Output System) chip to store the current information whenever you first switch on your PC has space for a four digit year, so in many cases you can simply type in '2000' when the time arrives, and your clock will then be correct for another century. However, most BIOS chips since 1995 will roll the year over automatically, so the first time you switch on in the new millennium the year 2000 will already be displayed. However, a few won't be clever enough to cope if the PC is actually running during the changeover, and will need updating by hand.

The quickest way to check your PC is to download one of the many free Y2K utilities, such as:

http://www.y2000fix.com/download/pc2000.zip

from Year 2000 Consultants. This will test your RTC, BIOS, and Windows for correct rollover for the year 2000 and subsequent leap years, so you'll know whether you have to change the year setting manually or not.

However, very few Windows applications that need the date actually get it directly from the BIOS or the RTC. Instead, Windows reads the date and time from the BIOS value as part of its bootup procedure, and then carries on updating this initial value by itself as long as the PC remains switched on. Thankfully Windows (3.1, 95, 98 and NT) also knows about the Y2K problem, and will interpret the date correctly even in the very unlikely event that your BIOS or RTC have insurmountable problems.

The Mac doesn't suffer any of these anomalies, but does enter the picture where software compliance is concerned. Any software that uses the date in any way has the potential for Y2K problems, which may range from incorrect calculations in spreadsheets and databases, to a total refusal to run after 1999. The problem is once again due to storage of dates in a two-digit form (this time internally in the software) such that a value of 17/05/02 might be interpreted as either 17th May 2002 or 17th May 1902. As you can imagine, financial packages would react very differently with each interpretation, so it's very important that you check such applications for Y2K compatibility before the end of 1999.

Thankfully music applications are unlikely to use date information, but there are still possible issues with the timing of automatic backup functions (like those of Cubase) and installation and authorisation routines (some of which may be time sensitive). You should therefore still be a little cautious unless you get an assurance from the software developer that there are no Y2K issues. Frankly, I was a little surprised at the lack of Y2K information on most music developers' websites. Emagic was the only one I came across who seemed proud to display its full Year 2000 compliance in a prominent position. Steinberg had a couple of sentences tucked away in the Knowledge Base to similar effect, and Cakewalk had no information at all (although they have in fact tested their products, and there should be no Y2K problems). However, I personally doubt that many musicians will experience any such problems, apart perhaps from the odd rogue utility written years ago.

If you want a utility to check for software compliance then both Norton 2000 and McAfee2000 Toolbox will scan your hard drives for applications, and then check these against a constantly updated database of Y2K compliance. Sadly there are few music developers represented in it, but at least you should be reassured about the bulk of your mainstream software, and can download any required updates.

There is one final aspect that needs to be discussed, and this is the main source of worry for many organisations with large amounts of existing data. Despite the compliance of the host application, there are still a number of possibilities for users to have already entered data in non-compliant ways, by specifying years with two digits in their own data, or, even worse, by setting up calculations in spreadsheets or databases that rely on there being only two significant digits in the year.

The unfortunate thing is that sometimes there is no easy way to check for this, other than by re-calculating the answers by hand – this is how many companies have been checking their own data during the last year or so. However, Norton 2000 will scan all common database and spreadsheet data files for such possibilities (including Excel, Access, Lotus 1-2-3, Quattro Pro, dBase III, dBase IV, and Paradox formats). It includes the Fix Assistant to guide you through the most common Excel problem areas, but with all other formats you are on your own. If you have a database of customers held on your PC then you should be checking your data for compliance now as a matter of urgency. As for the rest of us, we'll be stocking up with more cans of beans and beer!

Mac Audio

News of Apple's OS9 and G4 is all but ignored as Brad Watts goes web-snorkling for freeware.

There have been a couple of exciting developments for Macintosh users recently. Firstly is the impending release of Mac OS9. At the time of writing, I'd only seen a version running at an Apple press launch, so I can't relay too many first hand experiences. But, OS9 does have a few tasty little extras for internet work, and Apple has continued to improve the system. For example, being able to have different users 'log in' to the system could be useful for studios wanting to keep a machine in one configuration for clients and in another for more personal computer work. All up, though, I didn't see anything that would see the entire Mac audio community stampede for a copy.

Apple news to actually jump up and down about is the release of the G4 machines. The G4 really turned some heads at the afore-mentioned press junket. Less than a year after the release of the Blue and White G3s, we've seen G4 status occurring – as we all know, things just keep getting faster in the computer industry. Alas, I've merely had a quick mouse around with a G4 so it's probably best left to be covered in the next issue of AT, in a full review.

This issue I thought I'd direct your 'mousie pointy thing' to some of the worthwhile, and often indispensable, free software available for the Mac platform.

Our first stop is at the homepage of Audio Ease. This company is responsible for the only software I've found to do a decent job at sample rate conversion: Barbabatch. But Barbabatch isn't free. What Audio Ease will give you is a little application called Make a Test Tone and Thonk. I've often found the need for good quality test tones (I'm sure that's a story in itself) and Make a Test Tone is brilliant. It will let you create audio files of test tones at any sample rate and level – either sweeping from one frequency to another or as a straight tone. Files can be saved at 16- or 24-bit and as either AIFF, SD2, WAV or the Paris file format. The next fabby bit of freeware from Audio Ease is called Thonk and is a godsend for any sound designer. Thonk takes an already recorded file, and by using granular synthesis, turns it into nothing like the original file! One for the middle of winter I reckon.

Another must-have is SoundApp. This application will convert just about any sound file into any other type. AIFF, SD2, WAV, Sun, NeXT, Amiga MOD files, Quicktime, Mac System sound format and PSION – it'll even rip files from audio CDs. A very handy tool that every Mac audio person should get and it's all over the web, just run a search for 'SoundApp'.

Another little known but quite comprehensive piece of freeware is TWE or Tiny Wave Editor. Don't let the title fool you, this app is quite a comprehensive tool. TWE is made by Yamaha for dumping samples to and from your CBX-D5, A3000 or EX5/EX7 Yamaha sampler (probably why it's not often heard about!). Worth checking out. Not only is it a very usable and stable program but it's damn well written. TWE reads directly from disk (doesn't have to load its data into RAM) and performs fades, normalising, cross-fade loops and a whole bunch of other essential audio related tasks. It will also deal with any sample or bit-rate and it even lets you assign your own key commands for pretty much every function. If editing without piles of plug-ins is what you do then TWE could save you paying for a two-track editor.

Another great freebie is Prosoniq's sonicWORX Artist Basic. Prosoniq is responsible for some great software. One of the company's founders, Steven Sprenger, was in fact one of the main developers of Steinberg's VST plug-in regime. SonicWORX Artist Basic is a 'cut down' version of sonicWORX Artist. While 'cut down' may not seem like the full kahuna, it's still a very powerful piece of kit. All the mandatory DSP and editing functions are there, like in Yamaha's TWE, but sonicWORX Artist Basic has other neat stuff like phase vocoding, pitch and time scaling, and reverb. Prosoniq actually designed the time and pitch algorithms for Logic Audio's 'Time Machine', so it's likely you're getting a lot of bang for no bucks at all. While you're at Prosoniq's site you can also download the Northpole Resonant Filter, a VST plug-in that gives you a resonant filter with an envelope follower and a digital delay. Not bad for nix.

So there you go, there's a wealth of stuff out there to keep you editing away at professional sample and bit-rates while you're designing sounds yet to be heard by the human ear. Happy and cheap editing!

Audio Ease – http://www.audioease.com
Yamaha TWE –
http://www.yamaha.co.uk/synth/download/current/a3000/twe200.hqx
Prosoniq – *http://www.prosoniq.com/*

Rupert's word

Rupert Neve

Noise and Bandwidth

I've just returned from the New York AES Convention. It was a good show for me, with the US launch of the AMEK 9098i console and the new Pure Path outboard range. The stand was crowded solidly – I could not get away for even the shortest break, let alone see what others were doing. I spent my time between presentations talking with enthusiastic owners of vintage Neve consoles and disenchanting those whose consoles I did not actually design! (My last design for the original Neve Company, in 1978, was for the famous three consoles at George Martin's Air London Studios).

I can't remember how many such shows I've been to, but I do remember the days when professional shows on both sides of the Atlantic were quiet and dignified. There were reserved professionals; restrained owners, directors, and salesmen in dark suits; engineers and lesser mortals in grey flannels and sports jackets (Harris Tweeds for the more senior chaps); short back and sides haircuts; discipline, order and design: All were seekers after knowledge, and those anxious to impart it did not disappoint us.

Crowds? Yes! But you could sit down with another engineer and discuss real engineering specs. Noise? Not much, as I recall. No demos – no loudspeakers allowed. There seemed to be more time and there were certainly fewer choices: three tape machines, three consoles, three microphones. Oh yes, and there's that crazy chap trying to sell ready-made mic leads – obviously he can't succeed, we always make our own.

Today the dominant impression at these shows is that of noise! It's difficult to hear or to make yourself heard. What does it reflect? The shows used to be for us. Who are they for now?

AES shows are comparatively restrained in comparison to certain others. Ever been to one around the Pacific Rim? Noise levels approaching the threshold of pain. Conversation is possible only with lips three inches from listener's ear! (Something to do with Inverse Square Law, remember?)

I find that thinking time is at a premium. The best ideas float below the surface and don't clamour. They have to gestate and emerge, get massaged and grow… well, you get my meaning, I'm sure. But noise is the order of the day. I'd go as far as to say that noise is the curse of our generation. Every product demo is characterised by noise: car chases, crashes, explosions, jet aircraft supposedly in your lounge. There's thunderous single note bass from the car beside you at the traffic lights, bouncing its wheels off the road.

Noise suggests chaos in the sense of multiple signals clamouring for our attention. Deep down, embedded in the discord, is the stuff we want. It just has to be sorted out, which is often hard work.

Back in 1992, I spent many a late night driving 200 miles back and forth between my home, in Suffolk, and the offices of my client AMEK, in Manchester. Tuning around the FM band I came across a mysterious transmission: birds, quiet woodland sounds, country atmosphere, a cuckoo with that strange distant reverberation, occasional sounds of a distant car or aircraft. A scene familiar and well remembered, drawing on memories of Summer woodland walks, striking a note of peace and tranquillity. Nothing strident. No voice. No music. Just nature enveloping me in reasonable two channel car stereo. I listened for hours. Tension and worry departed and the journey was shorter. Classic FM test transmissions prior to launch! I found many others had the same experience. Everyone was sorry when, after some weeks, the tests were over. I'm sure those tapes are saleable. Where are they?

This is not just philosophical stuff. I'm reflecting on what it is that we're really looking for: satisfaction in a listening experience. To attain satisfaction, part of the process is to discard the noise! Beneath that noise there may be stuff we are only recognising subliminally.

"Forget it, Rupert," I hear you say; "If I can't hear it, it's not there. It's below the noise level!". So what is noise? The dictionary says:

Noise: In physics, an acoustic, electric, or electronic signal consisting of a random mixture of wavelengths (see Sound). In information theory, the term designates a signal that contains no information. In acoustics, 'white' noise consists of all audible frequencies, just as white light consists of all visible frequencies. Noise is also a subjective term, referring to any unwanted sound. Noise pollution (see Airplane) is a serious environmental problem, particularly as sound levels above a certain intensity can be physically damaging.

It might seem that noise must be all bad. Noise is really a necessary component of the beautiful and satisfying sound that we actually do want to hear; it's irrevocably linked to overall performance. There is always 'something' that is not strictly part of the performance. It's not all bad. Without that 'something', there is no realism. Noise is part of the world around us; it makes sound believable, 'realistic', if you like. Music in a sound proof and acoustically dead studio without noise is clinical, brittle and unsatisfying to artist and listener alike. How often have you heard a chamber music reviewer say that he can hear the artist breathing? Is he criticising or authenticating the performance?

Then there's audience noise, which helps both the artist and the audience in appreciation of a performance. 'Someone' is listening and 'someone' is never completely silent. From thunderous applause to a rapt silence that breathes. How much is right?

Reverberation is a form of noise, usually not part of the sound directly generated or controlled by the artist. In a hall or in a studio, it is needed – not too much; just the right amount to set the scene. How much is right? It's a value judgement made by a human, and here, of course, noise is specifically related to the original sound.

In my last column [AudioTechnology Vol. 1, Iss. 5] I identified sources of noise which are definitely in the "noise is bad" category. Studios, microphones, and amplifiers. Let me remind you.

Electrical noise is inescapable. Every electrical conductor exhibits motion of atoms which generate small voltages of random frequency. This voltage depends upon the resistance value, the temperature, and bandwidth. Here is the classic noise formula:

SQRT(4*1.39e-23*T*R*B/W)

for which we are indebted to.

Boltzmann, Ludwig (1844-1906): Austrian physicist, who helped lay the foundation for the field of physics known as statistical mechanics. Boltzmann was born in Vienna and educated at the universities of Vienna and Oxford. He was a professor of physics at various German and Austrian universities for more than 40 years. During the 1870s Boltzmann published a series of papers that showed that the second law of thermodynamics could be explained by statistically analyzing the motions of atoms.

Boltzmann's work was strongly attacked by scientists of his time. However, much of his work was substantiated by experimental data soon after he committed suicide in 1906.

Using Boltzmann's formula, at a temperature of 25° Celsius (approx. 300° Kelvin), the voltage produced by a 10,000 ohm resistor measured over a 20kHz bandwidth is 1,820.37nV, which in more familiar terms is -112.6dBu.

If the bandwidth is increased by four times to 80kHz, the noise voltage is doubled. Here we can see a link between noise and bandwidth.

Having a frequency response way out there to dog-land means more bandwidth and, therefore, more noise. It opens the window to more of those minuscule high order harmonics and switching 'splats' generated by all but the best equipment.

Is it important? Yes it is, because it also opens the window to those minuscule musical harmonics which are a true part of music.

Can you hear it? No, you can't!

"Forget it, Rupert," I hear you say again; "If I can't hear it, it's just not there!".

So, what am I trying to sell you on? Something that's only a bee in my bonnet? You've read my views on the need for wide bandwidth beyond audibility [AudioTechnology, Vol. 1, Iss. 1]. Surely if it's beyond audibility we must be wasting our time! Human hearing usually cuts off somewhere between 12kHz and 18kHz, depending on age, health, educated listening experience, and machismo.

Try these two simple experiments which I have carried out with audiences in many parts of the world using no special equipment, just what was available plus a patient colleague:

Experiment #1

A sine wave is a single frequency, totally pure and free of harmonics. A square wave is extremely rich in odd harmonics: 3rd, 5th, and so on. If you generate a square wave at, say, 7kHz, you will not hear even the first of its harmonics (the 3rd harmonic), which appears at 21kHz. (Unless, of course, you're a dog!).

You need a Variable Frequency Audio Oscillator with an output that can be switched between sine and square waveforms without a large change in level (2dB to 3dB is OK).

Feed this into an amplifier and loudspeaker to produce a fairly loud Sound Pressure Level, around 80dB SPL. Start at around 3kHz. As your colleague switches from sine to square, you will easily hear the third harmonic, at 9kHz, as a high pitched 'whistle' superimposed on the 3kHz fundamental. Depending on your hearing, you may also hear the fifth harmonic (15kHz).

Get your colleague to raise the frequency slowly, switching back and forth between sine and square. At the fundamental frequency corresponding to one third that of your upper hearing limit, the 'whistle' disappears. Above this frequency you should only be able to hear the fundamental, all harmonics having been filtered out by the upper limit of your hearing.

But you can still detect the difference between the sine and square waves. Some audiences have been able to report a difference reliably when the fundamental is as high as 18kHz. We do not 'hear' these frequencies but, in some way that we don't yet understand, we are able to perceive their presence.

Experiment #2

For the second experiment you need a Pink Noise Generator (PNG) and a Variable Frequency Audio Oscillator, as before.

Feed the PNG into one channel of a mixing desk and the oscillator, at about 3kHz, into another. Adjust the two signal levels so that when the faders are at maximum, you are reading approximately the same level from each on a meter (or meters) across the channel outputs. (The PNG signal will obviously be fluctuating.)

Mix the two signals and take the resulting bus output to a power amplifier and loudspeaker, setting up a fairly substantial level as before, say, about 80dB SPL.

You will hear a most unpleasant sound and will quickly want to reduce the 3kHz oscillator signal. Use the oscillator channel fader to do this until you can hardly hear the oscillator signal. Rock the oscillator's frequency slightly back and forth as you reduce the level on the fader. (This is because your hearing develops a de-sensitising 'notch' when listening to a single frequency.)

As you continue to reduce the level, you will reach a point at which the pink noise totally masks the oscillator, and you can no longer hear the 3kHz signal. When you look at the fader scale, you will find that you have reduced the oscillator level by between 30 and 40dB. If you use a square wave instead of a sine wave, you will have to reduce the level even further.

What does this prove? It is simply evidence of the ear's ability to discriminate in favour of a 'wanted' sound in the presence of a much louder 'unwanted' sound. It also suggests that unwanted harmonic distortion has to be substantially below the noise level before it is inaudible. So much for the often heard engineer's statement: "Distortion is below the noise level!"

Next issue I'll look at the results of these simple experiments and consider what they mean to sound engineers and equipment designers. But I'd like to close this column with John Reith's Mission Statement for the BBC, which is inscribed in the Entrance Hall at Broadcasting House (translated from Latin):

"This temple of the arts and muses is dedicated to Almighty God by the first governors in the year of our Lord 1931, John Reith being the Director-General, and they pray that good seed sown may bring forth good harvest, that all things foul or hostile to peace may be banished hence, and that the people inclining their ear to whatsoever things that are lovely and honest, whatsoever things are of good report, may tread the path of virtue and wisdom."

EQUIPMENT TEST

Sony DRE-S777 Sampling Reverb

A sampling reverb? It's not what you think. Brad Watts investigates a fascinating new concept in reverberation.

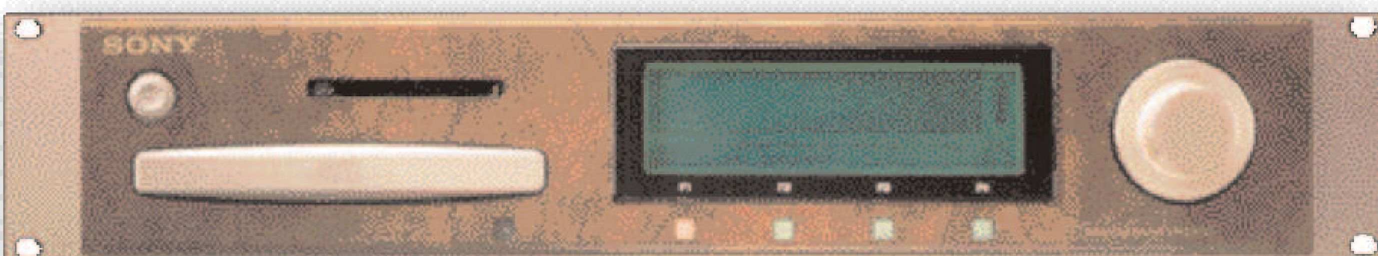
Sony are one of the big players in the audio world so I suspect they can sink obscene amounts of money into developing new angles in professional audio devices. One only has to see a Sony Oxford digital recording console to realise how dedicated Sony are to pro audio. Everything from MiniDisc players to multitrack recorders wander out of Sony's factories, and have done for many years now. (I wonder how much Sony's interest in the artist and repertoire side of the recording industry affects their professional recording products?)

With their newest product, the DRE-S777 Sampling Reverb, Sony have combined a number of technologies to realise a new direction in creating artificial reverb spaces (or should I say recreating real reverb spaces?). Sony generously lent AudioTechnology one of only a handful of these units currently available in the world, so we could give our impressions of this new reverberation method.

When I first heard of this 'sampling reverb', I must admit, I was slightly sceptical. I was thinking "oh yeah right… as if any recording engineer or studio person is going to have the equipment and room to make it worthwhile sampling their own reverbs". And what use is a sample of a reverb, anyway?

My initial visions didn't prepare me for what this 'sampling reverb' really did. The fact of the matter is I was sort of half wrong and half right (depending on which end of the glass you call full). The DRE-S777 does use sampled reverb recordings, but they're not 'samples' in the same way that a sampler has samples – they're not fixed recordings that are played back at the push of a button.

In fact, Sony's use of the word 'sampling' is more related to when a chemist takes a sample of a substance for further analysis. Because that's how the reverbs in the DRE-S777 have been created. The ambience of a room has been recorded as it responds to different forms of audio signals, which provides the reverb 'sample' data. This data is then analysed to create a 'construct' of how that room responds to differing types of audio signals.



This 'construct' is then stored in the DRE-S777, allowing it to apply the sampled room's ambience behaviour to whatever signal is passed through it.

A processor that records and analyses a room's ambience and then allows you to adapt that ambience to your own dry signals sounds like a terrific idea, but surely the end result will be too reliant on the quality of the original reverb recording? To solve that problem, Sony assembled a crack team of engineers from Sony R&D (and affiliated researchers) and sent them on a tour of great churches and concert halls known for their acoustic ambiences (stop-overs included Amsterdam, Vienna, Berlin and Spain). The reverb qualities of each church were sampled and analysed, and these are the 'spaces' found in the DRE-S777.

The Theory

The method Sony use to apply these reverb recordings to an input signal is called 'convolution'. We've all (I hope) walked into a space and clapped our hands to get some idea of the room's reverb characteristics. You clap and then hear the reflections of that clap bounce between the walls, floor and ceiling until the energy of that initial handclap dies. This clap sound can be considered as an impulse which contains a wide band of frequencies, so it's a good start towards hearing how differing frequencies will behave in an environment. Sony have taken a 2kW PA system into their chosen halls, played computer generated signals through the system, and recorded the corresponding reverb. Back at Sony HQ, the reverb recordings are analysed to define how that room behaves under different signal conditions. This information is then loaded into the DRE-S777 and applied to whatever signals pass through it.

For example, let's assume a sampling rate of 44.1k. Think about how a single 44.1k sample word might sound by itself – very much like a handclap that only lasts 1/44,100th of a second. The DRE-S777 treats every one of those sample words as an individual handclap, and generates a reverb tail for each one based on the sampled reverb data taken from Sony's tour of famous churches and concert halls. So at 44.1k the device is computing a complete reverb tail 44,100 times each second (one for each sample word), and taking into account how the previous computations will affect those about to follow. As you can imagine, it requires a lot of processing.

It's a clever concept that requires the use of custom built Sony processing chips working with the data derived from the initial recordings. This is what gives the DRE-S777 the title of a 'sampling reverb', which could be slightly misleading. It really uses data collected from very good natural reverbs to compute reverb envelopes for every sample word it processes. You'll see from the following interview between Michael Gissing and Sony's Andrew Hingley that the potential exists for end users to sample their own reverbs for use in the DRE-S777, but most likely not for a few software revisions yet.

The Outside

When I first saw the DRE-S777 I was quite chuffed to see the imitation wood front panel. It gives the unit a look to suit its price tag, very much in keeping with where the DRE-S777 is aimed to compete, with the likes of the Lexicon 300L.

Michael Gissing talks with Sony's Andrew Hingley about the DRE-S777…

Michael Gissing: How would you describe the difference between synthesis reverb and the sampling reverb ?

Andrew Hingley: *The way to explain the difference is to relate it back to what happened to keyboards in the mid '80s. Original synthesisers came out 10 years earlier using subtractive synthesis, where you start off with basic waveforms and you filtered out frequencies, but then, with the Fairlight and E-mu Emulator, we saw the sampling keyboards arrive. So the easiest way to describe the difference with the DRE-S777 is to say that we are offering the same kind of change in technology as the change from synthesisers to sampling keyboards.*

MG: Except that keyboard sampling is the recording of a real sound in a digitised form, which is then played back from a keyboard command. But when we talk about a sampling reverb, you are accurately capturing the characteristics of an acoustic and using that to generate a reverb signal.

AH: *Well, it's based on a process called 'convolution'. The classic example is that, when you walk into a room and do a hand clap, you hear the reverb of the room responding to it. If you saw the envelope of that hand clap on an oscilloscope, or on a graph plotting time along the horizontal axis and amplitude along the vertical axis, you'd get a shape of the room's response to the hand clap. For the DRE-S777, we sample the room's response and feed that into our convolution 'engine'. Sampling theory says that if we then perform this convolution on an incoming signal, we will be effectively reproducing that reverb.*

If you imagine your digital audio samples as a series of spikes in the time domain, and each spike is the amplitude of each sample (represented by the numerical value of the digital audio signal), what you do is take the first sample into the convolution engine and create a whole series of samples which are the envelope of the room's response to the hand clap. Then you go forward one time period and multiply the second sample by that amplitude. The first sample has already generated a whole lot of values and then you add the appropriate values together which effectively is recreating the reverberant sound.

The difference between sampling reverbs and the more traditional synthesis reverb is very similar to the difference between synthesiser keyboards and sampling keyboards. The actual sampling process is different, but the quality difference between trying to mimic a sound using a synthesiser and actually sampling it is fundamentally on the same level. So it is accurate to say that we are sampling the room.

MG: So the only thing that sampling reverb does not accurately recreate would be the resonance of say a string instrument, played in a concert hall like the Concertgebouw, where the reverb causes the instrument to self resonate.

AH: *Well the analogy of sampling keyboards continues. If you listen to a piano sample, it sounds wonderfully like a piano, but it doesn't sound exactly like a piano. Our sampling system uses a PA and microphones with their inherent distortions. We choose where to put the speakers and the microphones so we are not trying to say that we are totally copying the reverb. We have inherent distortion in the measurement system we use, but the net result is an extremely natural sounding reverb. It sounds fundamentally different from synthesiser reverbs. But, in the same way that the piano sample depends on the type of microphone and its position, the sampling reverb's sound depends on the positioning of the PA and microphones within the room. Place them in different positions and you will get a different sound…*

MG: Yes, of course. I had the luxury of recording the Australian Youth Orchestra in the Amsterdam Concertgebouw, but my microphones and the mic placement were very different to the house microphone array. If they had lowered their mics into the recording position we would have had a very different recording.

AH: *Exactly right. Also, I think that for classical recording there is an important role for our reverb in post production. When editing 'takes' together or favouring a close mic, it would be useful to use our reverb with a sample of that room to add reverb that is very closely akin to the recorded reverb.*

MG: In a lot of mix environments the preference is to send an

auxiliary signal to the reverb unit and then mix back the wet return with the direct signal in the desk. Now given the huge processing that goes into convolution is there any significant signal delay?

AH: *The DRE is like all digital reverbs. They all have a processing time. And they all have A/D and D/A converters…*

MG: Not in my studio!

AH: *Well you are one of a small band. The DRE works in the frequency domain, not the time domain. Again this goes back to signal theory and our 'hand clap' and the reverb envelope. That description works in the time domain. But we actually work in the frequency domain because it reduces the amount of processing needed. You can do things called Fourier Transforms to move between the frequency domain and time domain. You can fully represent an audio signal either as a time sequence of samples, or as a number of frequency spectrums of the signal. Now there is some processing time required to convert the time domain signal to frequency spectrums and put them through our convolution engine, but listening to it doesn't seem to create any problems. It seems the design of the chip means that the processing time is fast enough to not matter.*

MG: There's a mixer and EQ stage in the DRE-S777. Is that EQ only applied to the reverb component?

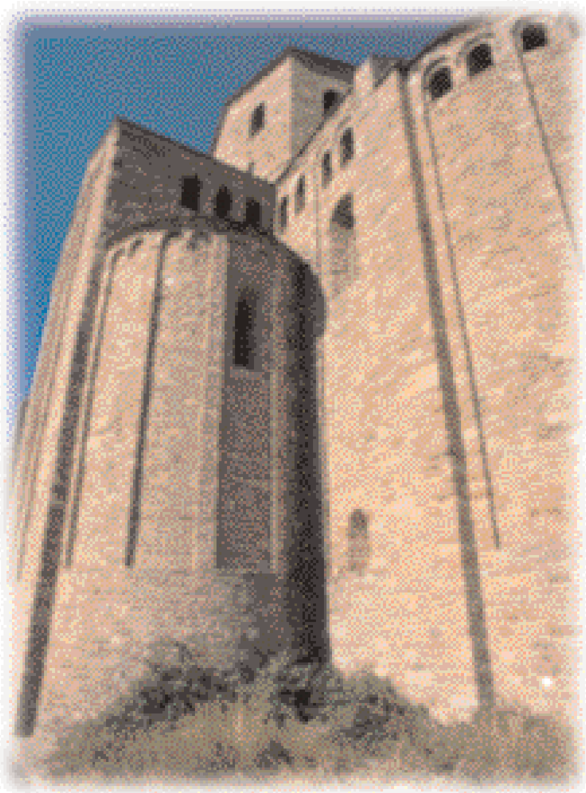
AH: *Yes. One thing we have found is that when you hear the real reverb compared to synthesis reverbs, people naturally want to add a bit of tops to the reverb returns, but the real reverbs produced by the DRE are actually much more broad band. You do get a situation where the low end can get a bit wobbly, with standing waves bouncing off the architecture and so on. We sampled a church in Spain that was built in the eleventh century, with lots of pillars around and a lot of low frequency oscillation in the reverb. Now whether that is useful or not is for the producer to decide.*

MG: So if it is not wanted, you simply roll off the bottom end of the reverb return.

AH: *That's right. The other thing is that the analogue performance of the DRE is also very high end. You can switch it to a mode where it is an A/D and D/A converter. So for a project studio it might seem a bit expensive as a reverb, but when you consider that you can also use it as a high quality stand-alone A/D and D/A converter, it becomes more affordable.*

MG: I am sure a lot of audio professionals will want to know if they can record their own samples. The Holy Grail for film sound people is to be able to match post sync sound to location sound. Unlike the music world, most of the material we deal with is recorded in a diverse range of acoustic spaces.

AH: *The DRE-S777 has only recently been made available and is running version 1.0 software. There is a plan to allow the device to do its own sampling in the future. So anyone who owns one will be able to take it to venues and, by connecting the analogue outputs to speakers and feeding the analogue inputs from a microphone, record their own samples. But it does need a lot of care when it comes to collecting samples. If the sample is no good then the box won't sound good. There is a big opportunity for film and TV people to use this reverb for exactly the application you mentioned.*



St. Vincente church in Cardona, Spain, one of the medieval acoustic spaces sampled by the Sony team.

And there is no reason why it is limited to just sampling rooms. If someone has an interesting spring reverb or plate reverb, they can sample that, too. Anything that resonates, like, say a pipe, can be sampled. It seems that we have a device that is capable of reproducing things so accurately, it isn't just room reverbs that we can emulate.

MG: Like the sound of a wire fence?

AH: *…or underwater. The test tones that we use are actually swept tones, so anything you can put transducers on or in to generate and pick up that swept tone could end up as samples in the DRE-S777.*

And, of course, we can sample any device that has analogue inputs and outputs, like synthesised reverbs or flangers/phasers, although there are some inaccuracies in the sampling process and you are probably better of using the original device.

But although the sampling process introduces some inaccuracies, and there is not much to be gained by sampling synthesiser devices, there is one thing about the sampling process that makes it better. When we take the samples, we actually take lots and lots of them and run the tests many times, like a time lapse photograph, and if you do that you can average out the samples. The actual response you want always adds constructively, but background noise is randomised. So by taking lots of samples we can average them out and get a better signal to noise ratio. We sampled a plate reverb and, by using about 16 samples, we got 10dB better signal to noise ratio than the original device. So there is some validity in sampling old tape or spring reverbs. But for sampling modern digital reverbs or flangers, the only advantage you would have is the convenience of having it all in one device.

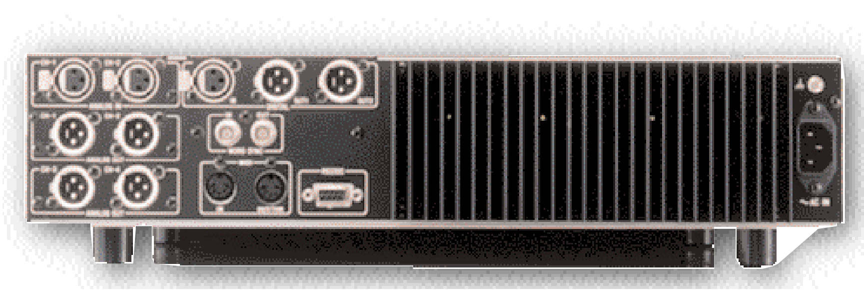
The DRE-S777 is two rack units high and has a very simply laid out front panel. On the left is a CD drive for loading the reams of data required. Above that is a PCMCIA card slot that, with the help of an ancillary card, accepts Sony's new 'Memory Stick' data storage technology. The Memory Stick is quite cute – up to 64MB of data in a card no larger than your house key.

The right hand side of the front panel is home to a large backlit LCD display with four software keys beneath it. To the right of the display is a very large data wheel that wouldn't look out of place on the dashboard of your Daimler Super V8. Overall there's not a lot to look after when driving the DRE-S777, definitely the mark of a professional tool. The unit is very deep, over 500mm, and weighs about 15kg with all the optional cards installed. Yes it's heavy, but so is a Daimler!

The Insides

You get some idea of the amount of data the DRE-S777 has to churn through after you've booted it up and found yourself waiting a good five or six minutes for all the reverb information to be loaded from the CD-ROM! This fills seven of the nine memory locations, and you get two medium halls, two churches, a 'studio' 'verb and two plates. Once you have the settings loaded from the CD-ROM it doesn't take long to find your way around, and there really isn't a lot to take care of.

Reverb time, or decay, is adjustable between 0.3 seconds and the length of the reverb you are using (determined by the room you have chosen). The only other parameter for the actual reverb is pre-delay.



The Sony DRE-S777 back panel, packing a complete set of optional I/O.

After the reverb settings is a mixer page where wet and dry signals are adjustable and separately mutable. Following the mixer is an EQ page. This is again relatively simple with a high and low shelf and two parametric mid bands. EQ can be applied to either the wet or dry signal, or applied to whatever arrives at the input of the machine.

There's not a great deal to look after parameter-wise in the DRE-S777. The spaces sound so real and spacious that these few simple parameters are all you'd want. Come to think of it, I have three or four reverbs that I use with the machines in my studio and that's about all I adjust anyway: pre-delay, reverb time, and then EQ the result back at the console. It's nice not to be burdened with diffusion, density, spin, or width functions. The DRE-S777 sits there merrily convoluting its spaces to your program material with a delightful absence of finicky options.

The Ins and Outs

With a box of this calibre you'd expect to see some top notch analogue to digital conversion occurring. The DRE-S777 is certainly not a disappointment, and I'd have no doubts about using the A/D converters for high quality analogue conversion. Signal to noise ratio and dynamic range are quoted as 110dB. In its standard configuration the DRE-S777 will run as a mono in/stereo out or full stereo device at 44.1k and 48k sampling rates. One XLR AES/EBU digital input and two outputs also grace the standard configuration. These will accept 24-bit data, but using 88.2k or 96k sampling rates requires the addition of optional cards. I don't think making 96k an optional feature will upset anyone. Most facilities are only just reaching 96k capability, and any prospective buyer of the DRE-S777 would appreciate the choice and lower cost of the 44.1/48k unit rather than pay for the unrequired 96k spec.

And the sound?

That's the boring stuff out of the way. How this reverb sounds can only be described as exceedingly real. Lexicon really do have something to worry about here. The depth of the image is quite amazing. Reverb tails don't so much as dissipate as melt into nothingness, just as real world reverb does. It's remarkably easy, with a light touch of EQ, to pull some quite staggering spaces.

This really is a new angle for reverb processing and I think the method will be as revolutionary as the 16-bit sampler uprising of a few years ago. I won't be rushing down to the bank increasing my credit to buy one because it's just a tad outta my league. But it has been awfully splendid having one of these rare (at the moment) machines hanging around the studio. For an in-depth look at where the DRE-S777 will find its vocation, check out the following interview between Michael Gissing and Sony's Andrew Hingley. In the meantime I'm getting my microphone technique ready for what could be a very worthwhile direction in effect and ambience processing.

Distributed by

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Price

• $8600 (DRE-S777);
$1600 (DABK-S701 A/D converter board);
$1100 (DABK-S702 D/A converter board);
$3100 (DABK-S703 Expansion DSP board, stereo input, 4-channel output, 88.2k or 96k Fs);
$1150 (sampling reverb software).

Røde Classic II

Konrad Skirlis takes the latest generation Rode Classic for a ride.

The Rode Classic II is a large diaphragm capacitor microphone with a dual pressure-gradient transducer, assembled in a side-address orientation. Developed from the original Classic (now out of production), the Classic II features improvements in diaphragm and circuit design while maintaining the retro look of its predecessor. Its large cylindrical shape is easily identifiable, making a lasting impression on anyone who lays eyes on it. The Classic II's classy appearance is derived from a semi lustrous nickel finish on a hand-polished solid brass body. The wire mesh on the head grille surrounds a finer internal pop screen – the overall look and feel is solid. The large dual one-inch diaphragm assembly is gold-sputtered (six microns thick), while the entire capsule is hand assembled. The new edge-terminated diaphragm has no centre connection and is therefore 'free floating', contributing to a better performance than the original Classic. The tube is a 6072 (twin triode tube) which is shock-mounted and matched to an internal/output transformer designed by Jensen. Like the original Classic, the trademark gold dot below the head grille indicates the front of the mic.

Classic II Power Supply

The Classic II's power supply includes a thick multicore cable 10 metres long that terminates in gold-plated custom made connectors. On the front panel, three large knobs control the mic's high-pass filters, polar patterns, and attenuation. Filtering options are flat, -6dB or -12dB/octave roll-off at 125Hz. There are nine polar patterns in total! These are adjustable from omni through cardioid to bi-directional in nine steps. This gives you three intermediate patterns between omni to cardioid and between cardioid to figure of eight. Both the roll-off filter and polar pattern controls switch positions silently. The mic can be padded 10dB or 20dB, or left with no attenuation at all. Care must be taken that monitoring is at a low level when switching pad settings. Overall, the front panel is well designed with all functions clearly labelled and knobs spaced apart, allowing easy control. A blue front panel LED lights when the supply is powered, which is handy as the power switch on the rear-panel is unmarked. Also on the back is a standard three-pin XLR (pin two hot) output and an earth lift to get you around earthing problems.

The Classic II's specifications are what you'd expect from a valve microphone. Sensitivity is rated at 13mV/PA which is typical. Maximum SPL is 130dB but can handle 150dB with the 20dB pad selected. This offers considerable SPL handling capabilities making it appropriate for use on drums. The wide frequency response is 20Hz to 20kHz with the flattest response in the figure-of-eight position showing a slight presence around 5kHz; the cardioid and omni position display a slight boost around 10kHz. The self-noise is rated at 22dB and is an improvement on the original Classic's 24dB rating. With tube circuitry, it's almost unavoidable to have a higher inherent noise level. Therefore, with more ambient recordings, care should be taken on critical applications – the Classic's self noise may be audible when used on low level acoustic instruments.

The most attractive feature of the Classic is the range of polar patterns available. With a total of nine different settings, many different spatial and timbral possibilities are available – a refreshing change from the multitude of fixed cardioid mics that have flooded the market of late. The figure-of-eight pattern is the flattest, while the omni mode has a slight boost in the upper mids around 2kHz with an additional boost around 11kHz. The cardioid mode also displays a slight boost in the upper midrange, rising slightly to 3dB to 4dB around 11kHz. The rear side of the mic in figure-of-eight mode offers the 'thickest' sound and can be put to creative use by fattening an otherwise 'thin' sound source.

The Rode Classic II microphone, power supply, cable, cradle and mic adapter come packaged in an attractive foam-lined aluminium flight case which is lockable. Using the supplied cradle will offer protection against vibration in the studio. The cradle is sturdy and will not fall apart with repeated use. You must be careful to lock the cable to the mic in order to secure it onto the cradle. It will then be safe to adjust the cradle without the mic falling out. Miking vocals in cardioid mode at about six inches seemed to give the richest texture. The Classic II's lower frequencies are big, yet the high frequencies are quite detailed. For a tube mic the Classic II appears fuller in the lower midrange sounds. Even spoken word sounds bigger yet maintains detail. However, I did find that natural sibilance was slightly emphasised, but at the same time fattened up. On louder vocal performances, the Classic II handled the dynamics without audible distortion. Midrange frequencies were enhanced with up front vocals, but, in essence, the integrity of the performance was maintained. I can see most people using the Classic II for vocals but other instruments should be considered.

Recording acoustic guitar with the mic positioned about 15cm away and angled towards the edge of the sound hole with a cardioid polar pattern produced rich overtones and generally created a fuller acoustic sound. When miking a guitar amp, I found that by switching between patterns I could come up with the right blend between the room and amp sound. So by offering in between patterns, more control is possible without necessarily resorting to equalisation – thereby promoting greater phase coherency. By utilising the thicker mids on the rear side of the bi-directional pattern a Fender valve amp was made to sound astonishingly good with the Classic II placed about 10 inches away at a 45 degree angle. The amp remained detailed yet full with ample 'body'. Working the many available patterns on the Classic II provides a variety of possible overhead drum sounds that would not otherwise be an option with fixed or traditional three-position studio condenser microphones. In general, one can control the overall openness of a drum sound simply by switching between patterns! The Classic II was able to handle a kick drum with the pad switched to its 20dB setting. Again, by switching between cardioid and omni I found an open sound that was appropriate to the recording. The low frequency output was enormous! The Classic II's self-noise was at times audible during quiet passages but I feel that this is the trade off you make for 'fatness' from a valve circuit. The majority of noise lies in the low frequency region which is often rolled off anyway. In general, the Classic II can fill out a sound and maintain timbral integrity.

Classical Studies

The Rode Classic II is indeed a rich and full sounding microphone. The cost of manufacturing a quality vacuum tube is not cheap, yet Rode has not only improved on the original Classic mic design but made the Classic II affordable for the valve mic that it is. The Classic II's construction is solid and its performance won't let you down. Accept the higher noise of a valve circuit and rejoice in the Classic II's performance values – rich and detailed. The total package includes power supply, power lead, cradle and mic stand adapter all presented in an aluminium case. With an abundance of rich valve sound, the Classic II has the capacity to fatten anything it comes across while maintaining favourable valve specs for a large diaphragm valve condenser.

Distributed by

- *Rode Microphones*

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Rode on WWW: 'www.rode.com.au'

Price

- *RRP: $2194*

IKonrad Skirlis corners Rode boss, Peter Freedman, for more of the story behind the Classic II.

Konrad Skirlis: Rode's success has been built on a reputation for good value for money. Does this perception make the Classic II a 'hard sell' in the professional recording market?

Peter Freedman: *Well, its interesting you should define it like that. It's undeniable that Rode, now a global brand, achieved that status in no small part to us breaking the price barrier. When we started, you had to pay thousands for a good recording mic, we changed all that. Our initial success however was due to two major factors. Firstly, the mics sounded great! We built the initial buzz around L.A studios (Rode's first export success!) because of the sound quality. The U47, the C12 had that sound, but the electronics and acoustics boffins working at the established firms had managed to get rid of this pleasant distortion! We listened to what the musicians and recording engineers said, and then gave it to them. If you add the fact that we were one third to half of the price you would normally be forced to pay, you can see why it exploded. The funny thing is we led the way! These Australian nobodies forced the rest of the industry into changing their ways.*

The Classic I was released after we had broken into serious world-wide distribution. The customers literally asked us to develop it. It was our attempt to recreate a valve mic that would stand among the best in the world. We have sold over 3000 of these, which is pretty good, and there are so many major records where the vocals were cut using them- we're extremely proud. The Classic II was released at the October 99 AES in New York. Why change the Classic? Well, because we can! We can do whatever we want! If we find a better way of doing something technically, or we see we can produce a better product, which offers our customers more, we will. We are not held back like some large companies that have to continue with something just because they spent a lot on R&D money on it. Rode products are evolving. We recently finished a tour in the States. We visited a lot of major studios and dealers. Every time we offered to do a mic shoot-out and without fail on each occasion we were asked to go up against the big money esoterics. This was obviously an effort to kick our heads in. We blitzed them every time! The Classic II is a very special microphone. So, to make a short story long! we don't find any resistance to our mics, we are selling as many as we can make.

KS: It says, 'Made in Australia' on the box, but which Taiwanese sweat shop has it really come from!?!

PF: *Okay, you got me! We don't really make them at all! Seriously, there is a basis for this rumour. When we started Rode microphones about nine years ago, we did it with a mic we sourced from China. We pulled the thing to pieces and rebuilt it using good electronics. This was the first NT1. That mic did really well here and it was the reason we jumped into the whole industry. Since then, however, we have built up a local manufacturing facility that would equal any of the major brands. We have invested over A$5 million in the facility so far, and will do the same in the next two years. We can now manufacture microphone parts at a lower cost and at a higher quality than they could be bought. This is all due to the sales volume we have managed to achieve. We buy our electronic components from all over the world, which we must because Australia just doesn't make electronic components. We have a very well equipped metal work facility, which produces tens of thousand of bodies per year. We have Australia's only microphone Transducer R&D Lab, which is currently located at the CSIRO, this has already produced some interesting capsules, and we look forward to releasing some revolutionary technology soon.*

KS: The edge-terminated diaphragm is a new feature of the Classic II – what are the technical benefits of this design?

PF: *The edge-terminated capsule is one of the major changes we have made. While this style of capsule has been around for many years, and has been used in some of the legends (AKG C12 for example) our implementation is different. We spent a great deal of time developing this new transducer, and used some quite clever analysis gear in the development. One analysis process employed is called laser interferometery. This allows us to see the diaphragm in action as it responds to the audio spectrum. I could tell you what we found and how we incorporated that information into our new transducers, but then I'd have to kill you. Instead listen to the Classic II or NTV, and you will hear the results.*

KS: The Classic II's manual refers to "refining the electronic circuitry" – exactly what refinements have been?

PF: *The changes are major! The whole Classic circuit was re-assessed by the head of our audio engineering division, Doug Ford. Doug joined Rode about three months ago, and anyone familiar with the Australian audio industry will know of Doug. Without doubt he's one of the finest analogue engineers there is. Totally mad (In a good way), but then that's the audio industry! Anyway, the major differences are the operating voltages and currents at the parallel triode sections. These have been very carefully optimised for lowest noise. He used a bipolar voltage-follower stage after the tube, which gives low source impedance to the hi-pass filter and transformer. It also presents a high impedance to the tube, dramatically reducing distortion and allowing the 'sound' to be defined only by the tube, and not by the transformer or uncontrolled reflected load impedances.*

KS: The Classic II gives you two high pass filter options – what is the nominated frequency and dB/octave slope for each of these?

PF: *Both filter shapes are fairly gentle (i.e. cascaded first-order filters). The 'mild' roll-off is 3dB down at 100Hz, 10dB down at 35Hz and -20dB down at 12Hz. The 'steeper' roll-off is 3dB down at 130Hz, 10dB down at 48Hz and 20dB down at 22Hz.*

KS: What's the life expectancy of the 6072 tube?

PF: *The manufacturers' specified life is 2000 hours. Actual life will be much longer than this, because of our slow filament turn-on, and delayed HT turn-on as well as relatively low anode voltages and currents.*

KS: Technically speaking, how does the microphone pickup pattern change between omni, cardioid and figure-of-eight?

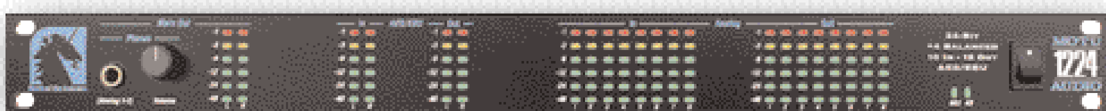
PF: *The mic capsule has two diaphragms. Each one exhibits a cardioid response – one to the front, and one to the rear. The acoustic contributions of the diaphragms can be remote controlled by controlling the DC polarising voltage to the rear diaphragm relative to the front diaphragm.*

If the rear DC voltage is the same as the front, the two cardioid patterns add to form an omnidirectional pattern (ie. pressure mic). If the rear DC voltage is zero, there is no contribution from the rear diaphragm and the response is cardioid. If the rear DC voltage is negative and equal to the front, the two cardioid patterns subtract from each other to form a figure-of-eight (velocity) mic pattern. We have a total of nine polar patterns on the Classic II. A lot of people think this is overkill, but in use the flexibility of being able to widen or narrow a polar pattern can have dramatic effects. At a recent session I was on at Westlake Studios in LA, the engineer blew everyone away when he adjusted the Classic II while we were setting up to record a vocal track. The sound changed so much as he pulled in more 'room'. Natural EQ! That's when you see why these guys get paid so much and why these mics are made with these features.

MOTU 1224

Scott Christie finds the new MOTU 2408 companion stands up well on its own.

Following on from the release of the MOTU 2408 audio interface [AudioTechnology Volume 1, Issue 4] comes the MOTU 1224 from the American company Mark of the Unicorn. Both the 2408 and the 1224 provide an audio input/output solution to computer-based hard disk recording systems. However, whereas the MOTU 2408 was designed specifically to provide extensive interfacing with digital tape-based systems such as ADATs and the Tascam DA series multitrackers, the MOTU 1224 is designed squarely at interfacing with the world of analogue mixing consoles. The MOTU 1224 is largely promoted as an expansion option for the 2408, but it is important to note from the onset that the 1224 can be configured as an independent 'core system' in its own



right – thus making it an obvious candidate for interfacing your hard disk recording system to the numerous 8-bus analogue mixers on the market.

Both the MOTU 1224 and 2408 serve as the front end to what is known as the MOTU Audio System, which is essentially made up of three components: the audio interface(s), the MOTU PCI-324 card, and a choice of digital audio software. The software that currently directly supports the MOTU Audio system includes: Cubase VST (Win/Mac) via ASIO, Logic Audio (Win/Mac), Opcode Vision DSP and Studio Vision, Cakewalk, Syntrillium Cool Edit Pro, and Sonic Foundry's Sound Forge. The MOTU Audio system also provides multi-channel I/O for any Windows audio software which supports standard multi-channel Windows Wave drivers.

Let's look at the unit's back panel first. The MOTU 1224 provides eight channels of balanced +4dB TRS jack inputs, eight channels of balanced TRS jack outputs, two balanced XLR outputs (for the main left and right outs) and stereo AES/EBU digital I/O. The 1224 features 24-bit converters and a quoted 116dB of dynamic range. The unit also features wordclock I/O which allows the 1224 to synchronise with other digital audio components, such as providing a master clock to drive a digital mixer.

The only garden-variety I/O missing on the 1224 is S/PDIF I/O. The 2408 does feature S/PDIF I/O but itself lacks the AES/EBU digital I/O the 1224 features. Apparently the idea here is a system incorporating both the 2408 and 1224 won't double up on features, but will collectively provide enough audio I/O to connect with anything. Metering on the 1224 includes dedicated six-segment LED meters for every input and output, including stereo AES/EBU. The 1224 also comes with a 24-bit stereo headphone output with front panel access and its own volume control. The high quality built-in headphone amplifier provides more than enough output to drive power hungry studio headphones or lengthy cable runs.

The second component in the MOTU Audio system is the PCI-324 card. The PCI-324 audio card acts as the 'transit lounge' for all data coming in and out of a 1224 or 2408 audio interface, and all data coming in and out of your digital audio computer software. Though relatively small in size, as audio PCI cards go, the PCI-324 can handle an impressive 72 channels of audio input and 72 channels of audio output via three AudioWire ports. The actual AudioWire data transfer technology is a proprietary derivative of FireWire — an adaptation of the IEEE 1994 to 1995 standard — and is a relatively new protocol capable of data transfer rates of 100, 200 and 400Mbits/s between computers and peripheral devices.

Three MOTU 1224s hooked up to a single PCI-324 card equates to a 24-track studio setup featuring 24 analogue inputs (plus six digital ins) and 24 outputs (plus six analogue XLR and six digital outs), and there's always the option of including a MOTU 2408 in the setup to provide comprehensive interfacing with ADATs and Tascam DA series DTRS machines. However, bear in mind that the ability to process all this is still dependent on CPU power, RAM and hard drive specs.

Mark of the Unicorn are well known for their pro quality Midi+audio software and hardware components, and their broad product base places them in a healthy position to deliver a comprehensive recording system incorporating audio, Midi and sychronisation integration. The 1224 is the latest component in this system and provides the critical high quality analogue digital audio and mixer software interface. With 24-bit A/D conversion all round the 1224 sounds clean and quiet. At the core of the system is the MOTU PCI-324 card which offers powerful processing capabilities in terms of the quantity of I/O possible, and, in conjunction with the MOTU 2408, provides a flexible upgrade path for the future, capable of handling large scale multitrack, transfer and synchronisation requirements well into the future.

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Price

• *$2695; $3495 (MOTU 1224 plus PCI-324)*

Mindprint Envoice

Konrad Skirlis sets his mind to reviewing a new voice channel.

Envoice is a voice channel that offers a high quality mic preamp (with line and instrument inputs), an equaliser section and a valve compressor. Furthermore, the unit allows for an optional digital I/O card. Mindprint is a recently formed German company which are planning to release more units down the track. A standalone valve compressor, a valve-based parametric equaliser and an A/D converter are some of the products coming our way!

The Envoice is a 1U rackmounting box attractively coloured in earth red on the front panel. It's easy to view and operate, with the front panel layout being representative of typical signal flow. Looking at from left to right, its three main sections are: mic/line input, a switchable three-band equaliser, a compressor, valve and output section. All the Envoice controls are recessed against shiny reflective metal, all up an attractive design. The input section has a recessed LED-style meter. The meter is switchable to show input or output levels and beneath this is an analogue/digital switch and a mic/line input switch. An input jack allows instruments to be DI'd with the rotary button to the right adjusting input gain. Alongside this is a switchable filter with 50Hz/100Hz roll-off options.

The next section of the Envoice is an equaliser divided into three bands, each having a bypass switch with a red LED display. The low section has a 20Hz to 300Hz bandwidth with ±15 dB gain. This bandwidth allows narrower cutting with a wider bandwidth for boosting! An interesting bell shape that can effectively reduce specific low frequency problems. The mid section of the EQ is parametric with a variable Q control tunable from 100Hz to 11kHz with 15dB of cut/boost available. The high section of the EQ has a range of 1.8kHz to 22kHz with a symmetrical bandwidth operating again at 15dB cut or boost.

The next section is the compressor with associated gain reduction metering featuring tube saturation, threshold (+2dBu to -28dBu) and compression (1:1 to ∞:1) controls. Tube saturation will affect how much signal is being driven through the 12 AX7 valve. An LED next to this setting indicates green for an underdriven valve, yellow for moderate valve signal and red for an overdriven valve. Additional switches select slow/fast attack and release times for the compressor with a switch providing filtering for reducing low frequency related compression pumping. The output level is adjustable and an Effects switch (with LED display) acts as a bypass for the whole unit.

The rear panel of the Envoice unit has a female XLR for balanced mic input with an associated 48V phantom power supply switch. Balanced line inputs and outputs offer both XLR and 6.5mm TRS jacks. The power connector is IEC with an earth lift switch.

I found the Envoice straightforward to use with the right features present. The mic input is of good quality with enough gain at hand. There was no problem powering up a number of condenser microphones with the on-board power supply. The equaliser is smooth and clear while the compressor seemed to handle a wide variety of spoken and musical styles. The attack/release times for fast and slow compression is 15ms/60 ms and 150ms/600ms respectively. With compression times being preset, you could argue this might be a limitation, however, it does facilitate quick and easy setup. The inclusion of a sidechain filter allowed the compressor to bypass lower frequencies below 300Hz thereby avoiding bass related pumping. The saturation control is at the gain stage before the output and therefore doesn't require compression to work. Overdriving the valve produced smooth and pleasing harmonic distortion.

The En-voice processor is a solid unit that offers considerable flexibility. The mic preamp is quiet and the inclusion of direct injection is a plus for direct to disk recording. Add the Di-mod interface and you keep the signal digital once it hits the Envoice. Furthermore, the parametric mid EQ section adds versatility to an otherwise smooth sounding equaliser section. While the compressor may lack some variable settings, it does work effectively and provides the added bonus of low frequency filtering – thereby preserving the bass frequencies of heavily compressed signals. The valve circuit warms things up and effectively displays saturation levels. For anyone after additional outboard gear or working with hard disk recorders, the Envoice deserves your attention.

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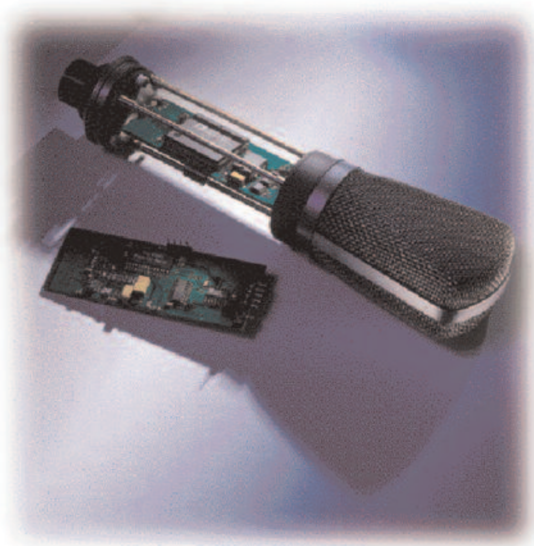
Price

• RRP $1295

Beyerdynamic MCD100

Michael Gissing of Digital City Studios, was given a 'digital' mic to wrap his digits around.

Beyerdynamic have made the world's first series of digital microphones – well, that's their claim and I can't find any evidence to the contrary. The MCD range includes large diaphragm omni and cardioid mics, shotgun style mics, and capsule mics with interchangeable omni, cardioid and super cardioid heads. The term digital microphone means the analogue signal undergoes A/D conversion at the microphone, resulting in an AES3 digital output. Digital City Studios, as its name suggests, has worked hard to maintain a totally digital path at all stages of the recording, editing and mix processes, and the prospect of a microphone that outputs a digital signal certainly caught my attention. For years I have relied on just one type of microphone to cover the full range of recording work that I do – from dialogue and effects recordings for film and television, to CDs of acoustic instruments recorded in real reverberant spaces. My trusty benchmark mics are the Schoeps CMC capsule series. But those Shoeps are getting a bit nervous in their mic cabinet, I may have found some serious competition for their spot.



Digital Watch

I was given two Beyer MCD100 large diaphragm cardioid mics for the purposes of this review. You don't just plug these mics into a digital input of a desk, they need to connect to a box that supplies phantom power to two mics. The MCD100 came with the MPD200 power supply box, which can power two mics. The MPD200 has two XLR connectors at the microphone end to connect individual mics and supply them with DPP, (Beyer speak for Digital Phantom Power). The mics are made to withstand the accidental connection to an analogue desk feeding +48V phantom. A wise bit of safety engineering as, frankly, most studios will make this mistake at some stage. The MPD200 has error LEDs to indicate whether the connection to the mic is correct and a digital signal is reaching the box. The MPD then has a single XLR to output a two-channel AES3 digital signal to the recorder or digital desk. The mic outputs a 24-bit/48k signal by default, but, according to the manual, the MPD200 can accept an external word clock signal which will reclock the mics to anything in the 32k to 48k range, (and yes they do promise 96k support in the future). For multi mic studio setups, Beyer also supply a rack mount DPP (the MPD800), which powers, gain controls and word clocks eight MCD mics.

The final feature on the MPD 200 is the ability to cut 20dB or gain 36dB in steps. According to the brochure, the cut is an analogue trim within the MCD100, but signal boosting is done by digital gaining. This surprised me, as I had assumed that gaining the input to the A/D converters in the microphone would be the approach. Beyer digitally gain with their on-board DSP chip, using 32-bit floating point processing, so they're using a proper digital gaining technique. To cover the range of recordings that we experience these mics did need gaining and I would be happier to see a +20dB switch on the mic which steps up the analogue gain to the A/D, in spite of the fact that this would doubtlessly complicate Beyer's anti digital clipping feature. I have been told that Beyer can deliver the mics with different gain settings, at the cost of headroom.

I do applaud Beyer's 'digital' honesty when they reveal in their specs that they achieve 22-bit A/D resolution. For the record, unscrewing the mic case revealed an Analog Devices ADSP-2115 chip converter. The microphones come with a heavy duty suspension mount, while Beyer accessories like foam wind covers and a bolt on pop screen are also available.

When it came to connecting the microphone to my O2R I encountered some clocking problems. The MPD200 comes preset from the factory in master mode, which means it ignores any external clock signal and will output an internally referenced 48k signal only. So out came the tool kit and the box was opened to set an internal jumper for external clocking. Once the jumper is set, the box needs an external clock. Frankly this is poor design. The choice to clock internally or externally is an on-the-spot decision for a roving recordist and I am used to boxes with switches or auto detecting. Unscrewing box lids in the field is just not on. Next problem was that the mic just wouldn't clock to 44.1k wordclock, which is my studio reference. Every other digital device I have tested locks to my house clock, which, I hasten to add, is low jitter and passes the stringent Prism analyser tests. To overcome this problem, I routed the mic through a Z-Sys sample rate converter and then into the O2R at 44.1k. Not the best way to connect a digital mic, but at least I was now ready to hear it!

Smooth & Accurate

Initially I was impressed by how quiet, smooth, and accurate sounding the mic was. The gain structure has been optimised for music recording, so for dialogue/effects type recording, I used +30dB gain. This was partly to hear whether such a boost compromised the digital signal, but also to have it match the gain settings for the Schoeps. When the levels were matched I recorded a variety of voice and effects signals, with varying proximity to the mic. I also tried to overload the MCD100. Beyer says the mics employ a digital clip protection circuit, which is not compression or limiting and has no 'discernable effect' on the signal. I found peaks were held at maximum level (digital zero) very effectively without introducing distortion or signal pumping. I had to resort to blowing in the mic to overload it, which did give me a nasty limiting effect and did go digital 'over'. But for most practical applications this clip protection will give a margin of safety for a recording. This is important because, once you are recording, you do not want to change the levels on the MPD200. As I stepped through the gain settings on the box, a noticeable 'zipper' effect was heard, apart from the fact that the gain steps in +6dB or -10dB steps.

As I played back the recordings off a dSP hard disk recorder, I was struck by the fact that the Beyer was quieter than the Schoeps. This is impressive, as the Schoeps has a low self noise. Beyer claim in their specs that the MCD has a dynamic range of 105dB (CCIR). This realistically would equate to a usable 100dB signal to noise range, an excellent spec for a microphone. The second surprise was that the Beyers had a flat, smooth sound and certainly more bottom end than the Schoeps. I liked the sound. I wanted to test the on-board converter, and a good test is to record the rattle of a bunch of keys. A bit of 'frying' was evident with the Schoeps going into the standard O2R convertors. Meanwhile, the MCD100 stayed clean, smooth and accurate. So far the Beyer was a clear winner. Moving back off the mic, it became very hard to pick the difference. Without the proximity, they both showed very similar tonal qualities, with the MCD100 smoother in the mid to high frequencies. Both mics conveyed a clear sense of the room acoustic.

Off axis response was a surprise. The volume was down, as you would expect. High frequency response was also smooth off axis, but the bottom end fell away off axis. To be fair, the off axis test was done with close proximity micing, so the fall off of bottom end was perhaps due to the fact that on axis, up close, the MCD100 does give a nice bottom end lift. Also the rear of the MCD100 certainly had a high frequency attenuation characteristic. Again this was not surprising for a large diaphragm mic in the typical 'shaver' shape.

As a mic for singing or dialogue recording, the MCD100 is very smooth and accurate. It doesn't have that typical large diaphragm characteristic of warm bottom end/brittle high end. It is warmish in the bottom end when used up close, but the stand out feature for me is the smoothness in the mid and high frequencies. Not everyone likes accurate flat sound, but I certainly do, and the Beyer impressed. I would certainly sling a pair over an orchestra, particularly for ambience mics. I wouldn't hesitate to put them up close for percussion, woodwinds and bass either. With 130dB SPL equating to digital zero and the ability to handle 150dB SPL with the -20dB trim, this mic can take high level hand to hand fighting on the chin.

Beyer also has a trick up its sleeve with these mics – it has an EPROM which controls the on-board DSP. Beyer offers a service of providing preset filter patterns recorded on custom EPROMs. So if you want to spec a mic for a particular purpose with a particular filter characteristic, then Beyer can provide that. By starting with such low noise, flat, accurate sound and high quality A/D conversion at the microphone, Beyer has got a winner with the MCD series. A recommended price of $4,500 for one mic with suspension mount and the MPD powering box may seem a bit pricey, but remember that it saves the need for an expensive A/D converter – and finding an ultra low noise, great sounding microphone is like finding the holy grail.

Distributed by

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Syncrotech on WWW: 'www.ssd.com.au'

Price

• Approx. $4500 for MCD100 & MPD200

Quested VS2108 self-powered Monitors

Can an active near-field monitor possibly sound better than Simmo's ATCs? Brad Watts thinks so...

Roger Quested has been designing and building monitoring systems for some of the world's greatest studios for many years now. Abbey Road, Sarm and Wisselord are all proponents of his larger speaker installations. But Quested's philosophies in sonic accuracy are not limited to loudspeaker design – he's also very involved with his studio design business, QSound, which has installed systems for corporate post, broadcast and surround installations around the globe. This expertise has been imparted to Quested's relatively recent designs in near- and mid-field monitoring.

Quested, the company, now manufacture a range of active monitors specifically for smaller studios and surround sound work. The subjects of this review are the VS2108s, with eight inch drivers and ferrofluid cooled soft dome tweeters. The review pair were rather tastefully augmented with a single Quested VS1112 sub-bass cabinet – a very large cabinet that consequently pumps out a very large amount of lovely bottom end. For the purposes of this review we'll focus on the VS2108 monitors alone.

The nuts and bolts.

The VS2108s are a two-way self-powered monitor. [Quested use the term 'self-powered' to describe active monitors with built in amplifiers, and reserve the term 'active' for their larger soffit-mounting monitors which are supplied with external active crossovers and amplifiers. – GS] They aren't light – at 22kg each they'd appreciate a good set of road cases if they were to be moved often. Part of the weight could be attributed to their magnetic shielding, which allows them to be placed near computer or video monitors without distorting the on-screen image. The front baffle and walls of the cabinet are made from 22mm MDF, which helps minimise any panel or cabinet resonance. They're a considerable size as well: 400mm high, 338mm wide and 340mm deep. At first they almost look like a cube.

You get some idea of the Quested ethos when reading the manual. After a cheery "thank you for buying our monitors" and a description of the units, the manual immediately goes on to warn of the dangers of hearing damage. The manual states that the VS2108s can deliver sound pressure levels in excess of 108dB, and provides a chart of acceptable listening durations for particular dB levels. This reflects Roger Quested's experience and holistic approach to monitor design.

From the sides of the cabinets you can see the blue anodised aluminium heatsinks for the power amps. These get quite warm and require reasonable ventilation – Quested recommend a clearance of 100mm around the heatsinks when used in ambient temperatures up to 30°C. To the back of the unit are connections for an IEC power lead, and signal input via XLR. Above this are a pair of three position 'frequency contour' switches which allow compensation for room conditions and mounting positions. The first of these determines the high frequency contour, allowing a flat response or a choice of 1dB boost or cut from 5kHz to 20kHz. The second three position switch determines the low frequency contour and is usually set to the 'extended' setting for nominally flat response. The next position ('off') reduces low frequency extension but increases the headroom available between 50Hz and 200Hz by an average of 1dB, good for bass heavy material or for monitoring in rooms with difficult LF behaviour. The third position sets the VS2108s to be used with external sub bass cabinets.

Below the frequency contour switches is a recessed rotary switch for adjusting the input sensitivity, from -14dBu to 0dBu in 2dB steps. The input is electronically balanced and has a nominal impedance of 10k ohms. A rocker switch for power on/off is present, with intelligent switching to ensure safe noiseless power-ups. Everything is clearly labelled in white screen print.

Behind the rear backplate is housed the two amplifiers which drive the show. Although the manual doesn't go into great detail about the amps it does offer some essential specifications. The LF driver's amp has a rated output power exceeding 110 watts RMS, while the HF driver's amp exceeds 100 watts RMS. Total harmonic distortion is less than 0.03% at levels up to 1dB before clipping point, and typically 0.005% at normal operating levels.

On the front of the cabinet a dual colour LED is built into the Quested emblem, just below the bass driver. The LED lights green during normal operation but will turn red 0.5dB before clipping. If the LED turns red and stays that way then it's likely you wont be hearing anything at all, as this mode notifies you that the amplifiers' thermal protection has cut in. The VS2108s will remain silent until the amps drop to a suitable operating temperature.

The Business End

On the front of the cabinets you can see the two drivers and the two port openings. Between them they deliver a frequency range from 60Hz to 19kHz, ±2dB. The low frequencies are delivered by a custom designed 200mm high excursion bass driver with dual rear suspension which, according to the manual, "offers exceptional transient response and low distortion even at high listening levels". I've experienced a few bouts of 'high listening levels' with these cabinets and I can certainly vouch for their low distortion characteristics.

The high frequency driver is a Morel ferrofluid damped 28mm soft dome tweeter. Its wide dispersion ability negates having to involve any 'waveguide' or horn loading techniques. Indeed, the frequency response curves show very good off-axis response.

Either side of the tweeter are two ports. Ports at the front equal a good thing as far as I'm concerned. It means you can build the cabinets into a wall (soffit mounting) and it's better having the result of the port coming straight at you rather than being bounced around the room. Plus I quite enjoy getting a blast of low frequency air current in the face. Groovy!

The sound

As for how they sound? I can't recommend them highly enough. I've had a swathe of monitors through my studio in the last couple of years and these are the first ones that have really got me excited. I've even ditched our illustrious editor Greg Simmons' beloved ATC SCM20s in favour of the Questeds (well, I should say he's taken them off me now that I've bought the Questeds – all privileges revoked!). I luv 'em. They're designed for reference quality monitoring when recording, mixing and mastering. Right up my ally, in other words, and so I've acquired a pair for myself.

I've played all my favourites through the VS2108s, and I'm still astounded by the incredible bass perceived from them. Drums and bass have the impression of being right there. They really kick! I wouldn't advise putting them in a small (read: 'bedroom') studio, as you'll probably have the neighbours beating on your door. They're more likely to find themselves in mid-sized rooms or as the nearfield monitoring in larger control rooms.

The softdome tweeters are lush – I can work with these monitors for extended periods without loosing my judgement for what the top-end is doing. The overall sound is smooth and balanced, with a hearty degree of oompf thrown in. The stereo imaging is very solid and, thanks to their wide dispersion, not narrowed to a tiny 'sweet-spot'. (Of course, things get even more addictive when you match them up with the VS1112 sub-bass cabinet, but that's another review.)

In short, the VS2108s are extremely accurate high power reference monitors, and a pleasure to own and to use. They won't suit everyone's credit limit, but they certainly rise above the current offerings on the market.

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• *ATT Audio Controls*
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Price

• *RRP: $9590/pair (available individually for surround work at $4795/box)*

And furthermore…

During the course of reviewing the Quested VS2108s and the many interesting conversations which ensued, Brad suggested I write a companion piece to his review. So here it is…

Unlike other top-of-the-line monitors we've reviewed in the past, Quested's VS2108s don't boast any exciting new design concept or groundbreaking technological discovery – just good solid engineering and design principles, and the results speak for themselves. Once we'd sorted out the correct mounting (see my 'First Word' column in this issue) and tweaked the input sensitivity and contour switches to our liking, these monitors really came into their own.

Interestingly, though, you have to sit further back from the VS2108s before the sound takes shape. Sitting in the traditional 'sweet spot' in Brad's room and listening to the Questeds just didn't work at all – the sound was confused and unimpressive. However, moving back about 45cm made all the difference in the world – everything snapped into focus. Keep this important point in mind when auditioning the VS2108s, or you won't be judging them properly.

Personally, I'd rank them right up there with ATC's SCM20A Pros (reviewed in Vol. 1, Iss. 1) and JBL's LSR32s (reviewed in Vol. 1, Iss. 4). They're more forgiving than the 'take no prisoners' ATCs, which I attribute to their larger diameter HF driver (28mm vs. 25mm), while their ported design and larger enclosure gives them greater low frequency extension and 'slam'. Overall, the Questeds have a warmer and fuller sound than the ATCs, though slightly less critical. Being an active two-way, they don't have the dynamic agility and 'speed' of JBL's three-way LSR32s (few monitors do!) but they would be right at home in rooms that may be a little small for the JBLs. It's not surprising to learn that Frank Hinton, Quested's Australian distributor, is getting many requests for Quested monitors from dance music producers – their portability, accuracy and low frequency performance make them ideal for this application, although I must point out that their capabilities extend way beyond that market alone.

I've said it before and I'll say it again: choosing monitors is like choosing prescription lenses without the help of an optometrist – you've got to find the pair that bring your hearing into focus. Between the ATCs, the JBLs and the Questeds, we've looked at three similarly accurate but tonally different monitors over the last two years. I could not recommend any one pair over the others without considering your personal requirements and tastes.

- Greg Simmons

Roland DS90

Roland 24-bit monitors? Derek Johnson & Debbie Poyser go digitally in.

Digital studio monitors are a rare breed, with only two examples currently on the market (Genelec's 2029A nearfields being the other). The general advantages of digital monitoring are still not crystal clear in a studio market where few mixers offer digital monitor outputs, but there's a use for them in desktop music. Computers are increasingly used for multitrack recording, via digital audio cards, and a card's S/PDIF digital out could be connected directly to digital monitoring, bypassing the card's possibly noisy analogue outs.

Roland has further justification for launching digital monitors: as a partner for some of their popular 'VS' hard disk multitrackers and new VM-series digital mixers. Many of these units do offer digital monitor outputs, plus an innovative COSM (Composite Object Sound Modelling) feature intended to reproduce the listening characteristics of a range of speaker types.

The DS90s' sizeable (370 x 229 x 330mm) enclosures aren't adventurous in appearance, but they are magnetically shielded, for use close to VDUs, and have a rounded corner profile to reduce cabinet-edge diffraction, thus encouraging a smoother response. Their specially designed 6.5-inch polypropylene bass drivers are complemented by a one-inch soft-dome tweeter and twin bass ports, combining to produce a respectable 48Hz to 20kHz frequency response.

The bi-amped design provides discrete bass (60W) and treble (30W) amplification, with an active crossover set at 2.6kHz. I couldn't find an SPL figure in Roland's printed material, but the speakers seem capable of very high sound pressure levels. They can be tailored somewhat to fit a room acoustic with HF and LF trim controls, offering up to 3dB of cut or boost, on the back of each cabinet, where you can also find a power switch and input level control. The latter two would be much more convenient on the front panel!

While the DS90s are billed as digital monitors, they can function as normal analogue monitors, fed from an analogue mixer. A switch determines whether the digital or analogue input is amplified. If analogue is chosen, a combi-jack socket accommodates balanced jacks, unbalanced jacks, or XLR connectors.

Digital interfacing is provided by coaxial and optical connectors in the 'consumer' S/PDIF format, along with a switch selecting between them. The speakers are connected to each other digitally (unlike the Genelecs) via a coaxial 'thru' connector which routes the other half of a stereo signal to the matching speaker in a pair. Digital signals of up to 24 bits, with a sample rate of between 25kHz and 55kHz (allowing for varispeeding on digital sources), are handled automatically.

As befits their size, the DS90s produced a big, deep sound picture. With acoustic and classical material they performed perfectly capably, but they tested particularly well with rock music. They also coped admirably with hard-hitting dance, showing an affinity for edgy, synth-based music. HF detail was good, but the DS90s seemed a little too pumped-up in the bass area, with a degree of uncontrolled boominess on certain material. This can be mitigated somewhat with the LF trim control. Stereo imaging was fine, and the monitors could be driven painfully hard – though this isn't advisable, naturally!

Having a Roland VM3100 Pro mixing station knocking about, I was able to check out its COSM speaker modelling with the DS90s. And, indeed, radically differing speaker characteristics are simulated by the different models – potentially useful during mixdown. For example, I was able to confirm the radio-friendliness of a kick drum I was using on one dance track, but was unsure about, by simply switching to the 'Radio' simulation. Swapping instantly between COSM-modelled speaker responses can helpfully widen your picture of the material you're mixing, by highlighting different facets of a track. I still used my normal methods of mix-checking too, but during the course of this review I went from being a tad dubious about the advantages of the speaker-modelling system to feeling it deserves a cautious thumbs-up.

I can see how Roland believe these monitors to be a logical linkin their chain of digital gear, especially given that they're designed to work with COSM. But, I suspect their price tag may be a little high for the average VS workstation owner. But, to be fair, they do save the cost of an amp, and, with the modelling, could save on the cost of another monitoring system.

Finally, these are robust speakers with well conceived, straightforward digital connectivity, many pleasing sonic characteristics, and more than enough power for any normal monitoring situation. Their slightly exaggerated bass response gives some cause for concern, but they're certainly well worth a listen, especially if you own a Roland VS recorder, VM mixer, or feel you need digital monitors to go with your computer-based recording system.

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- *Roland Corporation*

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Price

- *RRP: $1495 each*

Electrix processors

Christopher Holder get 'hands on' with the Electrix range of effects.

It's hard to make an accurate assessment of the essence of '90s pop music when we're yet to wave the decade goodbye through a cloud of pyrotechnics or via a Clive James new year's special. But I'll hazard a guess. I'll go out on a limb and say the pop music of the '90s will be characterised by the creative use of signal processing. The advent of computer-based recording (with the swarming cloud of plug-ins that followed) has meant that once your work was committed to disk it was fair game – flanging, phasing, distortion, filtering, beat-munching, de-hiss, re-hiss, anything you're heart desires, and many things it doesn't.

Soon after, the fact that you could have your drums sounding like they were being played backwards in the engine room of the trans-Tasman Jet Cat just wasn't enough. To stay current you needed to be able change it all in real-time, so the sound morphs into the drummer swapping his sticks for a Bamix and playing double time on a dog sled. Now we're cooking! But real-time tweaking is best achieved with machines that have real knobs. That's why the likes of the Shermann filter bank has proven to be so popular – with loads of knobs, it's basically a full blown synth without the oscillator.

Electrix has taken note of this desire for real time processor tweaking. There's three units available thus far. The filter factory (a stereo filter bank with distortion), the Warp Factory (a vocoder) and the Mo-FX (a multieffects bank with distortion, flange, tremolo, and delay). All are 2U rackmounts, but have a 'wedge' shape so they are equally happy as a table top unit. When I first saw this range in a brochure I concluded they looked a bit tacky and flimsy. On the contrary, in the flesh these units are very sturdy in construction. The pots and buttons feel good to touch and are very positive in their action. The aesthetics are a matter of taste, but personally I like the bright front panels. Each unit shares certain features. They all have Engage and Momentary buttons, which allow you to punch in/out the effect. The Mo-FX and the Filter Factory can both use Midi Clock to synchronise their effects and LFOs, which is exceedingly handy. On the rear panel, all the units sport RCA phono I/O (as well as balanced TRS jack I/O) to keep the DJs happy. All units use an internal PSU.

The Warp Factory vocoder isn't like the vocoders of yore that use multiband analogue filters. The Warp Factory uses a physical modelling DSP algorithm. For those who don't know the deal, vocoders works by imposing the formant shape (the vocal distinctive vocal sound) onto another (carrier) signal. But if you didn't know how a vocoder works you most certainly know how it sounds. Yes, you can sound like a robot – but there's more. Once you have your source signal setup, then route your formant source (normally the vocal sound) to the Warp Factory, or plug a dynamic mic directly into the unit. In operation the Warp Factory has a good degree of editability a wide range of sounds. The three main controls Gender, Q, and Order, do most of the work, while Robot Pitch and Noise Mix, do what you'd expect, adding a synth tone and white noise for further variation. My favourite control was probably the formant freeze button. It captures the vowel sound present at the time and constantly applies it to the source. I found that it added a whole new effect to drums and bass, or anything else for that matter. The Warp Factory is a useful processor. Once you've got over the 'I'm a robot' stage (which takes some time I can tell you), you can get it working on a variety of material.

The filter factory is a stereo analogue resonant filter bank with a analogue overdrive section. The overdrive can add a little grittiness or out and out demolition of your source signal before being routed to the filter. The filter is two-pole 12dB/octave with resonance. If you're after something more extreme, then hit the mono four-pole button. There's nothing shy about the resonance, it boldly races into self oscillation. There's a button to toggle between high pass, low pass, band pass and notch filtering. An LFO section offers a good selection of waveforms speeds, and a depth range. With Midi clock sync you'll have the LFO doing something very useful. This filter has no delusions of musicality, it's meant to sculpt your sound with a vengeance. The heavy weight filtering effects we so often hear in most forms of music these days is easily achievable. Lots of fun.

The Mo-FX is the newer of the three processors to these shores. Think of the Mo-FX as a multieffects unit with knobs on. From left to right you've got distortion, flange, tremolo, and delay. The distortion can be used on the input signal and routed directly to the mix bus or used and routed to the rest of the effects chain – which is good thinking as the distortion can be as extreme as you want it and if it's automatically routed to the other FX blocks you can really end up with a complete cacophony. The flanger is next, with mix, depth, speed and regeneration controls. A frequency Band button toggles between seven default combinations of frequency bands that the flange can effect. There's a good amount of movement and interest in the flange. Drums and percussion enjoyed the Electrix flange experience, as did strings and anything else to be honest, including bass sounds. Next up is the tremolo, which, as we know, is all about amplitude modulation. There's Mix, Speed, and Waveform controls, as well as Band and Auto-pan buttons. With the tremolo you can achieve anything from mild and smooth modulation of the volume through to extreme stutter gating effects. The delay section includes mix, regeneration and speed controls, (the speed ranges from 1ms to 2600ms). A ping pong button gets the delay panning left and right.

Electrix are really onto something here. From the moment you see the boxes with their mock 'road case' cardboard, to when you first see and use the buttons, and look at the features, you know they've done their homework. Electrix has obviously conferred with their potential clientele extensively and come up with processors that could hardly be more hands on. And, as much as I hate to admit it, we have the DJ fraternity to largely thank for this. DJs have had frequency 'kill' switches and momentary punch-in buttons for years, it's only now that we have access to such a tactile effects range in the studio. Much of the key to the enjoyment lies in the time sychronisation of the effects. We've all probably rigged our multieffects units into our Midi chain believing we'll use the on-board fiddly Midi-controllable parameters, but rarely do. Meanwhile, the Electrix units (the filter factory and the Mo-FX especially) fairly cry out for a Midi lead. The Mo-FX can have the flange speed set to one cycle a bar, the tremolo can be chopping things up every semiquaver,

while the delay can be 'ping ponging' on every eighth note. Thankyou Midi clock and Electrix! As for Midi control, to have every (and I mean every) button and pot sending and receiving Midi controller info is brilliant – forget about a steaming spa bath and a cake of Dove, this is luxury. Set the sequencer rolling and tweak away. The manuals are handy, with useful information and sample setups. Minor niggles include the omission of input and output level controls and the basic metering. I suppose presets would also be handy, but almost defeat the purpose. What you see is what you get – and what you get is something you won't want to take your hands off.

Distributed by

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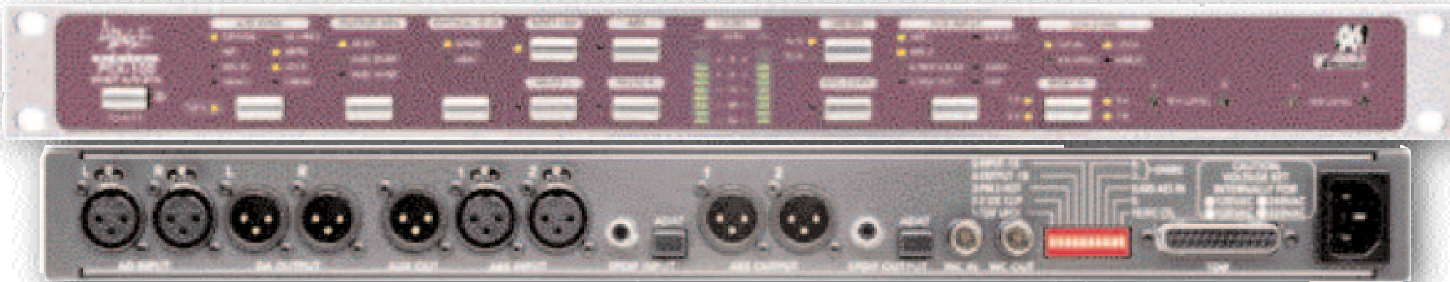
Electrix on WWW: 'www.electrixpro.com'

Price

• RRP: $1095 for each unit

APOGEE PSX-100

Apogee Electronics are addressing the 24-bit/96k market with this combined A/D and D/A converter. William Bowden samples the outcome.



Apogee Electronics have specialised in building high quality A/D and D/A converters for some time now, but the PSX-100 is their first foray into the emerging 24-bit/96k domain. With an asking price of $5795 the Australian market may be tough to crack, but for the money you get plenty of features and some nice extras.

The PSX-100 is an unassuming looking 1U rackmount box that sports quite an array of features: an A/D converter (24-bit with 44.1k, 48k, 88.2k and 96k sample rates), a D/A converter (24-bit with a 32k to 106k range), stereo PPM-style metering, an analogue limiter, extensive format conversion, and 'Bit Splitting' (which allows 24-bit material to be recorded on 16-bit machines), all controlled via 11 front panel function buttons. It's powered by a 25W internal supply connected via an IEC lead. The interface looks good and offers a variety of input and output formats: S/PDIF (optical or coaxial), TDIF, AES/EBU and ADAT. You can also sync the unit via BNC wordclock or export wordclock and take advantage of Apogee's low jitter (no more than a quoted 22 picoseconds RMS) internal clock. A rear panel DIP switch (10 by two-way) offers further functions and tweaks.

It's a well built unit in terms of layout and useability, and almost any configuration is only two or three button presses away. Each button has a status LED beside it, so at a glance you usually know what's what. The unit runs warm due to the power supply's toroidal transformer. This is not unusual and the left hand side of the casing is dominated by a large anodised purple heatsink which warms up in operation but is certainly not as hot as previous Apogee designs. Looking inside the box reveals a spartan layout, comprising two boards. Next to the power supply is a large digital board (with a promising looking ROM OS chip) and on the far right is a separate analogue circuit board. The casing is very strong and heavy and, no doubt, would easily survive a fall or the rigours of life outside the studio.

A unit of this calibre will normally offer plenty of features and customisable parameters, and the PSX-100 is no different. The first notable feature is Apogee's well known UV22 dither process. You'll be needing this because 24-bit audio involves more information than a CD burner or most DAT recorders can handle, and UV22 will reprocesses the output resolution of the A/D from 24-bit to 20- or 16-bit wordlengths, so you can master to 16-bit formats. It's an effective and good sounding system, but it only works on the lower sampling rates. The manual states that, "sample rate conversion is a tricky business at best and you need the maximum resolution to do it". I must confess I was a little disappointed to find that I couldn't record at 96k/24-bit and output it at 44.1k/16-bit all in the one box (for example, dbx's Quantum mastering processor will do this). I can only guess Apogee assume this level of accuracy requires specialised resampling, and one box really can't do everything as precisely as a dedicated unit.

On the positive side, you can record a 96k/24-bit stereo program onto a 44.1k/16-bit multitrack recorder by using the PSX-100's Bit Splitting option. This only works via the TDIF or ADAT multichannel interfaces as it requires four tracks on your digital recorder. Because there are only two AES/EBU outputs, you can only record and bit split at lower sample rates through those interfaces (bad luck for owners of the Sony PCM800, which doesn't offer the TDIF or ADAT interfaces). Nevertheless, Apogee's Bit Splitting makes 24-bit recording a viable option.

Soft Limiting is another feature that has made Apogee famous. On board (but bypassable) is a dedicated analogue limiter circuit with a soft-knee style approach to attenuating transients. It is factory set to begin its action at -4dB FS and gets progressively busier as you pump more level into the converter. If you want to vary the threshold you can, but not via the front panel. You have to journey inside the unit and adjust variable resistors, and to avoid a L/R mismatch you'll need some test gear. Apogee did make a user adjustable front panel mod for their AD-500 converter, so here's hoping they do the same for the PSX-100.

In practice this limiter works rather well – at times much better than its digital counterparts in my mastering studio. But like anything, it's highly 'program dependant' and too much limiting is usually way too much. I'd like to have seen a metering mode that displayed the amount of limiting occurring, or even just a simple LED to show limiter action.

The front panel meters are small but very fast. They only go down to -50dB, but half the steps are devoted to the last 10dB before 0dB so they are most informative about the transient nature of the upper range of your signals. The 'Digital Over' indicator is resettable with a push of the Meter button, and you can choose between one to four consecutive full-scale samples as constituting an 'over' (via the rear panel DIP switches).

Piano Role

I had this converter for several weeks and was able to do quite a number of different tests. I guess the most obvious (and one of the most revealing) was to mic up a piano and listen to single notes being played at various volumes and lengths. This has long been one of my favourite tests because a piano provides such complex waveforms, replete with harmonics and overtones. I performed this test in Festival Records' Studio A, with two house engineers and one technician present. Going 24-bit/96k all the way, the PSX-100 was set up against an Apogee AD-500 and DA-1000 (16-bit/44.1k) combination and, of course, the piano itself. We miked the piano with a Neumann U69 FET stereo microphone – it's not the quietest microphone we have, but it is certainly one of the most popular and provided a good reality check for the two engineers present.

After the laborious process of lining up the converters with test tones, we were ready. The PSX-100 had quite a 'smooth' sound to it and revealed the evolving harmonics in the midrange strings quite well. In this area the older AD-500 was a little more 'honky' sounding in comparison. The PSX-100 also had excellent characteristics in the bottom end of the piano. In regards to the top end though, we began to notice that notes were slightly less present, and, while this sounded quite appealing at times, it definitely sounded less open and 'airy' than the AD-500/DA-1000 combination or the original.

Listening to the tails of notes, everyone noticed that the PSX-100 was not reproducing the extreme top end (mainly hiss from the mic) as well as expected. When we just listened to studio ambience (mainly the air conditioner) the difference was just as apparent in the tops but some of the low rumble seemed a bit attenuated as well. The notes did seem to sound well into the noise floor, but the sound of the reproduced noise attracted the most comment. While listening to the background noise, one of the engineers, Matt Lovell, immediately volunteered for a blind test which he passed 100%. He correctly identified all three sources, though he found the difference revealed by the PSX-100's sound was the easiest to spot. Hmm… I retreated to the controlled environment of my mastering studio and began to test the PSX-100 further. Recording solo instruments is one thing but mastering complex program material is another. I wondered how the PSX-100 would perform in this environment.

Tonally, what I heard in the studio was repeated in the mastering room but with some interesting results. For example, I was required to deal with a particularly nasty sounding track that I had been excitedly told was "all digital" in its recording and mixing, and of course it was on DAT – a budget job with no budget. In this case the PSX-100 provided me with an ideal finishing touch after tape, EQ and some 'retro' style processing had got me 95% there. Additionally, this was done by the PSX-100 at 44.1k using the UV22 process.

On the subject of UV22, I recorded the input noise of an AD-500 and the PSX-100, both calibrated to the same reference level. Despite the fact that the PSX-100 exhibited noise that was roughly 4dB louder than the AD-500 (they were in the region of -76.3dB and -80.8dB respectively – see waveform picture) the sound the UV22 produced was far more pleasing to the ear than the AD-500. It sounded more like tape hiss, had almost no rumble, and (when the gains were matched) UV22 came out sounding quieter than its forebear. Very interesting. I also recorded an identical program into both units at the lowish level of -70dB (peak) to simulate low level resolution on something other than noise, which is generally more or less 'stable state'. I'd suggest that the PSX-100 performed best of all on this test, subjectively sounding more coherent and less 'zippered' than the AD-500. While few CDs possess a dynamic range of this magnitude and remain listenable (apart from their fade outs, of course), most live (studio or location) recordings have plenty of information at these lower realms and when you finally come to mix or master them, the increased resolution will come into its own. The imaging of this converter is solid and generally the subtle lack of bite sounds very pleasant on a wide range of programs. I must admit that I found it was not as dynamic as it could have been, especially in the centre image. Dance music sometimes benefited from the capable bass and sub-bass response of PSX-100 but most 'four on the floor' kick drums were not quite as punchy as they were on some other (admittedly more expensive) converters.

Conclusion

Throughout this review I have been comparing the PSX-100 to Apogee's previous AD-500 A/D converter and DA-1000 D/A converter, and I must point out that, in its day, the AD-500 alone cost around $9000 (with power supply). The DA-1000 wasn't cheap, either. In contrast, the PSX-100 retails at $5795 and offers both A/D and D/A conversion, along with 24-bit/96k resolution.

There simply isn't room here to go into all the possible applications and strengths/weaknesses of the PSX-100. All converters colour the sound in some way, as do so many other highly sought after bits of kit, from consoles to microphones to cables. The PSX-100 has its own flavour but you may find that, if most of your work is entirely digital, it will help smooth off some of those rough edges. (If you're want total purity then start saving: you might just find an A/D converter that will get you close for about $10,000 or so, and just as much again for a similar quality D/A converter.)

My tests were largely based on a single pass with stereo program material. Of course, the 24-bit conversion would really come into its own when you are adding multiple signals together – the better the resolution the more accurate those mathematical approximations of mixing, bouncing and processing become. With the new breed of workstations capable of 24-bit/96k recording appearing everywhere (and the possibility of DVD high resolution recording), expect to see a host of 24/96 converters appearing throughout the next couple of years.

Apogee has clearly aimed the PSX-100 at pro and semi-pro users, and, by including ADAT and TDIF interfaces (and the bit splitting feature), has opened up 24-bit/96k recording to the masses. A very wise move considering the high market penetration of these digital interfacing protocols in the new breed of digital consoles, existing recorders and processors.

Choosing a converter is a bit like choosing a brand of beer. It's a matter of taste and your specific requirements are entirely up to you. Personally, I enjoyed working with the PSX-100 because it added another colour to my mastering palette, but I'm on borrowed time – this review unit has someone else's name on it.

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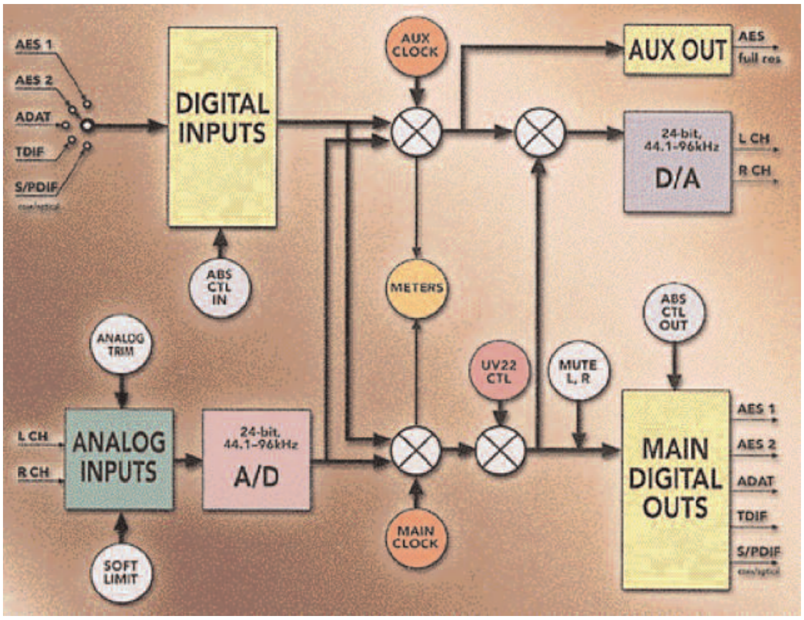
Price

- *RRP: $5795*



The PSX-100 consists of four main sections: digital inputs, digital outputs, A/D converter, plus additional functions such as aux outputs, UV22HR and Soft Limit. The way in which the main blocks are interconnected is determined by the current mode of operation. In Confidence Monitor mode, the A/D and D/A are essentially separate, the D/A deriving its input from the digital inputs and the A/D driving all the main outputs. Digital Copy takes the selected digital input and delivers it to all the unit's outputs, analogue and digital. This is intended for digital copying and format conversion without repatching. Analogue Monitor mode takes the A/D output and feeds it to all system outputs and to the D/A, which therefore monitors the conversion carried out in tht A/D section.

Apogee's reply

William mentions some of the limitations of the Soft Limit function. Here at Apogee we tend to view Soft Limit as an extension of the available dynamic range, not as a full-featured limiter. Thus, we've resisted the temptation to add the various features which are requested from time to time. The Soft Limit threshold is adjustable only from the interior. In my experience, external calibration has proved to be of limited use. If the threshold is raised above -4dB FS the gain reduction curve doesn't offer much protection against overs, while lowering the threshold may engender artefacts.

On the point of sample rate conversion: we have considered manufacturing a sample rate converter on a number of occasions. However, we have never been happy with the quality offered by available solutions for handling non-integral values (e.g. 44.1k to 48k, or 96k to 44.1k). This is almost certainly due to the fact that a very complex digital filter is required to handle non-integral sample rate conversion – and it's hard to get them to sound good.

Do bear in mind, however, that while in 'fast' mode, the Aux AES output on the PSX-100 gives you every other sample – so at 96k it gives you 48k, and at 88.2k it gives you 44.1k. Many people report that this sounds excellent, even though, strictly speaking, the filter coefficients are not correct for this application.

To address the issue of the 'sound' of the PSX-100: we do not deliberately 'flavour' our converters. However, it is a fact that the higher the quality of the conversion, the more like analogue it will sound. If you add in our proprietary technologies such as Soft Limit and then use UV22 on the result, you will end up with a signal that sounds very much like the analogue original. In the final analysis, however, the sound of a converter is very much in the ear of the beholder. Technology has changed a great deal since the days of our original AD-500, and we believe that our current converters provide far higher quality digital signals than our designs of a decade ago.

It's also worth noting that in the very near future we will release a firmware update adding single-wire 96k capability to the PSX-100. Thus you'll be able to interface the PSX-100 with virtually any 96k product on the market.

Roger Robindore
Apogee Electronics

Steinberg Cubase VST v4.1

Derek Johnson and Debbie Poyser look at the latest VST update.

Cubase VST is one of the world's most widely used Midi+Audio sequencers, benefiting from active and imaginative development by its manufacturer, Steinberg. The latest Mac version, v4.1 (and v3.7 on the PC), has some fabulous new features which maintain its position as a market-leading program and further integrates cutting-edge software tools from other manufacturers.

More than 20 audio and Midi enhancements have been made to VST for v4.1, and while they're all very welcome, there are a number that deserve special note.

Right up there at the top of the list has to be v2.0 of the VST plug-in standard. This allows for Midi control of plug-in effects, and introduces an even closer relationship with third-party software-based synths, samplers and drum machines than Steinberg's existing ReWire inter-application technology. These so-called 'VST Instruments' can now be chosen as a destination for Midi tracks in the Arrange window, just as if they were external instruments attached to a Midi interface. VST Instruments have their own mixer tracks, and can access the same processing plug-ins and global effects as normal audio tracks.

Steinberg are planning instruments of their own, and are whetting our appetites with Neon, a graphically stunning, simple but nice-sounding polyphonic 'analogue' synth. Third-party software which already integrates with VST 2 includes Koblo's Vibra monosynth, Stella sample player, and Gamma drum machine.

Also reaching v2.0 in this version of VST is Steinberg's audio in/out protocol, ASIO. The major development here is the implementation of 'direct monitoring' with soundcards which support ASIO. Monitoring is now handled by the audio hardware, so the monitored signal doesn't pass through Cubase VST. The result is the eradication of latency-induced delays in monitoring while overdubbing, for example. An elegant solution to an irritating problem.

When Cubase went VST it came as a surprise that the free bundle of effects and processors didn't include dynamics processors. Such studio essentials are of course available on the healthy internet shareware plug-in scene and from commercial developers, but now, finally, Steinberg has built some into the program. A gate, compressor, limiter and 'soft clipper' are available for each audio channel, in addition to the channel's four standard inserts. The modules are sourced from Spectral Dynamics, their implementations are quite comprehensive (although there's no side-chain access), and all four processors work well.

A few welcome operational changes have been made too. The new Mixer Views feature makes the program's Channel and Group Mixers more flexible by allowing users to set up their own channel configurations, showing only desired channel strips – helping to keep edit windows as uncluttered as possible, especially when working with large numbers of audio channels. Editing the values of certain parameters has been improved by allowing more direct access to value fields. The VST user interface is wonderfully graphic, with many on-screen knobs designed to be tweaked with the mouse. But, this can also be a frustrating way of changing parameters, especially since the knobs don't have end-stops and can be quite easily overshot, resulting in a sudden (and mostly unwelcome) change from one extreme to the other of a parameter's range. Also it's still not always possible to simply type in the required values for all parameters. Those small niggles aside, it's now much easier to change many parameters.

Another neat new improvement is in the way the VST mixer can now be remotely controlled by the likes of a Yamaha 01V. Most of the parameter controllers in the VST mixer can be controlled by the equivelant pot or fader on the O1V (other digital control surfaces are supported). Furthermore if you're running Yamaha's DSP Factory you'll be pleased to learn that v4.1 VST/24 is now very compatible. A mixing controller window for the DSP Factory can now be called upon.

Thanks to the new Midi Retrospective Record feature, there's now no need to accidentally play something fab while idly poking at the keyboard, think 'that was good!' and then curse the fact the sequencer was not in record mode. Retrospective Record means VST is always listening and capturing Midi input, which it will save if asked to. The facility has its limitations (the amount of material the buffer will record is restricted), but it's a bonus nonetheless.

We haven't the space to mention all of v4.1's enhancements, but it should be clear that this is a worthwhile upgrade. Once more, Steinberg is demonstrating its commitment to the continuous improvement of what is already a powerful sequencer, as well as the forward-looking qualities which makes them among the hippest of software developers. Steinberg is eagerly embracing the approaching reality of software instruments running alongside Midi+Audio sequencers, and their particular talent for setting standards and getting other software houses to come on board assures VST a sound future.

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• *RRP: $795 (VST); $995 (VST Score); $1295 (VST/24).*

EQUIPMENT TEST

Yamaha MSP5

Christopher Holder makes room on his meterbridge for Yamaha's new nearfields.

The NS10 phenomenon has been well documented. But if there is only one fact that should be noted about Yamaha's NS10 and the NS10M studio monitors it's this: they're everywhere. That unmistakable white speaker cone can be found atop meterbridges world wide. With the spectacular sales of the NS10 it's not surprising that Yamaha has not bothered to pour their R&D money into new nearfield technology. But times change. In the last five years active and powered monitors have become very popular. Manufacturers like KRK, Genelec, Mackie and Event have all played a part in popularising the speakers/amplifier combination. The benefits of active and powered designs is also well documented. A few of these benefits include: optimal matching of the amplifier with the drivers, negligible loss of signal integrity due to the elimination of cable runs from the amp to the speakers, protection of the electronics from spikes with the use of active crossovers, and, of course, portability.

Maybe its because Yamaha has seen a drop in the sales of NS10s (although I'm assured by Yamaha's pro audio product manager, Mark Amory, that NS10 sales are as bouyant as every), I'm not sure, but, whatever the reason, they've busted back onto the monitoring scene with a new active monitor design: the MSP5. Of course, with the release of a new Yamaha monitor the world at large raise their voices as one and declare, 'Yamaha have made an active NS10!'. Truth is, they haven't (neither was it their intention I should imagine), but then I'm sure Yamaha won't mind the implication of the statement – namely market domination with one ubiquitous product.

The MSP5 is a small monitor. It's less than a foot high (in the old money) and about half a foot wide. It's built into a die-cast aluminium, magnetically shielded enclosure. The finish is matt black. It looks good and sturdy, and at 7.5kg a piece it feels like a substantial piece of studio gear. Each MSP5 uses a custom designed five-inch woofer and a one-inch titanium tweeter surrounded by a wave guide horn, which is designed to offer uniform dispersion. Each unit is powered by a 40W amp for the LF driver and a 27W amp for the HF driver, with a crossover point at 2.5kHz. While we're talking specs it's interesting to note that the MSP5 runs quite flat all the way up to 40kHz. With this in mind, it's also interesting that Yamaha suggest that these monitors are ideal for mastering applications where high sample rate recordings are regularly assessed – if you're recording at 96k, I suppose it's worthwhile having monitors that can reproduce those upper harmonics (but I won't be drawn into that debate right now).

It was a pleasure to use the MSP5s. My first reaction was being impressed with the high output from such a small monitor – they can really pump. The energy emitted is such that you can be listening in the next room (of a regular plastered wall studio or house) without any apparent loss in high end and upper midrange! They're exceedingly illuminating on most mixes, they flesh it apart as if with a sonic scalpel. The result may not always be a completely pleasurable experience, but certainly a revealing and instructive one. There's a decent extension in the low frequencies, with a good allusion of deep bass. The quoted lower frequency range figure is 50Hz, which is entirely respectable for a speaker of this size.

Fact is, there are many smaller studios that need smaller monitors like these. I believe that many small project studios can get themselves into trouble with having too large a speaker. The MSP5s are ideal for small spaces. They also feature a four position LF trim switch and a three position HF trim switch on the back panel, so getting the sound right in your room is made easier. I also like the portability factor, the MSP5s are genuinely portable, and could be integral to a 'go anywhere' recording setup. Yamaha are individually boxing the MSP5 as well, and, if sold separately, I can see them becoming a cost effective solution to small scale surround sound setups. It's not worth dwelling overly on the NS10 comparisons, but suffice it to say that if you are familiar with the sound of the 'old favourites', then it won't take too much to accustom yourself to the sound of the MSP5. The sound isn't perfect, I certainly felt a little underdone in the low mids, but it's difficult to complain too much when you have excellent stereo imaging and definition in the lows and highs. These monitors are priced well and sonically have much to commend them – well worthy of an audition. AT

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Price

- *RRP: $1590 a pair*

Korg Triton

Korg continue to hone the workstation concept they made famous with the M1. Gordon Reid gets hyper integrated.

EQUIPMENT TEST

Everybody seems to be talking about the Triton, which is not surprising… Korg defined the affordable workstation when it introduced the world-beating M1 in 1988, and the O1-series maintained Korg's position well into the early '90s. A number of other manufacturers seemed to be catching up towards the middle of the decade, but in '95 Korg demonstrated that it had lost none of its innovative zeal releasing the Trinity. The Trinity offered physical modelling, digital I/O and hard disk editing options, and proved to be perhaps the most revered synthesiser of the last few years. So maybe it's not surprising that many players see the Triton as 'Trinity 2: The Sequel'. Indeed, it's hard to approach the Triton without referring to its predecessor. It shares the same brushed aluminium appearance, and is dominated by a similar, large, touch-sensitive screen. But what's this… knobs? And buttons marked 'arpeggiator' and 'sampling'? While the cosmetics are out of the Trinity mould, the guts of the Triton are significantly different. So let's cast preconceptions aside and dive in.

The Basics

The Triton is a 62-note polyphonic, 16-part multitimbral digital workstation. Like all Korg's recent flagships it comes in three models: the basic 61-note Triton, the 76-note 'Pro', and the weighted 88-note 'Pro-X'. All the models share the same 'HI' (Hyper Integrated) sound engine, dual arpeggiators, sequencer, high-density floppy drive, MAC/PC interface, and a sampler with (initially) 16MB of RAM. Irrespective of model, you can then add two PCM expansion cards, a six-voice multitimbral physical-modelling synthesiser, a SCSI interface, and up to 48MB of additional RAM for the sampler.

As always with Korg instruments, the Program is the fundamental unit of a sound. In the Triton this is a complex patch based upon (up to) four PCM multi-samples. In my experience Korg don't necessarily lead the way in good multi-sampling, and if you remove all the effects and filters from certain Triton Programs they suffer from a number of sounds that go ee-ee-ee-oo-oo-oo as you play up the keyboard. But there are 425 multi-samples and 413 percussion samples in the instrument's extensive 32MB ROM, and a good many of these are excellent.

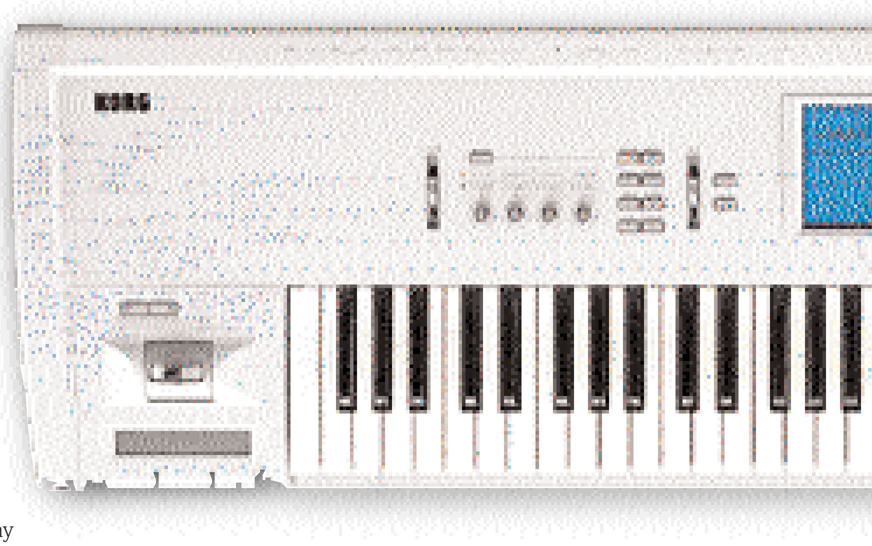
Once you have selected your waveform(s), you have the Triton's extensive Program architecture at your fingertips. This offers a bewildering array of filters, LFOs, and amplifiers, plus the most elaborate modulation matrix of any current synthesiser. But Korg's forte has, for the past few years, been in the effects sections of its synths. These allow you to take almost any voice and warp it into the most refined or outlandish sounds imaginable. The Triton does not disappoint. There are no less than five Insert effects busses offering 102 algorithms, plus dual Master effects busses that support 89 of the aforementioned 102 algorithms. There's also a three-band Master parametric EQ, and flexible routing from all the busses to the six outputs. Some of the effect algorithms are 'double size', and using these reduces the total number of effects and busses available, but the effects section remains a sound mangler's dream. You can even use the audio inputs to direct external sounds to the effects, making the Triton a powerful multi-effects unit and a vocoder in its own right.

Despite its potential complexity, the Triton encourages experimentation and, once mastered, it simply feels like a powerful analogue-style synth soaked in digital steroids. The signal path remains resolutely oscillator-filter-amplifier-effects, so it should strike terror in no-one.

As for the sounds themselves? A trip to your local music emporium will demonstrate that the Triton maintains – or perhaps extends – Korg's pre-eminent reputation for luscious pads and ethereal textures. There are also many excellent organs (no other manufacturer comes close to Korg's Leslie rotary speaker effect), genuinely deep basses, powerful brass and smooth string sounds, plus scores of other notables contained in the 512 Program memories. Oh yes, and there is also a GM bank compatible with both Yamaha's XG and Roland's GS formats. Is GM incongruous on an instrument of this cost? Yes, but it's potentially useful nonetheless.

Room to Expand

The Triton also has space for two 16MB PCM expansion cards, and there are currently two boards that you can slot into them. [News is that there are two new boards just released: Drum Loops and also Synth Waves/Dance – CH.] The first – Pianos/Classic Keyboards – majors on acoustic and electric pianos, and (if my sonic memory serves me correctly) appears to be derived from Korg's SG piano sounds. The second, called Studio Essentials, is essentially an orchestral board with a handful of choirs thrown in for good measure. Both cards have their pros and cons. Many of the electric pianos on the



Pianos/Classic Keyboards are excellent, while I can live without the acoustic pianos, an area which I consider has never been Korg's major strength. Meanwhile the Studio Essential board is more consistent and is, for my taste, a worthwhile addition to any Triton. As a bonus, each board comes with a diskette of new Programs and Combis.

Arpeggiators are becoming de rigeur on other manufacturers' synths, but the Triton is the first of Korg's flagship instruments to feature one. The company hasn't stinted on the specification. There are 180 pre-programmed arpeggio patterns, plus 20 user-programmable slots in banks A and B. If you have the expansion boards fitted, you'll find a further 16 slots in bank C, and yet another 16 in bank D. The factory patterns include guitar strums, bass riffs, drum patterns, brass stab patterns… it makes the standard up/down/random of other synths look paltry in comparison.

Each arpeggio offers up to 48 steps, and each step can host up to 12 notes simultaneously. You can use these to produce everything from analogue-style sequences, to full polyphonic-accompaniments, to movement within Programs that sounds very much like wave sequencing. I'm hoping that there will be a software editor for this soon (as there was for the Z1's almost identical arpeggiator) because, fully utilised, this will help you to open musical doors that you hardly knew were closed.

If PCM-based sounds do not satisfy your requirements, the optional MOSS board adds six voices derived from Korg's OASYS development system. In essence, the MOSS board is half a Korg Z1 and offers the same thirteen physical models as the earlier instrument. These include a selection of analogue synth models, plus FM, plucked and bowed strings, reeds, organs, and more. The expansion board can access all the Triton's Insert and Master Effects, offering greater flexibility than the Z1 itself, and it is six-part multitimbral, which is more than can be said of the mono-timbral version in the Trinity V3. Unfortunately, you must insert all 'effected' MOSS sounds through a single Insert Effect buss. This severely curtails the benefits of multitimbral use. But on the other hand, the six MOSS voices add to the HI system's polyphony, making the expanded workstation 68-note polyphonic.

Combis

Anybody who has played a previous Korg workstation will be fully conversant with the philosophy behind the Triton's Combis. This is where you can take up to eight Programs and map them to the keyboard in complex layers, splits, and multitimbral setups. For some reason, Korg has always done this more elegantly than other manufacturers, and the inclusion of the truly multitimbral Insert Effects structure makes this aspect of the Triton truly powerful.

The dual arpeggiators in each Combi reinforce the power of this architecture, because you can map each of them freely to the keyboard and the Programs within the Combi. Indeed, with careful programming, you can freely create Combis with (for example) complex drum patterns and bass riffs playing underneath your own real-time six-part multitimbral playing. This effectively places the Triton in the same league as most specialised auto-accompaniment keyboards and modules.

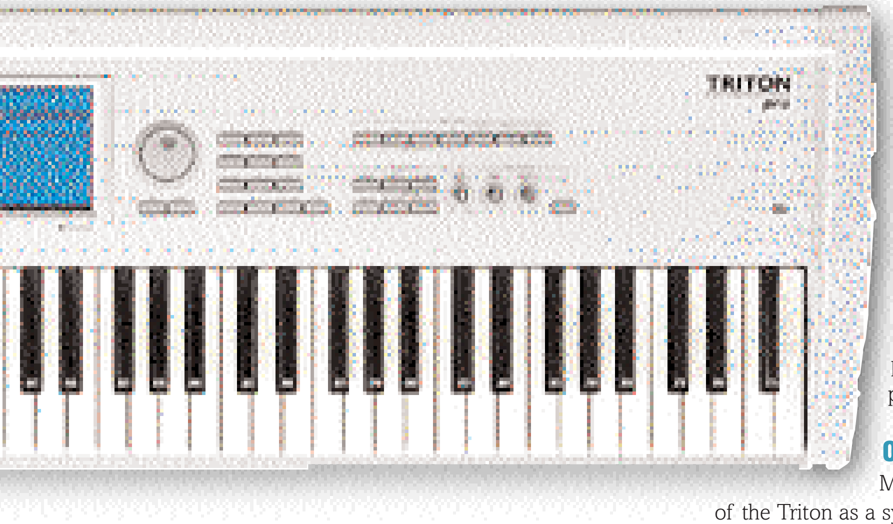
Tri Sampling

If all of the above was the full extent of the Triton's capabilities it would still be a big step forward. But perhaps the most exciting addition is that of the fully integrated sampler.

Korg is not losing its sampling virginity here: the DSS1 combined sampling, synthesis, and rudimentary effects as far back as 1986. But, despite offering sample options for the T-series and Trinity, Korg has not released a fully featured sampler for 13 years.

The basic specification is adequate enough: 16MB RAM expandable to 64MB, with 1,000 multi-samples available per 16MB bank. The sample rate is 48k, and down-sampling is possible after sampling to a minimum of 2k. You can sample directly from the audio inputs or through the Insert Effects section (neat!) and, once you have captured your samples, you can transpose them, truncate them, normalise them, reverse and/or loop them, apply rudimentary envelopes, cut rhythms to a tempo grid… in fact, almost everything that you would expect. You can also import Akai S1000 and S3000 format samples, PC-format AIFF files, and PC-format WAV files. These are in addition to the Triton's own format.

Okay, that's the good stuff. Now for the not so good. If you import an Akai program you will get the full multisample, with the keymap and a loop for each sample, but because the Triton isn't be compatible with many of the Akai capabilities I more often found myself returning to the raw waveform. Also, the Triton lacks the equivalent of Roland's 'Timbre' layer, so you can't assign filters, ADSR envelopes or key-scaling individually to each of the samples. Less damning, but omissions nonetheless, are the lack of 'alternate' and 'cross-fade' looping, and the absence of the time-bending capabilities of latter-day Akai and E-mu offerings. Furthermore, there are no S/PDIF or ADAT digital inputs for direct



loading from DAT, ADAT, or CD-ROM. As a subjective observation, I also found Korg's method for creating keyboard maps rather arcane. And finally, you can't export any of the Akai, WAV and AIFF formats.

Perhaps the 'Convert MS to Program' command displays Korg's true intentions for the sampler. This tells the Triton to translate the complete multi-sample – envelopes, loops, effects, and everything else – into a Program that you can treat identically to the ROM-based sounds. But there is a catch, the Triton will not save a single Program/Multi-sample combination and allow you to recall just that one Program with all the samples correctly positioned. This means that, under certain conditions, you must reassign the samples when you reload them. This is crazy… the samples' locations are saved when you write the Program, but if you don't reload them in the right order, the Triton does not appear to be able to determine which is which by name alone. Also, don't forget to save your samples before powering down, as the samples themselves are held in RAM, not Flash ROM. Speaking about saving, if you're going to take your sampling seriously you'll want to invest in the SCSI option from day one, because saving to floppy disk will soon become a drag.

If all this sounds a little critical, I may have overestimated Korg's intentions. Despite its faults and certain omissions, such as the lack of re-sampling, the Triton's sampler is very quick and easy to use, and provides a totally integrated way to get your sounds into the synth's Program and Combi structure.

Seq' time

Finally, we come to the sequencer. This offers up to 200,000 events spread over 16 tracks, and all the standard high-resolution composition, editing, and looping capabilities you would expect. You can load up to 200 Songs (or sections of songs) simultaneously, and a 'Cueing' capability allows you to string these into finished tracks and/or sets.

The sequencer includes three unexpected bonuses. The first is the 150 preset patterns that you can drop into sequences simply by pressing an appropriate key while the sequencer is running. You'll see this facility on the likes of Korg's N364 and iX300, and I found it makes it possible to build tracks extremely quickly and easily – especially in styles that may not be your forte. If the factory patterns are not quite what you are looking for, there are a further 100 locations into which you can drop your own patterns. The second bonus is the ability to sequence arpeggios and arpeggiated sounds intact. The third is the provision of sequence templates that provide starting points for your compositions.

Over The Moon?

My first, second and third impressions of the Triton as a synthesiser workstation were exceptionally positive. Okay, so it isn't quite as physically sexy as my Trinity Pro, but the keyboard does invite you to play, the Z1-esque knobs do invite you to twiddle, and the improved operating system overcomes many minor niggles. My feelings about the Triton as a sampler are, obviously, a lot more mixed.

On a technical level, the appearance of six audio outputs raises the Triton into the premier division, as do the dual audio inputs. Admittedly, my Trinity's hard disk recorder/editor and digital I/O options have disappeared but, on the other hand, what is the cost of an 8-track HDR today? Not much, in fact, probably less than an optional Triton upgrade would cost – so perhaps we should be applauding Korg for its astute market knowledge. Already some users have criticised the Triton for its lack of a serious on-board storage medium (a Zip drive for instance), and I tend to be sympathetic to that view.

In thirty pages I could tell you all about the Triton and convey a balanced view of all its great strengths as well as its annoying weaknesses. Unfortunately, three pages is hardly enough to give you a flavour. But you need to know this: the Triton sounds stunning and, despite a few omissions, offers more facilities than most players will ever fully explore. No, it's not all of a Trinity, a Z1, and an Akai S5000 in a single box, but it's not a million miles away, either. With its combination of superlative sound, expandability, and the sweetest user interface in the keyboard world, it deserves to be a permanent presence on any serious player's wish-list. Let's hope that software updates comes thick and fast, especially in the sampler department. If they do, I reckon that the Triton can be Numero Uno.

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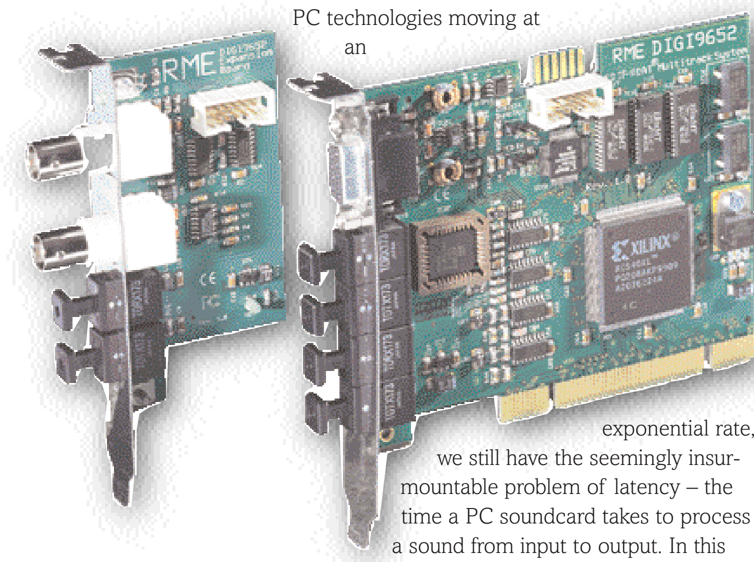
Price

- *RRP: $5,499; Triton Pro: $5,999; Triton Pro X: $6,999*

RME DIGI 96 Pad and DIGI 9652

RME have used new ASIO2 drivers to take their soundcards one step further. Guy Harrison goes hammer and tongs.

The dream of a recording studio in a PC with all its 'obvious advantages' has been with us for the best part of a decade. While the concept has obvious appeal, many of the pioneers in pursuit of this electronic Nirvana gave up in frustration. Problems with system overheads, poor HD access time, and newly developing software all contributed to their woes. In 1999, with PC technologies moving at an exponential rate, we still have the seemingly insurmountable problem of latency – the time a PC soundcard takes to process a sound from input to output. In this regard the PC still falls short of the mark, expecting users to put up with figures in the range of 50ms or more (ASIO), to hundreds of milliseconds (MME). Long latency makes transport controls feel sloppy, and it makes internal monitoring for the purpose of overdubbing impossible.



It's nice to know that RME Audio in Germany are equally fed up with monitoring their recordings out of time, and are the first company to include the new ASIO2 spec in the drivers for their new range of soundcards. ASIO2 is a Steinberg development first implemented in Cubase 3.7 for PC (just released), and promises 'no latency monitoring' of input signals. As an added bonus, RME has had the forethought to future proof their latest products by giving them all 24-bit/96k compatibility.

The new range from RME includes the DIGI96 (equipped with S/PDIF I/O only), the DIGI96/8 (S/PDIF and ADAT I/O), the DIGI96 Pro (which adds 20-bit analogue outputs), the DIGI96 Pad (includes 20-bit analogue inputs and outputs) and, the daddy of them all, the DIGI9652. The DIGI9652 is code named 'Project Hammerfall' and is devoid of any analogue I/O but supports up to 48 channels of ADAT I/O with wordclock as standard, and includes nine-pin ADAT sync. All impressive stuff on paper – let's see if RME can really make my software feel like a tape machine.

DIGI96 Pad

First up is the DIGI96 Pad, which is based on a standard sized PCI card and should pose no problem for those tight on space. The card features the popular ADAT optical interface plus S/PDIF and AES/EBU (on a nine-pin D-sub break-out cable). There's also provision to connect your CD-ROM drive digitally to the card, given your CD-ROM has a digital output. Analogue I/O is provided on two TRS jack connectors, one for stereo (unbalanced) input and one for stereo (unbalanced) output, these are -10dBv by default, switchable to +4dBu by way of a jumper on the card. An LED is provided which lights red when no valid digital input is found, extinguishing when one is. While not terribly exciting, this is a handy troubleshooting tool which could save a lot of head scratching. RME also supply an optional wordclock module for the DIGI96 range, which adds wordclock on BNC connectors and also provides an easy means of synchronising multiple cards. This is a small companion board that doesn't take a slot on the PC's motherboard, but connects to the DIGI96 card with a pair of three wire cables.

Installation

Installing the DIGI96 Pad was a dream. After physically installing the card and turning the computer on, Windows happily found the card and went into driver wizard mode. After a little prompting to let Windows know where the driver was residing (on CD), we were away. RME claim to have done their homework when it comes to simplifying card installation through the use of complete IRQ sharing and, to their credit, it seems to work. The card was installed in no time – a small card icon in the system tray being the only sign anything had changed. Clicking on this icon launches the DIGI96 Pad's ASIO and Settings control panel. The ASIO controls allow for the choice of four settings: 16-bit/46ms, 32-bit/23ms, 16-bit/12ms and 32-bit/6ms. While RME recommend the 16-bit/46ms setting for most stable operation, at 48k the DIGI96 Pad had no problem managing 32-bit/23ms on a (relatively) humble 200MMX PC, and happily ran eight tracks at the 32-bit/6ms setting on a P3 400MHz machine.

The Settings controls make light work of synchronisation issues through some clever options. Standard sync features are all here. A list allows you to nominate which input to use for synchronisation purposes, while an Auto Select option will continually scan the DIGI96 Pad's inputs looking for a valid signal, locking on when one is found. Auto Select used in conjunction with the Autosync feature means that after the card finds a valid input signal matching the playback frequency it will automatically switch from internal reference (master) to the clock arriving at its input (slave). All this allows for fail-safe on-the-fly recording from any input without having to be concerned with clock mode. This is the only card I know of that handles synchronisation in this way and it simplifies matters considerably, particularly for those unfamiliar with sync issues.

Padding Up

First up I tested the DIGI96 Pad in a 200MMX PC (64MB RAM) teamed with a Yamaha O1V digital desk with optional ADAT I/O fitted, and then repeated the tests on a P3 400MHz (128MB RAM) machine. In both machines the card performed admirably. Recording eight tracks through Cubase VST proved a simple affair and I must say that with ASIO2 direct monitoring enabled, it was quite easy to forget which computer system I was working on. That said, the 400MHz P3 machine had no problem achieving 3ms latency (and that's with VST compression and reverb running), while the 200MMX system really needed the 16-bit/46ms mode for serious multitracking.

RME has also done extensive research on busmaster technology to ensure very low system overheads, and this was evident on both systems. While the ASIO2 direct monitoring is a definite plus, it must be appreciated that with the DIGI96 Pad this option is essentially a hardware bypass – so your mixer in Cubase is no longer active for pan and level purposes on that channel. While this is a slight drawback, the fact that you can switch the hardware from within Cubase is a major step in the right direction. As RME's drivers are the first to implement the ASIO2 spec, I'm sure future refinements will soon see total control from within Cubase.

The DIGI96 Pad also performed well in the sound quality department. At 48k it sounded more detailed than my current 20-bit soundcard (with externally mounted A/D conversion). Switching to 96k, the sound was nothing short of stunning, with the top end taking on a level of clarity and openness that until now I hadn't heard in any sound card.

The DIGI96 Pad is a winner on all fronts. I'm sure other manufacturers will be rushing to update their drivers to take advantage of the ASIO2 spec. For now though, RME has a card and driver combination that seems to point to the next chapter of hard disk recording. With its advanced busmaster technology keeping CPU overheads to a minimum, and direct or low latency options, I'm sure the DIGI96 Pad will please many. Highly Recommended.

Hammer Time

The DIGI9652 Project Hammerfall comes packaged in the same compact box as its cousin, and, like the DIGI96 Pad, the manual arrives on a CD in Adobe Acrobat or Word format for printing. Unlike the DIGI96 Pad, the DIGI9652 is a two card package – a main PCI card and a daughter-board. The 12.5cm main PCI card houses all the basic electronics, and offers two ADAT inputs and outputs along with ADAT sync and S/PDIF on a nine-pin D-sub breakout cable. A CD-ROM connection point is also provided. If you need no more than 16-channel operation this card will operate without the daughterboard.

The daughterboard itself connects to the main board by a short cable, and space must be made to mount the cards side by side. The daughterboard adds a third ADAT I/O and BNC wordclock connectors, as well as an LED that lights green when a valid digital signal is received at any input. As I mentioned earlier, the DIGI9652 has no analogue I/O, making it more appealing for people with a fair investment in ADAT lightpipe technology. ADAT owners interested in taking advantage of the on-screen editing offered by today's PC programs, and owners of digital desks keen to bypass the tape medium or unnecessary D/A and A/D conversion, will appreciate the ADAT I/O. It's also worth noting that with the newly released MME drivers you can be assured your DIGI9652 will work with any program available.

Installation

Much like the DIGI96 Pad, installation of the DIGI9652 was straightforward. When the computer was powered up the device was recognised, and once the drivers were installed we were ready for action. This time a pink hammer was hiding in my system tray and, once again, the icon provides quick access to Settings, ASIO and MME options. The ASIO settings can be adjusted from a latency of 186ms down to 1.5ms, though it must be pointed out that just because a setting exists doesn't mean your computer will be able to cope with it – lower latency means smaller buffer settings, and if you set the buffer too small for your computer's processing power, audio glitches will occur. I was able to achieve stable operation with a 186kB buffer, for 3ms latency on a P3 400MHz computer. Latency that low makes ASIO direct monitoring somewhat redundant, although people with lesser machines (that can't manage buffers this small) will still be thankful. As expected, RME has carried on the excellent AutoSync and SyncCheck features of the DIGI96 range (see above for full details) into the DIGI9652 and added a timecode display to deal with sample accurate ADAT sync.

In Use

For testing, I got busy flying some vocals off an ADAT XT into Cubase through the DIGI9652 – in order to clean up some of the vocalist's irritating breath noises – then flew them back onto the ADAT, all in sample accurate sync. This took a moment of configuring within Cubase because transferring audio with sample accuracy requires two levels of sync: sample frequency (wordclock) and sample position (timecode). With nine-pin ADAT sync connected from the output of the XT to the input of the DIGI9652, and 'ADAT 1' selected as the sync reference within the Settings controls, we were ready to make the transfer. This all took place with a minimum of fuss, and a lock up time of no more than one to two seconds was achieved. After editing, the vocal was returned to the ADAT with the same ease in which it was removed. The ADAT and computer performed seamlessly and, used in this way, made for a very powerful and inspiring setup. Simply put, a pleasure to use.

A Real Hammering

The DIGI9652 is a powerhouse card that delivers on its promises. The 'virtual' manual is well written and the supplied CD contains a swag of useful utilities which I unfortunately don't have the space to discuss in this review. The RME web site also contains some useful information (like how they manage to get 24-bit/96k data to share two 16-bit ADAT tracks), and if RME's driver update policy is to be believed then they seem committed to keeping things on the cutting edge.

I've always believed that if you get the small things right, the big picture will fall into place. With their Autosync and SyncCheck technology, RME has indeed got the small things right. Now maybe we can get on with the big picture – audio engineering. Hallelujah.

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Price

• *RRP: $1595 (9652); $1495 (DIGI96 Pad)*

dbx Quantum

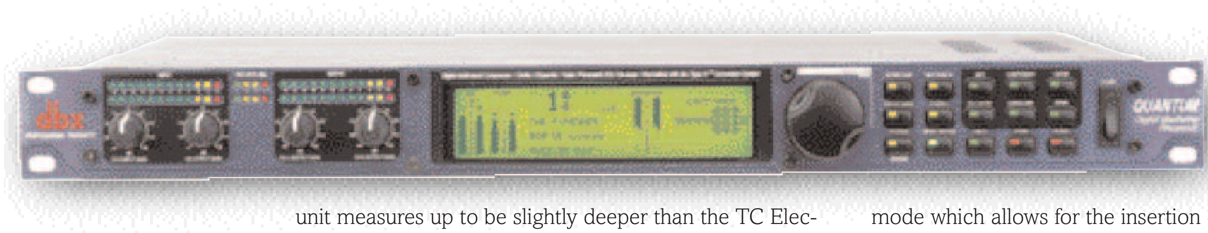
The Quantum is dbx's complete mastering solution in a box. Adam McElnea delivers a lesson in Quantum physics

The success of TC Electronic's Finalizer has been impossible to ignore, and companies such as Drawmer, dbx, and Behringer have all weighed in with their own versions of an all-in-one mastering box. The Quantum is dbx's effort. dbx has aimed to give the Quantum the jump on the opposition by packing in some serious DSP and serious functionality.

Hardware

The Quantum is housed in a 1U high rackmount unit with a brushed metallic gunmetal grey/blue finish. The unit measures up to be slightly deeper than the TC Electronic Finalizer Plus 96k. All Quantum parameters are adjustable via the display screen and data knob. The left hand side of the Quantum's front panel contains the analogue I/O section. There's two pairs of 12-segment LED meters (L/R channels) over each pair of indented level controls, and the Quantum's visual and mechanical analogue I/0 parameters are easily managed. Between these meters is another four-segment LED display indicating the operation of the Tape Saturation Emulation (TSE) – more on that later. The centre of the unit is dominated by the large LCD which has the task of providing most of the data information. Visual details include: peak and average digital I/O meters, current program information, I/O options such as TSE or Type IV conversion systems in operation, inverted multiband gain reduction metering, and finally, the parameters attached to the compressor/gate/limiter section, the EQ plots and the crossover band graphics. Further along the front panel is the large push button data wheel for selecting and editing parameters. Finally, there is a bank of 15 illuminated push buttons dedicated to calling up most processing and utility functions of the machine, and then a small mains power switch.

The rear of the unit houses a universal power supply (good thinking dbx), the digital I/O section – catering for both AES/EBU (on balanced XLR) and S/PDIF (on RCA phono sockets) – a pair of BNC connectors for wordclock in and out, Midi I/O, and finally analogue I/O – on balanced XLR and unbalanced 6.5mm TRS jacks. Internally, the unit boasts an impressive amount of DSP, accommodating five very powerful 100MHz Motorola 56000 series chips. On-board A/D converters come courtesy of Crystal Semiconductor while D/A conversion is by AKM, which presently produce the only 'truly stable' 24/96k D/A chip.



Software

Quantum offers users three different signal path options. Firstly, in stereo linked mode, the Quantum can be used for multiband dynamic manipulation. This tends to be the generic approach undertaken by most other all-in-one mastering boxes. Additionally, while still in stereo linked mode, the Quantum offers users a broadband (single band) mode which allows for the insertion of additional processors. Finally, the mode which differentiates the Quantum from its competition: dual mono broadband mode – that is, two completely independent mono processors.

In stereo linked multiband mode, the Quantum offers a flexible five-band equaliser, consisting of two high/low shelf sections, and three parametric filters. The high and low shelf sections have adjustable slopes from 3dB to 12dB/octave, a frequency range of 20Hz to 20kHz and the option to cut or boost by up to 12dB. Likewise, the parametric filters cover a 20Hz to 20kHz range, offer up to 12dB of cut or boost and have adjustable slopes from 1.5dB/octave to a precipitous 96dB/octave. Additionally the Quantum provides curves that are user selectable between either an adaptive Q or a constant Q response. Whatever your equalisation demands, this is a nice option to have available. Furthermore, equalisation is switchable to either pre- or post-dynamics processing.

The Quantum's multiband dynamics block offers a whole range of compression, limiting, and gating with an extensive array of editable parameters. The Crossover section is managed via three individual crossover points, rather than the typical two, with selectable slopes between 8dB or 18dB/octave. This provides for the manipulation of four separate dynamic bands. Furthermore, you can separately audition the I/Os of each crossover band – a very powerful and tasty feature! Compression ratios range from 0.75 to ∞:1. That's right, there is an upward expander ratio offering a solution to the modern day problem of over compressed, transient starved material.

Bands can be individually or globally turned on or off, and there is an Auto mode for automatically setting compressor and limiter parameters such as Attack, Hold, and Release times in real-time. Manual manipulation of the dynamics parameters can, of course, be made on an individual band-by-band basis or globally, providing a high degree of flexibility and a refreshing change from the norm. dbx's classic OverEasy control is implemented in the compressor and limiter sections. OverEasy control offers a natural and smoother, soft knee-style approach to dynamic processing – making it generally easier to squeeze out higher levels of compression and limiting without annoying artefacts such as pumping.

In stereo broadband mode the fundamentals are preserved, however, a de-esser replaces the crossover and a side-chain equaliser (SEQ) is made available for the dynamics section, providing a frequency sensitive compressor/limiter. The de-esser, used for managing excessive amounts of sibilance, offers filter frequency selection between 800Hz and 8kHz, and adjustable filter types between band-pass and high-pass.

The dual-mono mode supports the broadband processing options, as mentioned above, with the clever inclusion of being able to individually treat left and right signals. This function alone vastly expands the scope of the Quantum. No longer restricted to just stereo material, the Quantum suddenly takes on a new identity. This is an extremely flexible device!

Furthermore, the Quantum includes a stereo adjust parameter for expanding the stereo image of narrow material, or vice versa. Fundamentally, this process boosts your out-of-phase or stereo information, thus providing you with an increased perception of audio space.

Another innovative feature provided by Quantum is the Ambience function. This process increases the audibility of the low level detail in a mix by essentially using a combination of gain and compression on the lower portion of the mix, resulting in an increased perception of 'Ambience' while leaving the transient detail intact. Ambience Amount can be set between ratios of 1.0:1 and 5:1 and a Width parameter manages the portion of the signal which the Ambience is applied to.

Additionally, the Quantum has a few more tricks up its sleeve. There is a TCM or Transient Capture Mode, as previously seen in the dbx 172 Super Gate. This process is a look-ahead function for achieving intelligent limiting, gating and transparent compression. The signal is fractionally delayed, allowing for the gate to open or the compressor to compress, thus resulting in superior dynamic processing with minimal side effects. Additionally, the Quantum offers comprehensive sample rate conversion, which includes converting sample frequencies down from 96k to 44.1k or 48k. This function allows you to take advantage of Quantum's high sample rate capabilities and apply them to the 'real world' of lower sample rates. It should be noted that true 96k sample rate conversion is extremely DSP intensive, however, the Quantum manages to deal with the task admirably. Which is more than I can say about some competing products which don't have enough onboard DSP to offer high rate sample rate conversion at all!

Quantum also offers a Normalizing function for adding gain to the signal just prior to the output. This function automatically maximises the Quantum's output without going over 0dBFS. This process can be undertaken manually or automatically and is managed via a hard or soft-knee clipper for the final output limiter. A standard on most all-in-one mastering boxes, the normalising function on the Quantum is sensibly placed as the last process in the chain.

Last but not least, dbx's revolutionary Type IV conversion system incorporating TSE (Tape Saturation Emulation) completes the unit's feature list. Without becoming too technical, the Type IV conversion system is implemented through the A/D converter circuitry and claimed to give more headroom and the ability to add Tape Saturation Emulation to 'over' signals. The resulting enhanced dynamic range especially with very transient material is quite impressive. The conversion system also allows for deliberate colouration of the incoming signal from bright to dark as well as a neutral algorithm. Although subtle, the effects provide a good starting point for immediate signal enhancement.

Quantum Mechanics

Setting up the Quantum was straightforward. The manual was reasonably informative and I especially enjoyed the quick setup guides. Even the manual's appendix which outlined the fundamentals of dbx's Type IV conversion system provided for an interesting read [teacher's pet! – CH].

Initially, I went digitally into the Quantum to objectively evaluate the machine's internals without being influenced by the sound of the converters or the TSE process. In Multiband stereo mode, I began testing a variety of premastered material in a variety of genre. Immediately, I was pleasantly surprised at just how warm

Presetting the Wizard

An approach pioneered by TC Electronic, the now obligatory 'Wizard' function assists users in managing the daunting task of parameter selection and setup. The unit 'asks' questions that encompass topics such as: the type/style of music you want to treat, the capacity in which the treatment should be applied (mixing, mastering or tracking), and how heavy you want the treatment to be. 'Hey presto!' the Wizard function then assembles a setup in an appropriate manner. Intelligently, the musical style dictates the manner in which certain processing is performed. For example, 'heavy' compression on classical music is less than 'light' compression on rock music. Using the Wizard doesn't preclude you from tweaking the parameters manually..

If magic tricks don't grab you, an alternative to the Wizard function is to simply select one of the factory presets and go from there. There are around a hundred in all, showcasing a vast selection of styles and approaches, from classical, rock, jazz, pop, hip hop and dance. Also included is a large number of dual-mono combinations such as country vocal/twang E Guitar and fat kick/slam snare.

and focused the unit sounded. This is definitely the most transparent multiband crossover I have heard to date. After dialling up a few alternative presets (refer to: Presetting the Wizard boxout) I found myself quickly reaching for the manual controls as I felt the presets were a little too conservative. However, the Wizard approach did provide a solid foundation to work on, and it's best to have these presets erring on the side of the conservative than being too extreme. Getting around the various functions was straightforward and quite intuitive. The Multiband Audition mode became immediately invaluable and I found myself suddenly wondering how I ever managed without it. This function was extremely flexible for pre and post monitoring as was the 'edit all' or 'single band' mode approach. The crossover parameters were functionally and visually good, while having selectable crossover slopes provided for an extra level of creative control.

I initially used the multiband compressor as a type of 'dynamic' EQ with fixed adjustments to the make-up gain section. The results were impressive. Dance music sounded superb through the dbx. Setting the low crossover band to around 60Hz to 80Hz gave a formidable 'thump' unattainable via other multiband systems that only offer three bands of compression. dbx's extra band provides for further tailoring of the low mids/low end region, therefore achieving a tighter, deeper bottom end. On the other source material, the multiband compressor performed just as impressively. The multiband gates were as equally good and the auto parameter function provided for a quick setup approach. Of course, if you are looking for a specific effect, it's easy to get in and adjust the parameters manually. Unfortunately, I found the multiband limiters left a lot to be desired. Likewise the normalisation function, compared with competing units, was underachieving. Possibly a 'stronger' soft clipper may need to be put in place, to appease the level junkies. The equalisation section was flexible and pleasant sounding. Used appropriately, I can see the equalisation meeting the majority of requirements. Finally, dbx's 'extras' functions, such as the broadband stereo width adjuster and the Ambience process, strongly complemented the entire package. I especially enjoyed using the Ambience process on acoustic/live mixes – the results, although at times subtle, added a wonderful 'openness' and provided an innovative solution to sound stage enhancement.

In stereo broadband mode the dbx performed flawlessly. The addition of the de-esser in the signal path provided effective sibilance control and was also very handy for managing unruly high/mid percussion. Although this addition comes at the expense of the multiband processing, the frequency selective sidechain approach provided for a sensible compromise.

However, it was dbx's dual mono mode that truly extended this unit's functionality. I was able to beef up separate L/R kick and snare patches in no time. There are a host of applications for this sort of individual instrument enhancement, which gives the Quantum more uses than just in the mastering suite.

Finally, it was time to test the converters. Initially, I began by using transient rich material so as to evaluate dbx's TSE. The results were very pleasing. In moderation, the TSE, Type IV conversion system provided for a significant increase in A/D conversion levels with minimal artefacts. The A/D colouration options added to the unit's sonic pallette, but, after experimentation with certain types of music, I found myself safely sticking with the neutral approach. The converters themselves sounded very good – transparent, with impressive imaging, and were comparable to current upper-range professional boxes with 'upper-range' price tags.

Quantum Leap

dbx has done their homework on this product. There is a lot of power in the Quantum. Boasting an impressive 48-bit internal data path, the dbx requires no inter-processing word length reduction, therefore resulting in a more warm, open sound.

However, don't overlook dbx's limitations. The Quantum, unlike competing devices, doesn't offer a memory card option. Therefore use between studios becomes an 'under the arm' task. Also remember there's no level compensation bypass mode, harmonic generators, tube emulators, or multiband image spacialisers. Additionally, Quantum's normalising and limiting functions seem a little weak. Therefore, if sheer level, speaker blowing presets and brickwall limiting are your mastering requirements, possibly consider an alternative device. However, if you require pristine professional audio results with enhanced, flexible multiband dynamics manipulation, innovative functionality, true 24-bit/96k sample rate capability, increased flexibility through dual-mono operation and a competitively priced all-in-one mastering solution, then the dbx Quantum deserves serious consideration.

Fundamentally, mastering has always been about what sounds the best, and in my opinion Quantum has sonically taken the format of the all-in-one mastering devices up a notch. Suddenly my chequebook doesn't seem as safe as it did prior to this review.

Specifications

Frequency response: 20Hz to 20kHz within 0.5dB

THD and noise >0.002%

A/D: 24-bit/96k, dynamic range: 114dB (A-weighted) or 127dB (A-weighted) with Type IV conversion

D/A: 24-bit/96k, dynamic range: 115dB (A-weighted).

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Emagic Logic Audio v4.0

A new look, loads of plug-ins and 24/96 capability are some of what's included in this major upgrade. Scott Christie rings in the changes.

Logic Audio has undergone its most extensive upgrade to date and the result is Version 4.0. It features a new set of fundamentally redesigned real-time effects plug-ins, including EQs, gates, compressors, distortion, delay, modulation and reverb. High resolution 96k/24-bit audio capability is now available, as well as increased audio tracks, improved bussing architecture and effects plug-in routing. Along with the new audio features, a host of other improvements have also been implemented which improve the overall Midi, editing and score functionality of the program. Logic Audio 4.0 also boasts a new overall design – including the streamlining and restructuring of the menu layout. Version 4.0 also represents the culmination of the Emagic Audio Engine, introduced in Version 3.7, which includes a comprehensive set of new and updated audio drivers for use with Emagic Audiowerk, ASIO, Digidesign Direct I/O, Korg 1212 and Sonorus StudI/O audio hardware integration. EASI support, Emagic's hardware engine support, is now also implemented.

A quick overview of the Logic product range begins with the entry level MicroLogic AV and becomes increasingly sophisticated through Logic Audio Silver, Logic Audio Gold and ultimately the '24-carat' Logic Audio Platinum used for this review. Logic Audio Platinum features 96k/24-bit audio, up to 64 mono tracks with one audio card (or up to 128 tracks with multiple cards), 16 internal busses, support for Digidesign TDM (ProTools III/24) and ProTools (TDM), Akai DR8/16 (Arrange Editing only), simultaneous support for three Emagic Audiowerk8 PCI cards and unlimited multiple hardware driver support. For a complete comparison of the Logic Audio range, head for **http://www.emagic.de/english/products/logicline/compare.html**

System

In terms of system requirements, Emagic have all but given up trying to offer a straightforward recommendation as to which computer system would be most suited to running Logic Audio. Technology is simply changing too rapidly to make such a call. Suffice it to say, the demands of audio hard disk recording are such as to require the fastest CPU, largest amount of RAM, and the fastest/biggest hard drive you can afford. Slower machines should cope fine with Logic Audio's Midi and notation capabilities, however the number of simultaneously playable audio tracks, real-time audio effects and overall processing speed of recorded material are essentially dependent on the CPU/RAM/hard drive speed/size specs.

Having said all that, the manual does however provide a rough guide to which systems they consider suitable. For the Mac platform, a G3 with 128MB RAM is recommended, while for the PC Emagic point to a Pentium 200 MMX or Pentium II, high performance SCSI Controller and 128MB RAM. A separate SCSI hard drive with a maximum 10ms average access time and a minimum data throughput of 1.5MB/s is also recommended. The prickly question of which actual platform to go with is succinctly summed up by Emagic as 'ultimately a question of religion'. Amen!

Installation is straight forward, employing a CD ROM which installs the 4.0 version of the program and all support files. A floppy disk containing the latest version updater software is then employed, i.e. 4.0 to 4.04. Owners of the new blue and white G3 Mac should note that since installation requires a floppy disk drive – deemed as an external optional extra by Apple designers – you may have to access Emagic's website at **http://www.emagic.de/english/support/download/oldimagemac.html**, where a 'virtual' floppy disk, known as a disk image, is available for download. For a comprehensive discussion of the new G3/iMac and how they integrate with Logic Audio, head for **http://www.emagic.de/english/news/mac.html**

Plug it in

Most of the excitement surrounding the pre-release of Version 4.0 involved the huge swag of bundled plug-ins, and, as such, was one of the first areas that I investigated for this review. I've always found reviewing software effect plug-ins, such as reverb and compression, a difficult job. After the initial 'wow isn't it great, my computer can do all this stuff' reaction, you're often left with the strong suspicion that having all this added functionality isn't quite the same as using the hardware equivalents. The makers of the software will quite rightly point out the impressive cost savings of running a dozen plug-ins bundled free with your favourite Midi+audio application over the cost of owning a dozen dedicated hardware effects units, but there is no questioning the inherent compromise in expecting your computer to do everything that a well equipped studio can.

The good news is with processor speeds increasing, the quality of these effects has improved dramatically – even just in the past year. Furthermore, clever software manufacturers such as Emagic are writing their plug-ins to exploit the advantages of digital audio, therefore offering effects processing impossible to achieve in the analogue world.

Version 4.0 of Logic Audio comes with 24 new effects plug-ins which cover the full range to be found in hardware-based effects units, and then some. The accompanying

Logic Audio manual comes with a well written overview of the functions and application of each of these new effects. The following summary of the plug-ins includes the most commonly used and the most interesting. Naturally, Logic will still support your favourite VST or DirectX plug-ins in addition to the bundled pack.

The enchantingly named Fat EQ offers up to five fully parametric bands of equalisation. Inactive bands can be disabled individually to save on CPU resources. A combination of high/low shelving, high/low pass and fully parametric EQ is available. I found the Fat EQ plug-in easy to use and sonically satisfying in its results. It also features a meaningful graphic display of your settings. My only gripe would be that the frequency selector in all five bands is limited to 100Hz increments. Apart from the fully functional Fat EQ there are a number of more simple, single band EQ plug-ins offering band pass, shelving and parametric functionality.

PlatinumVerb is Logic Audio's top end reverb plug-in. Other less CPU hungry, and therefore lower quality, variations include the GoldVerb and SilverVerb. Typically, reverb plug-ins are the most demanding on processing power since the phenomenon of natural reverb is such a complex process. PlatinumVerb features a basic early reflection parameter section and a dual-band reverb section with crossover controls. It also has a useful early reflections/reverb balance control which when set to 100% in either direction allows you to audition either the early reflection component or the reverb component of the sound and make its settings accordingly before blending the two. The actual factory presets, of which there are only six, didn't seem to do any favours in demonstrating PlatinumVerb's potential. The main problem being an inordinately high ratio of reverb relative to the actual direct sound source. However, after some tweaking it wasn't difficult to come up with suitable reverb settings for all of my sound sources – which included solo piano, solo acoustic guitar and solo drums. A comparison with the popular Waves TrueVerb plug-in led me to the conclusion that PlatinumVerb didn't quite match Trueverb's sense of stereo space and would benefit from increased parameter functions. The results I was able to achieve with the PlatinumVerb were highly suitable for sounds as part of a mix, though it wouldn't necessarily be my first choice on a featured instrument, lead vocal or solo piece.

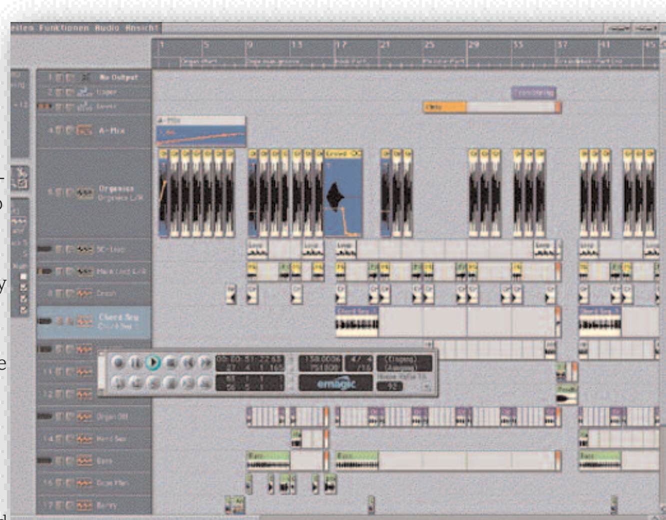
An innovative non-linear reverb plug-in known as Enverb is also included, which allows you to create reverb effects based on detailed control of the reverb tail's envelope. Attack, Decay, Sustain, Hold and Release controls form the basic controls for this plug-in, and as such provide a completely different approach to reverb building.

Logic Audio's Compressor features all the standard compressor controls along with a knee function which allows you to select 11 increments between hard and soft knee compression. Soft knee compression provides a more gradual transition from the 1:1 ratio below the threshold to the ratio above the threshold, resulting in a smoother sound than the instant hard knee compression. The analysis of the signal's threshold can also be determined by the plug-in in either RMS or peak mode. The RMS value is a better indication of how humans perceive loudness and therefore produces more 'musical' results. The peak value is useful when wishing to achieve a more 'brick wall' limiter effect. A useful Autogain function is available which automatically normalises the output of the compressor. The compressor plug-in sounded fine, managing to effectively control the dynamic range of the signal with a minimum of those nasty 'pumping' or 'breathing' artefacts, even when large amounts of gain reduction were applied.

A useful feature of the Noise Gate plug-in – specific only to hard disk audio systems – is the Lookahead function. This allows the noise gate to open up slightly before a desired signal passes through the gate and is especially useful on audio material featuring extremely sensitive transients. The plug-in does this by analysing the program level ahead of time and anticipates the point at which it can open the gate before the signal actually reaches the threshold value. The Noise Gate also features a basic filter sidechain that allows you to trigger the gate off a specific frequency band of the program material. This is useful for isolating a snare drum from a kick drum, for example, since these drums usually have their frequency energies in different areas. Unfortunately you can't use other audio signals as a sidechain input, as is possible with many analogue gates.

The Pitch Shifter plug-in manages to track reasonably well, but lacks any settings to 'intelligently' harmonise to a musical scale thus making its application limited. The Ensemble plug-in is what Emagic refer to as a 'Pitch Shifter on steroids'. It consists of eight internal, modulatable pitch shifters. Two standard LFOs and one random LFO enable you to come up with fairly complex pitch modulations which conjure up a thick ensemble-type chorus effect not unlike the classic Yamaha Symphonic patch.

A couple of filter/modulation effects include the AutoFilter and Spectral Gate plug-ins. The Spectral Gate plug-in allows you to tune in on a specific bandwidth in a sound while rejecting the frequencies above and below – hence the name – and then modulate the centre frequency and actual bandwidth. The Spectral Gate makes for some great filtering effects especially on drum loops and thus wins the 'Most Time Wasted Writing a Review But Actually Just Having Fun' award.



The nifty Enveloper Plug-in allows you to dramatically adjust the envelope of any input signal. The applications for this are endless. It allows you to effectively remove the reverb from a reverb drenched audio file, and increase or reduce the attack transients of a signal – thus adding snap to underachieving drum sounds or, conversely, de-emphasising the attack of rhythmically loose playing. The Enveloper becomes a very handy tool in the familiar 'of course we should have got it right when we recorded it but what can we do about it now' dilemma familiar to all engineers.

A quick 'rest of' list includes: the Direction Mixer, useful for processing acoustic recordings which have employed the

MS microphone technique; the Distortion/Overdrive plug-ins, which do a good job of 'nasty' distortion but I don't think any tubehead guitarists are going to sell their rigs here. The Stereo Delay and Tape Delay plug-in achieve exact delay/tempo matching by enabling you to choose a note value as a delay setting which is linked to Logic Audio's internal tempo setting as determined in the transport window. The term 'Tape' in Tape Delay refers to the fact that this plug-in applies a gentle saturation distortion effect to the delayed signal along with a high frequency roll-off to emulate the old analogue tape delay sound. An Oscillator Plug-in is also included for generating those metallic ring modulation effects. A final bit of good news on plug-ins is that you can now control all parameters of a plug-in via Midi, although the manual doesn't provide a detailed list of parameters and their corresponding Midi controller number. This is an impressive suite of plug-ins, and even if a number of them won't prompt you to throw away your hardware equivalent they constitute a powerful toolbox which gets results without your eyes straying from the computer screen.

Face Lift

An unmistakable change to Logic Audio 4.0 is the new look of the program, which I consider to be a nice touch. It incorporates modern clean graphics without burdening the CPU with over elaborate colour and 3D detail which can slow down the whole show. The new look also includes a restructured menu layout, designed to improve access to functions and simplify cross platform compatibility. The new global Options menu now makes for quick access to important settings previously buried in the Preferences and Song Settings sub menus, as well as containing all the tempo, synchronisation and marker operations.

Audio now also gets its own global menu which groups all the important audio functionality to a central menu. Access to functions such as Set Audio Record Path and Sample Rate selection – which previously required opening windows and selecting from sub menus – are now always only a mouse click and short scroll away.

The overall result of the menu restructure is a significant reduction in the number of mouse operations to get to important functions. These changes make Logic more accessible and more, well… logical to first time users who aren't familiar with the fundamental concepts, such as the Environment for example. Stalwart Logic users who employ their own virtual language of memorised keyboard shortcuts won't necessarily find Logic more suitable than before but Emagic have made a dedicated attempt to streamline the power of Logic into a friendlier experience.

Another important improvement to Logic Audio's interface is the Scroll in Play function. Scroll in Play keeps the SPL (Song Position Line) in the middle of the window while the background sequence objects scroll from right to left. The advantage is that your focus remains on the centre of the screen without losing the continuity of the waveform. In earlier versions, the SPL was continuously falling off the right edge and re-positioning itself on the left hand side of the sequence area. However, this new function does require a powerful graphics card and is therefore provided as an option to the old mode.

Other interface-based improvements include the ability to copy and paste Screensets from one location to another, which includes the ability to copy and paste between songs. Logic Audio 4.0 also has a completely new set of default key commands based on the key assignments of Logic users around the world. Key commands are now totally customisable – this includes conventional commands such as cut, copy and paste, and all User Defined Key commands are now displayed in the corresponding menu entries.

Midi Tracks

There's a good number of new Midi-based features. Simultaneous record on multiple Midi tracks is now available, which is activated by a new record button that appears on every Midi track in the Arrange window. When disabled, this record button also facilitates a Midi thru off option on each track. This would have come in handy recently when I was working with a multitimbral synth that didn't recognise the Local Off Midi message.

Multiple track Midi recording actually comes in two flavours – Layer and Split. Layer mode takes all incoming Midi events and sends them to all Midi tracks that have been record enabled. This enables you to quickly create layered sounds employing different synth/sampler modules and record the resulting blend, e.g. a piano blended with a choir and string section. After recording, only the selected track displays the recorded sequence. On the other tracks, aliases from the sequence on the selected track are displayed. In this way, any edits applied to the original sequence are thus applied to the aliases and all Layer tracks will remain identical.

Split mode is used to record multiple Midi sources to different tracks simultaneously. In this way a number of different Midi instrumentalists can be recorded onto separate tracks.

Other improvements worthy of squeezing into this review include: individual zoom on all Midi and audio tracks in the arrange window; a Pencil Tool in the Sample Editor which allows redraw of waveform to remove clicks pops and clipping etc; Hyperdraw, Logic Audio's graphics-based automation editor, now retains settings across newly created regions when snipping; plug-ins are now available across the stereo bus; Windows version now uses the Autoload song and automatically creates backup copies; Scissor and Glue tools are now available in the score; and, finally, several new parameters make it possible to now create and print the full score and all parts of a piece within one song file without having to change any settings.

Lab report

Logic Audio has always been a great program in terms of functionality and the ability to customise the interface to your own mode of creativity. Version 4.0 follows this up with enough new goodies to make it an essential update for all current users. The new audio plug-ins represent a marked improvement in terms of audio quality and interface design and the new menu structure combined with other interface improvements make it an even more desirable option for anyone about to get into the world of Midi+audio production software.

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Price

• *Logic Audio Platinum: $1499; Gold: $999; Silver: $499; Micrologic AV $279.*

TC Electronic Intonator

TC turn their gaze to vocal correction. Brad Watts hits the pitch hard.

Well, TC Electronic are nothing if not prolific! The ol' M5000 mainframe has given birth to a brace of brilliant boxes designed for everything from microphone preamplification and enhancement, reverbs and outrageous modulation effects, through to multiband compression and mastering processing. The FireworX blew me away some time ago, as did the Goldchannel, and the popularity of the Finalizer has reached almost religious proportions. I was beginning to wonder what on earth the Danish TC gods could deliver next. Lo and behold, they've unleashed another monster processor upon us – the TC Electronic Intonator. With similar ambitions to the rival Antares ATR-1 [AudioTechnology Volume 1, Issue 5], the TC Intonator is designed for pitch correction of vocals, with a few tasty extras thrown in.

Pitch intonation is a relative newcomer to the studio. While it has been possible to shift, stretch and re-record a vocal take in the past, it's often faster just to have the vocalist sing the out of tune vocal again. But then there's always the situation where the vocalist may have a solid performance marred by only one aberration or, worse still, is never in tune. Imagine the nightmare: the vocalist is a vocalist in name only, but the powers at be are telling you to make the so-called singer sound like a million bucks. The vocal is invariably the most important part of a recording, and generally takes the most amount of time and effort to devise. This is where a box like the Intonator is worth its weight in any expensive substance you care to mention – saving precious studio time and heartache for the engineers and producer. Pitch correction would also be ideal in a live situation, delivering a far more polished result should the vocalist get hit with a tomato and lose his or her concentration.

TC has employed the services of IVL Technologies, a company specialising in technology related to the human voice, such as pitch recognition, harmonising and karaoke. IVL clients include companies such as the Harman Music Group, Yamaha, Korg, Mackie and of course TC Electronic. Interestingly, IVL are the company behind the Electrix line of DJ oriented processors such as the WarpFactory vocoder [see our review on page 88].

As with all of the recent range of TC Electronica [very clever – CH], the Intonator sports a full complement of analogue and digital input and output choices. Two balanced XLR analogue ins and outs along with AES/EBU digital I/O, S/PDIF as coaxial RCA connections and ADAT in the form of optical TOSlink, and wordclock input to keep it all in step with the rest of the room. The back panel also includes a full house of Midi I/O for external manipulation of the Intonator's parameters. While the unit may have two analogue XLR inputs and outputs, it is not a stereo device. The Intonator includes a de-esser and a filter that can be set to function via the second I/O channel – useful for inserting other processors between the pitch correction and de-essing process. This scenario is not possible if the unit is run in Dual Mode. This mode sets the Intonator to handle 88.2k or 96k sample rates. This aspect alone puts the Intonator ahead of the analogue-only Antares ATR-1.

The build quality is typical of TC's one unit cases, i.e. impressive. If you approached a stack of their recent boxes from behind, you'd be hard pressed to spot the difference between each. But then… with all those boxes wordclocked and digitally connected they'd make an awesome effects loop/digital plug-in!

On the front panel are TC's familiar and logically laid out array of controls. You'll find analogue input attenuation with a single overload LED and 12 buttons laid out keyboard-style beneath a screen, showing the pitch of the incoming signal and the amount of correction being applied. Dead simple, and all the important information is right there as soon as the unit is on and accepting a signal.

The keyboard layout of the buttons can be used to adjust custom scales and decide whether or not a note will be corrected. Off to the right of the display is a rotary encoder for parameter adjustment, which doubles as an enter button. Very quick to use and to get used to. Next to this is a larger pitch knob for overall manual adjustment of the 'tune'. The next control adjusts the 'window' or area that an incoming signal has to be within before pitch correction will occur. Next up is Attack, which describes how quickly the correction process will kick in, and then there's a final control for the amount of correction.

To the far right of the unit are two controls for adjusting the two extra tools built into the Intonator. A de-esser with a dedicated bypass switch and a low-cut filter. Both are handy little additions for what the Intonator is designed for, processing vocals. The filter can be set to an 'adaptive' mode. In this mode the filter threshold will adapt to the input pitch or if used manually the filter will cut from 50Hz to 250Hz.

All of the Intonator's parameters can be controlled via Midi using continuous controller information, but what I found rather interesting was how the Intonator will output what it's doing as it processes a signal. Note number and pitchbend information depicts the pitch of the incoming signal and a stream of controller data reflects the amount of correction being applied. With this info recorded into a sequencer the engineer could make quick and very surgical edits to the pitch correcting process. Even as an analysis tool this feature would be useful. You could almost use the unit as a pitch to Midi converter at a pinch. Of course all the parameters can be stored as a system exclusive dump to keep a project's settings within the sequencers file.

Vocal Support

As with all of TC Electronic's gear the audio integrity is impeccable, with a dynamic range from analogue input through to output of 100dB (unweighted, by the way). Fortunately I had some badly out of tune vocals from a previous review of the Antares pitch corrector. I churned these recordings through the Intonator and was impressed with the results. The correction processing sounds true to the original vocal timbre with no detrimental effect on the original sound.

All of the digital I/O will handle 24-bit wordlengths and sample rates of up to 96k, making the Intonator relatively future proof. TC's dithering algorithms can be used to truncate a signal down to 8-bit if need be. With all of these digital options, any of this range of TC equipment can behave as a two-channel digital format converter.

There is little wrong with any of TC Electronic's processors and the Intonator is no different. The build quality is professional and the software does it's job without a hitch. Add TC's tactile and intuitive operating system philosophy and you have a very superior product. I'm sure many of this range are set to become classic processors, much like many other TC products have in the past.

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Last word

Tom Lubin

Akio Morita – Made in Japan

The recent and sad passing of Sony founder, Akio Morita, saw the loss of one of the greatest innovators and visionaries in the world of audio. From humble beginnings, Sony, under Akio's leadership, has grown to be one of the world's true corporate giants, and there's hardly a studio on the planet that doesn't possess Sony equipment – at the very least a Playstation! Over the years various items of Sony equipment have become the defacto standard in recording, post pro, MI, and broadcast. It's worth noting some of the contributions he and his company have made to the music industry.

My first example is quite recent, and rather than demonstrating Sony's uncanny prescience (which it showed with the introduction of the likes of the transistor radio and walkman, but more on that later), it shows how flexible the company is in its thinking. In 1987 the DAT recorder was introduced to the market. Sony optimistically expected it would replace Philips' Compact Cassette. But the major record labels saw DAT as a threat since consumers could make perfect CD and DAT duplicates. The record industry flexed its considerable legal muscle and blocked the wide introduction of the DAT on grounds that it would have a catastrophic effect on copyright ownership. The legal wrangling prevented major investment in the development of low cost DAT recorders and players. So Sony redesigned the unit and turned it into a professional product. (It's also worth noting that the opposition to the DAT format was led by record giant, CBS. The consumer DAT battle may have been won by CBS, but Sony won the corporate war. CBS was subsequently bought by Sony Music!)

On the other hand, another digital delivery system developed by Sony in association with Philips was embraced wholeheartedly by the record industry. The Compact Disc was introduced at the 1977 Tokyo Audio Fair. Japanese manufacturers flooded the market with players which quickly dropped in price. This new technology gave the consumer a reason to rebuild their music collections, and record labels to re-release on CD every imaginable recording of any significance. The consequence of this was a revitalisation of a world wide record industry. The 25 to 45 year old audience that the record companies had just about written off returned to the record stores to buy CD reissues of the music of their youth.

Even more recently, the fate of another Sony consumer digital delivery medium wasn't (unlike CD) met with open arms. MiniDisc was designed and released by Sony 1993 to offer the same functionality as CD in a more compact format. [A compact compact disc? I'm surprised they didn't call it CCD. When it was released that would've really confused the market, DCC v CCD! – GS] A compression algorithm (called ATRAC or Adaptive Transform Acoustic Coding) was devised to pull off this digital sleight of hand. The advantages of MiniDisc were many: it was smaller; with its casing it was robust; being a magneto-optical medium it was less susceptible to skipping; MD tracks could be titled in a table of contents; it had the capacity to be recordable; and Sony saw it usurping the role of the humble and out dated music cassette. Initially the consumer was indifferent. Sony continue to push the MD as a consumer format and inroads have been made, particularly as a Walkman/Discman replacement, but it is in the professional sphere that the MiniDisc has gained most acceptance. MiniDisc audio may be compressed but the convenience of the format meant broadcast and theatre have taken on MD, while, increasingly, recording studios will give their staff a portable MD recorder each to conveniently demo any on-going project.

Currently Sony continue to fight the good fight. Sony along with Philips are now in the midst of a struggle for the high density disk format supremacy. Sony's offering is the Super Audio CD, using DSD (Direct Stream Digital) rather than more traditional PCM to provide high bandwidth, high resolution two-channel and surround sound audio. Their plan is to provide hybrid discs which include a standard CD layer in addition to a new high density DSD layer, ensuring backwards compatibility with the millions of CD players already in the market. The struggle for supremacy continues, but if we can learn anything from the company's history it's that Sony don't mind a bit of a stoush.

Sony's first major triumph came early on in life and ironically it was based on an American invention the Japanese soon made their own: the transistor. In 1955 Sony brought out the first battery powered transistor radio and included an ear plug speaker. Soon after, it was so inexpensive that any teenager could own one, and its portability meant it could be taken anywhere. The availability of these radios provided access for a new generation of youth to hear a new form of music: rock'n'roll. Initially its light weight was seen by the public as indicating a lack of quality, so for the first few years Sony glued lead weights to the inside of their radios!

Another Sony product, the 'Walkman' has been one of the most successful consumer products ever made. In 1989 Sony produced its 50 millionth 'Walkman'. Its miniature headphones let the listener move inside their favourite music with the result that they were able to hear more of the subtleties of the production, a wider range of frequencies, and in particular stereo placement, separation and sonic effects – as well as risk permanent hearing loss! This in turn stimulated the producers to become more stereophonically aware and expand their use of the stereo image, and digital effects.

Just before the release of the product, it was officially named 'The Walkman'. There had been disagreement on the name by the US, English and Australian dealers who pointed out that it was bad English. The English wanted to call it 'The Stowaway', the Americans 'The Soundabout', and the Australians would have called it 'Freestyle'. But the eternal visionary, Akio Morita, would have none of it, and in 1986 the Oxford English Dictionary included 'Walkman' as a noun in the English language.