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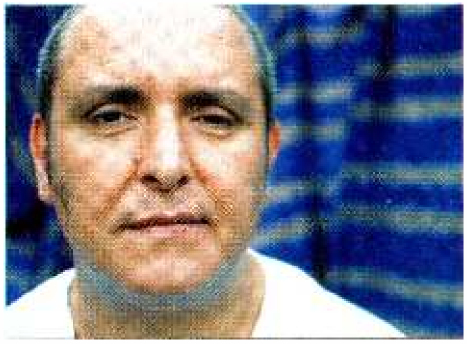
The story of the blues

AMERICA ONLINE recently ran a live online Q&A with Eric Clapton in which the guitarist gave his now increasing rare views on his music and his life. One observation struck a chord with me and was clearly close to his heart. When asked to give advice to a young blues player finding his way in today's music scene he replied: 'Go and find a vinyl shop, because its getting harder to find blues on CD. Go to a good music store and just search through the bins for obscure stuff, because the way it's going with video and the music TV channels, we almost don't know that there's a history to this music.'

What is most obvious about musicians like Clapton and others of his generation is that they all pay enormous homage to the influence of earlier generations of musician, and these mostly black blues players also had emotional and artistic ties to players who came before them. What we witnessed with the popularisation of the blues was an evolution of a musical form taken through natural progressions and across to different generations as a growing and living art. Yet the reality today is that if you wanted to do your own 'research' into this music's origins then you would have your work cut out to find the recordings that turned Clapton and his generation on. The same is true of a lot of folk music.

While you can buy the single and album of any of-the-moment boy or girl band in a multitude of formats and versions, a great number of seminal recordings are either no longer available on CD, or never have been. Much of this comes down to the condition, location and existence of the relevant tapes, yet you'll still



hear of master tapes turning up as found or held after decades of pitiful legal or monetary wrangles.

While I can concede that the majority of recorded produce doesn't even qualify as 'reasonable' in terms of sales success I have to believe that if it was important enough to record in the first place then it is important that modern distribution channels make it available as a priority.

Previously evolutionary musical processes have been superseded by processes of distillation in which ever greater focusing on particular musical genres has robbed us of the variety and richness of palette that characterised even my early years as a music consumer. Choices are reduced, variety is reduced and sources of influence are ever harder to come by. Is it any wonder that so much music is now so immediately and blatantly identifiable as derivative? They have nothing else to listen to.

Zenon Schoepe, executive editor

Media but no message

IN THE POST-IBC, post-AES battle zone, a large part of the clearing and rebuilding effort is necessarily directed at press releases. Like birthday presents from estranged members of the family, press releases containing uncertain pleasures lie around every corner—piled on the desk where you received pre-show news, on the bed where you unpacked your luggage, on CD-ROMs handed on by colleagues after the event, in email that has accumulated in your absence, in the following weeks' snail mail and the pockets of trousers left unwashed for too long. To order and assess this stuff is a major task; to determine the appropriate course of action for each release almost impossible.

On a good day.

On more mundane days, the process is further hampered by the need to deduplicate information that has been collected through more than one channel and to make sense of the obfuscations of peoples' filing systems.

Naming no names, those who have supplied such nuggets of PR as 'text_007.rtf' and 'kolpt51b.doc' during the recent show frenzy have not served their causes well. Given that the name of the game is publicity, every additional hurdle placed in front of a journalist reduces the likelihood of the news being spread. Other effective obstacles take the form of email in unfriendly file formats and with massive attachments. As for a war correspondent walking through the ruin of a city, the question 'why' comes unbidden and with monotonous regularity.

Why is it so often so difficult to make a press release accessible? Why don't text files bear a name that relates to their content? Why do picture files bear no relation to the text they accompany? Why do PR people regularly fail to place themselves in the place of their intended targets and structure their work accordingly?

Okay, the message here is tightly targeted at pro-audio media, but the moral of the story is more far reaching. In a world where communication is faster and wider than ever before, where the volume of communications far exceeds people's time to attend to them, transparency and brevity are precious attributes. It's the same on grander scales of human society, where we are unable to adapt to the social changes brought about by technological advance as quickly as they impact upon our ethics, morals and, ultimately, behaviour. The pace of progress has distracted us from the simple facts that helped us through yesterday. The sooner we recognise this, the sooner today will make sense.

Tim Goodyer, editor

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Lloyd Billing, Managing Director, on location in the oldest flying Tiger Moth in the world

Your sound effects and music samples are very valuable assets which can regularly be turned into cash. Whether you are a professional audio library with 50–50,000 sounds or more, or a sound enthusiast anywhere in the world that has the copyright to a 'unique' collection of high quality sounds, from a baby's first cries, Tibetan monks chanting, the kerrang of a guitar to a pin dropping. Your sounds have great value when you join the world's largest auditionable sound effects and music samples library.

The sound-effects-library.com, established as the world's No.1, where clients can audition, purchase and download instantly 'virtually any sound'. Together we can expand this further by co-publishing your sounds to meet the huge demands of an ever growing on-line market. For instance, one of our clients recently needed the sound of a 1930's Tiger Moth from an interior perspective. We did not have the sound, nor did any other library, so we went on location to record it. That's why we need your 'unique' collection of sounds; so we can offer virtually any sound from any perspective. Come and join us it's . . . free! Together we can maintain our position as the world's No.1. by having the most diverse range of high quality sounds possible.

When you join us you will be fully supported by a planned PR, advertising and marketing programme of 1 million US$.

Our marketing campaign consists of whole page colour adverts in specialist audio, internet, web design, B2B publications and related media.

In the PR field we have targeted all related journals world-wide, both national and regional. We are exhibiting at trade shows and will be at Las Vegas NAB 2001. We aim to emulate the successful picture library format to create a 'one-stop-shop' for sound effects and music samples. Please visit our web site to view our current press cuttings and advertising. All you need to do is contact us to unlock your valuable assets and turn them into cash.

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We look forward to hearing from you!

Lloyd Billing

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CONTRACTS

Netherlands recording studio Square Wave has become the world's first Amek Media 51 console installation. The Groningen-based studio swiftly put the 44-frame desk to use recording music and effects for a computer game from Team Sigma called *Enschende*, the DVD release of which will employ surround sound. The first US installation of the Media 51 has gone to Bernie Becker's studio in Van Nuys, California, whose 44-channel console is supported by a Miller & Kreisel 5.1 monitoring system and is the centrepiece of Becker's main mixing and overdub room replacing a 36-input API desk.
Sound Wave, The Netherlands.
Tel: +31 50 549 2349.
Amek, UK. Tel: +44 161 868 2400.

Slovak Republic The National Council has installed Maycom flash-based digital recorders as part of a system processing audio from Parliamentary meetings through digital recording, transcription, archiving and Internet access to transcripts, audio recordings and live audio broadcast.
Maycom, The Netherlands.
Tel: +31 481 377740.

New York Thirteen-WNET, a flagship PBS station, and a to-be-named audio postproduction facility have jointly purchased four AMS Neve Libra Series digital audio consoles having jointly designed two post rooms to bring audio production in-house. Thirteen's two Libra consoles will be used for live mixing and postproduction applications, and the new operation, scheduled to open next year, will use the Thirteen-WNET audio rooms as additional rooms for its audio post. Other networks who've also purchased AMS Neve Libra Live Series consoles include: ABC, CBS, NBC, Fox, Warner Brothers, and HBO.
Net: www.thirteen.org
AMS Neve, US. Tel: +1 212 965 1400.

Scottish Grampian Television has ordered a 24-channel Calrec C2 console as part of the upgrade of Studio B in Aberdeen. The C2 will be used on Grampian's North Tonight news programme, a daily live broadcast Monday through Friday.
Calrec, UK. Tel: +44 1422 842159.

LA-based 5.1 Entertainment Group has taken a second Soundtracs DPC-II console making it the fifth multi-DPC facility in the US. The new console forms the basis of a new

New media

US: San Francisco-based computer hardware manufacturer Castlewood has launched a new range of removable media drives which its UK distributor, Solid Storage Solutions, believes will challenge for supremacy in the music and video manufacturing markets.

The new-generation Orb drives offer 2.2Gb of removable storage and are available in SCSI internal and external, EIDE internal and USB external models. Average seek time is quoted as 11ms (read) and 12ms (write), with a 12.2Mb/s data transfer rate. Translating into 3½ hours of audio, over 2 hours of video and 3 CD-ROM games, these specifications have already attracted the attention of both Mackie and E-mu/Ensoniq.

UK representatives of these manufacturers are currently in talks with Solid Storage Solutions with a view to incorporating the drives into their products, notably Mackie's HR2496 multitrack recorder and E-mu samplers. Ian Charles, director of Solid Storage Solutions, expects others such as Roland to follow suit if the drives are accepted into the market.

'We want to show the end-user, rather than the reseller, the advantages of these drives, and we feel that the music business will give us natural links to film, video and animation,' he says. 'Castlewood in the States has already developed an OEM plan with Mackie in Seattle, whereby Mackie would badge the drives incorporated into the 2496—although the media would remain Castlewood-branded.

'Once E-mu realised that there was an internal option they showed an interest in replacing their 3.2Gb internal hard drives with the Castlewood EIDE Orb.'

Orb drives start at £111 (EIDE), and a single 3-pack of the 3½-inch disk media costs £55 (UK).
Solid Storage Solutions.
Tel: +44 29 20 610 102.
Net: www.solidstorage.com.

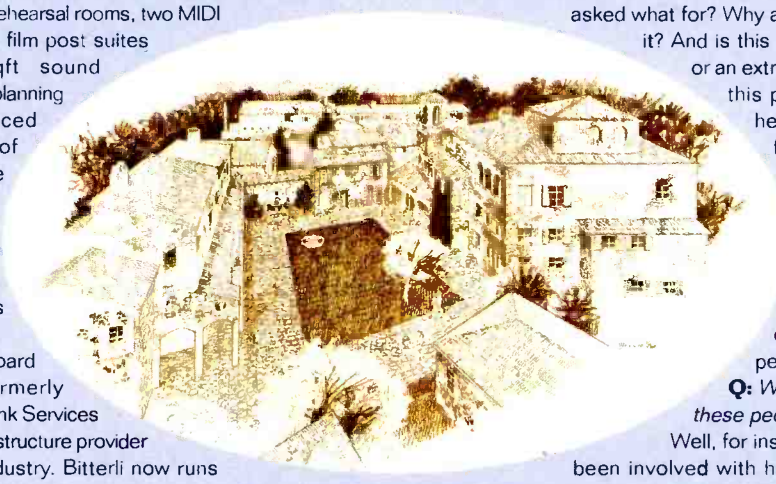
Foundry Increases Cast

US: Audio software pioneer Sonic Foundry has opened a Media Services Division, intended to service audio and video content owners with a range of digital management and delivery solutions. Building on its foundation in digital audio sequencing, recording and editing, the company has recently

Austin's power

RUPERT NEVE IS BACKING a studio build project for the first time in his career. The revered console designer, now settled in Austin, Texas, has teamed up with a consortium which includes Austin-born record producer Jay Aaron Podolnick and Bob Walters, one-time president of Media Sound and creator of Power Station, one of New York's best loved studios.

The Villa Muse project is an attempt to expand the curriculum of the recording studio way beyond its conventional limitations. Apart from an already generous blueprint of four large studios, eight smaller ones, its own theatre, three digital audio suites, two mastering rooms, five rehearsal rooms, two MIDI rooms, two video and film post suites and two 20,000sqft sound stages—Villa Muse is planning to exploit the advanced online infrastructure of Austin to place it at the forefront of media networking and delivery and to create an online university in order to exploit its cache of talent.



On the executive board is Judith Bitterli, formerly executive VP of Softbank Services Group, a £3 billion-infrastructure provider for the technology industry. Bitterli now runs Powered.com, a company that builds co-branded online universities for companies seeking to provide free educational courses for customers.

The site would also house an advanced audio laboratory for Neve himself, where he could further his research into audio technical solutions for Villa Muse and beyond.

A long-cherished dream of Podolnick's, Villa Muse has attracted Neve by its technical ambitions and by its location in Austin, for whose music and people Neve has great admiration. Speaking exclusively to *Studio Sound*, he outlined his hopes for the complex.

Q: *Why Austin?*

I settled in Austin about six years ago, and it's a beautiful city. In some ways, it reminds me of Cambridge, England—seriously academic, yet thrusting forwards in business and technology. I met some very fine, dedicated people in the course of lecturing in the city and visiting studios, and I noticed that there wasn't a top-of-the-range facility.

Q: *Is there a music scene?*

They have a tag for the city which you see as soon as you arrive—'live music capital of the world', and there are a great many very talented musicians around. There's a huge festival every spring when they close off the centre of Austin and stages are set up on the streets. Every bar along 6th Street has bands playing live, and it's fantastic.

Q: *Just popular music?*

No, there's the university, a concert hall and a symphony orchestra.

Q: *How did you become involved in the plan?*

I met Jay Podolnick, and discovered that his family used to own a chain of theatres across Texas. He told me his dream of building the best recording studio that there is. Now, I can't tell you the number of times I've heard people say this, so naturally I asked what for? Why are you telling me about it? And is this a commercial business or an extravagance? But Jay has this penchant for the best; he accepts nothing indifferently, and he puts his money where his mouth is. He introduced me to some people who I got to know and like very well. He's the sort of chap who gets the right people around him.

Q: *What convinced you about these people?*

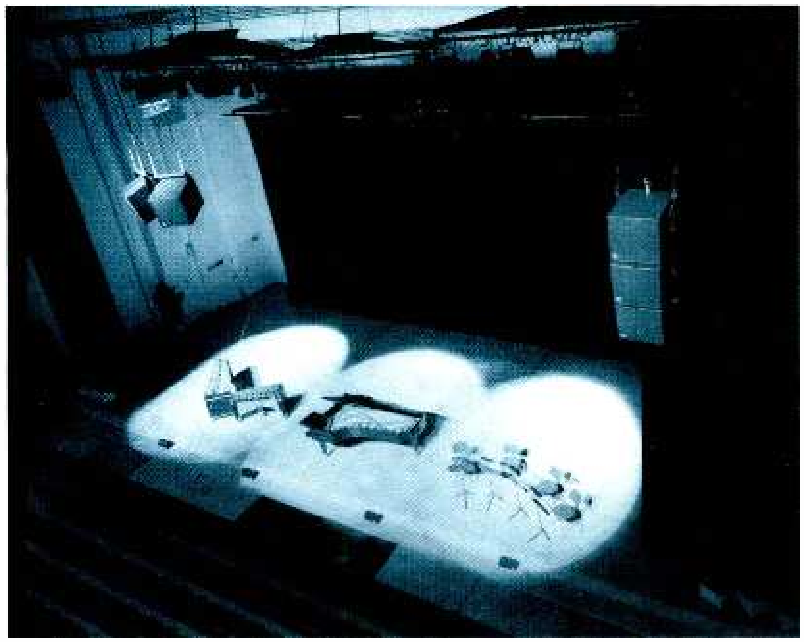
Well, for instance, Judy Bitterli has been involved with huge internet business developments and is top executive material, while at the same time being very charming to meet. She's an old friend of Jay's, too, which is a very solid bond. Another executive board member, Tom Frost, is the grandson of the founder of the Frost Bank, but when he first met me he said, 'Don't think of me as just a banker—I'm a frustrated musician'. So I realised that if I was ever going to do this with anybody, these were the right people.

Q: *How do you see your role?*

I can make sure that technically it's the best there is.

Q: *What stage is the project at now?*

Jay has optioned 550 acres of land about three or four miles from the airport. It's an area east of the airport, which is going to be developed into a new downtown area. We have the business plan completed, we have the design elements, and are now pursuing project financing. I'm an engineer, and engineers and business often don't mix. But engineers need the right partners, and I believe that with Jay and his crew I've got the kind of partners which one spends a lifetime looking for.



Sweden: The Malmo Concert Hall has taken six DPA Microphones Flamingo Series mic stand and six DPA Type 4023 cardioid mic stands. The microphones are the latest addition to the Concert Hall's inventory which recently saw a new d&b sound system installed by DPA's Swedish distrubutor Arva Trading. The DPA 4023s will be used on stage and for recording the Symphonic Orchestra as well as big band and jazz ensembles that perform at the hall.

expanded into video to become a multimedia specialist. and adds Media Services just as streaming media and broadband delivery are emerging as realistic markets for internet content.

Dan McLellan. senior VP and general manager of Sonic Foundry's Media Services Division, told *Studio Sound*. 'There are three cornerstones to the new division. The first is our ability to supply high-end audio and music encoding services. which we've been doing for quite some time. We have deals with several record companies and music aggregators in the States.

'We've also made two key acquisitions· STV, based in Santa Monica, an audio and picture encoding specialist which we're moving into our Media Services umbrella; and International Image, which supplies technological solutions to entertainment companies.

'International currently manages about 5,000 hours of programming content a year for clients such as MGM. Fox, Paramount, Warner Brothers and a host of key independents. It raises our credibility with the entertainment industry. and we will migrate our software expertise into this arena.

'We see the televison and feature film industries migrating from a film and videotape platform to a server-based platform.'

Sonic Foundry is also expanding its premises in The Netherlands. By January 2001 the company expects to be providing a comprehensive music encoding service for European clients. along with software support for its products. Currently six people are employed in Delft, near Amsterdam.

Satellite services

Asia: September saw the official switch-on of WorldSpace Corporation's digital audio broadcasting service. Singapore, India, and Indonesia are among the first countries being targeted, followed by Malaysia, Thailand and the Philippines early next year. Bringing CD-quality audio, the digital broadcasts will eventually be delivered by three geostationary satellites—two have already been deployed, with the third to follow soon.

The initial launch markets are countries in the African and Middle East regions. AfriStar was launched on 28th October 1998, while AsiaStar was launched on 21st March this year. The third satellite, AmeriStar, targeting Latin America and the Caribbean, will be launched sometime next year. Each satellite has three beams, with each beam capable of delivering more than 40 channels of audio, directly to listeners' portable radio receivers. Once complete, the global service will be able to transmit information, education and entertainment programming to a service area that will include an estimated 4.6bn people. Observers say that in the initial stages at least, countries like Singapore, Malaysia, India and the Philippines are more likely to be interested in the WorldSpace broadcasts, as radio broadcasts are expected to be largely in English.

WorldSpace radios equipped with satellite receiving antennas are already on sale in some shops in Singapore—with Panasonic's model costing US$390, JVC's $380, Hitachi's $295, while Sanyo's model is selling for only $299. However, WorldSpace is not limiting itself only to audio broadcasting. It will soon also be making a new multimedia service available. According to Tina Hubbell, Customer Service Manager at WorldSpace Corporation, the service will enable the download of huge amounts of the most popular web content, using a WorldSpace PC Card or USB Digital Data Adaptors. These will be directly connected to a computer, saving the content onto the hard disk, without the need for a telephone line. However, the company's multimedia system is a one-way broadcast and does not provide full Internet access, or television. The multimedia service, which is expected to be available in some areas by the end of this year, supplements traditional Internet services by offering popular, educational and informative web-style content with no 'per-minute' telephone charges. The service will include various subscription tiers, including those with free content, a basic service and premium channels. Even with download speeds as high as 128kbps, costs are expected to be considerably less than for standard Internet connection charges.

Headquartered in Washington, DC, WorldSpace was founded in 1990 to provide direct satellite delivery of digital audio and multimedia services to the emerging markets of the world.

WaveFrame converts

US: Following the partnership of the Lucid division of Symetrix with WaveFrame, Lucid I-O convertors will be available through WaveFrame's dealer channel for use with WaveFrame/7 (when it is released in November 2000) and FrameWorks/DX workstations. Lucid's ADA8824 (ADAT) I-O units will be available for use with WaveFrame's 16-channel digital-optical I-O and WaveFrame will also offer the Lucid 9624 A–D and D–A convertors for 96kHz, 24-bit resolution. WaveFrame President Ron Franklin said, 'WaveFrame is pleased to be working with Lucid to deliver its very high-quality convertors to our customers. We have tested the combination and it works very well to deliver a pristine signal path through the system. This provides an excellent value for WaveFrame customers.' WaveFrame has announced immediate availability for Lucid convertors for its FrameWorks/DX system.

IECEP 2000 Convention

Manila's PICC, or Philippine International Convention Centre, was the venue for this year's IECEP 2000, the Institute of Electronics and Communications Engineers of the Philippine's annual conference and exhibition (September 4th–7th. Having the

CONTRACTS

surround room intended to meet growing demand for the group's work on new recordings and reworking classical and jazz titles for surround release.
5.1 Entertainment, US.
Tel: +1 310 207 5181.
Soundtracs, UK.
Tel: +44 1372 845600.

Sao Paulo's Mosh Studios has opened a fifth digital mix room housing a 96-channel SSI Axiom-MT digital console to be used primarily for DVD mixes in 5.1 surround. The facility comprises four analogue recording-mix studios and attracts Brazilian artists of the calibre of Carlinhos Brown, Jorge Benjor and Milton Nascimento, with the new room already booked until the end of the year. Meanwhile, San Paulo's Midas recording complex has installed a 40-channel Amek Rembrandt console as part of an upgrade to its Studio 2. Artists may move freely between Studio 2 and the SSL-equipped Studio 1, with bookings up since the Rembrandt was installed. Brazilian studio designer, Carlos Duttweller, supervised the construction from plans by the Walters Storyk Design Group.
Mosh Studios. Brazil.
Tel: +55 11 3611 8206.
AMEK, US. Tel: +1 800 585-6875.
SSL. UK. Tel +44 1865 842300.

Korea's Munhwa Broadcasting Corporation has taken a 24-fader Studer D950 M2 Digital Mixing System complete with Studer's proprietary Virtual Surround Panning for HDTV dubbing and HDTV postproduction at its Seoul studios. Additionally, Korean Broadcasting Systems has purchased a 32-fader Studer On-Air 5000 digital console to be used exclusively for the station's FM channels.
Studer Revox, Switzerland.
Tel: +41 1 870 75 11.

Atlanta's Crawford Audio, a division of Crawford Communications postproduction operations, has opened a new studio centred around three 96-channel SSL Avant digital consoles networked through an SSL Hub Router system. Studio A has a 32-fader frame with Studios B and C employing 24-fader frames. The facility was designed and built by Tom Hidley for HDTV. DTV and DVD applications to service all audio formats for delivery mediums from broadcast to DVD.
Crawford Audio, US.
Tel: +1 404 876 7149.
Solid State Logic, UK.
Tel: +44 1865 842300

CONTRACTS

BBC's Radio Production Resources department has ordered a 72-channel Solid State Logic SL9000j analogue console for installation in its Golders Green Hippodrome studio, home of the BBC Concert Orchestra. The BBC has been operating SSL consoles for nearly 20 years, having been an early customer for the SL4000 series, one of which will be replaced by the SL9000. SSL, UK. Tel: +44 1865 842300.

New Orleans Trent Reznor's Nothing Studios has a MultiMAX EX monitor control system installed by systems specialist Paul J Cox who interfaced the controller between the studio's SSL G-series console and a 5.1 monitor system comprising KRK E-8 speakers and a subwoofer, with M&K's bass management system in-circuit.
Martinsound, US.
Tel: +1 626 281 3555.

Swiss radio station DRS has taken 25 Focusrite Platinum Compounders for voice processing at its studios in Zurich, Basel and Bern. The selection follows extensive evaluation and requires constant 24-hour use of the compressors.
Focusrite, UK. Tel: +44 1494 462246.

Manchester postproduction facility, Andrew Sumner Associates, has purchased two Digidesign Pro Tools AV systems, one of which is an AVOption XL. The systems were installed complete with a 24-channel Digidesign ProControl, and an SMM Pro Tools ControlUnit.
Digidesign, UK. Tel: +44 1753 653322.

Canada Canadian Broadcasting Corporation has installed a 36-input Audient ASP8024 console to be used in its A-Studio for live music and simultaneous broadcast, and music multitracking.
Expotus, UK. Tel: +44 1923 252998.

London-based engineer-producer Chris Tsangarides has assembled a new studio centred around a 32-channel TL Audio VTC console and Otari Radar recorder to record Gary Moore's performances in an adjacent rehearsal space for a 'virtual live' album. The sale joins others to Underworld in London, Cauldron Studios in Dublin, CX Music in Milan and Millbrook Sound Studios in New York.
TL Audio, UK. Tel: +44 1462 680888.

Sony Music Studios has installed a second Sony OXF-R3 Oxford digital console. The new 96-input board is

**UK: The Tape Gallery has branched in to on-line delivery of sound effects with the establishment of The SoundFX Gallery at www.sound-effects-library.com. Company MD, and *Studio Sound* consultant editor, Lloyd Billing said the Internet was an ideal delivery medium for the material and that models exist for guidance on the business plan. 'The most immediate analogy is photography where the print business went from dealing directly and commissioning photographers to dealing with photographic libraries,' he said. 'It's a billion dollar business and now around 20–30% of all pictures sold are sold for web use. I predict a similar ballistic for sound effects as the democratisation of digital camcorders, desk top video editing and website design will create an enormous extra requirement for sound effects.' The website permits sophisticated and refinable search routines and has on-line lower-quality auditioning of effects. Buying an effect instigates a download and a variety of purchasing schemes are available including subscriptions that permit unlimited access to effects for a fixed length of time. The aim 'to be the largest auditionable sound effects and music sample library on earth' looks well on the way to being realised with Crawfords, Valentino, Prosonus and Audio Interactive libraries already on-board together with many others. One of the interesting twists is that any recordists with effects that they have gathered on their travels can get them on-line with SFX Gallery, providing they can prove proof of ownership, in what amounts to a 50:50 split on any downloads. 'The potential is simply enormous,' said Billing, 'and the investment we have made in creating the infrastructure to deliver is simply massive.'
The Tape Gallery: +44 20 7439 3325**

theme, 'Onward to the Next Millennium', the IECEP event, part of its 50th anniversary celebrations, attracted as many as 300 attendees—around a third of them engineering students—and just over 40 exhibitors, including representatives of established international broadcast equipment companies.

Seminars on digital audio broadcasting, digital television, cable television, voice-over IP, WAP, e-commerce, and multiservice architecture for convergence often saw the meeting room packed with engineers and students of broadcast engineering. The accompanying exhibition had planned to feature product displays on telecommunications, networking and the Internet, broadcast and cable TV, multimedia, computers and information technology, cellular trunk radio, and value-added networks and services.

Although the range was certainly wide, some attendees expressed disappointment at the size of the show, and the number of visitors. Exhibitors included satellite companies Philcomsat, the Philippines Communications Satellite Corporation, and Mabuhay Philippines Satellite Corporation, fixed telephone line service provider PLDT (Philippines Long Distance Telephone Company), and cable manufacturer Duratube from the UK, along with mobile phone operators, and application service providers. Organisations that supported the event included the departments of Transportation and Communication, and Science and Technology, the National Telecommunications Commission, the Philippine Cable Television Association, and the Society of Broadcast Engineers and Technicians of the Philippines.

However, even with such a high-power line-up, Heinz-Michael Muller, president of Exi Systems Plus, a UPS supplier in Quezon City, declared, 'This being the first exhibition of its kind here, it was not as good as we had hoped, both from the point of view of the number of exhibitors and the number of visitors. But, yes, we will come again next year, providing we can be assured there will be more companies exhibiting, and more visitors.'

Ich bin ein Berliner

Germany: As of August this year the renowned recording and mastering studios at Universal Music in Hanover (the former Universal Recording Service) is to be known as Emil Berliner Studios. A division of Deutsche Grammophon (a Universal Music Company), the studios have previously been run primarily for the Universal labels, Decca, Deutsche Grammophon, ECM, Mercury, Motor, Philips, Polydor, Polystar, Zeitgeist, but will now operate increasingly for independent third-party customers.
Emil Berliner Studios, Germany.
Tel: +49 511 972 1232.
Email:Emil-Berliner-Studios@umusic.com

Who Needs Another Mic Preamp?

"...the finest sounding preamp I've ever used...as close to being the perfect preamplifier as possible. It is made well and it sounds unbelievable."

Russ Long, Nashville based producer/engineer, Pro Audio Review, June 2000

"The 1100 is the sweetest, cleanest, warmest, most flattering preamplifier I've ever used."

Jon Barry, Radio Personality, WMXB (FM), Richmond, VA

"The Aphex Model 1100 is a good example of something different... A work of art...The results were astonishing, providing an awesome sound that was natural, dynamic and absolutely free of noise."

George Petersen, Editor - Mix Magazine, April 2000

Model 1100
Discrete Class A
2 Channel Tube Mic Preamp
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from Aphex Thermionics

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CONTRACTS

housed in the facility's Studio G, a multipurpose suite offering complete music mixing and audio postproduction services for album projects, DVD, film and television. Sony Music Studios has also installed a 96-channel Solid State Logic Avant digital console.
Sony Studios. Tel: +1 212 833 5520.
Sony Corporation of America.
Tel: +1 201 930 1000.
SSL, New York. Tel: +1 212 315 1111.

Tokyo Broadcasting Systems has installed a customised Amek 9098i analogue recording console in its Yokohama City-based Midoriyama Studio complex. The 56-input desk is fitted with 40 mono and four stereo input modules, vu and Yamaki bar-graph metering, and is the centrepiece of Studio 5, a purpose-built HDTV facility, the first to be designed from the ground up for Hi-Vision format work on television drama. TBS recently refurbished and upgraded Studios M1 and M2, also at Midoriyama, and Amek 9098 custom consoles were installed throughout. These 48-input chassis consoles are being used for direct recording of spoken word and sound fx to D2 Digital VTR.

US. Signet Sound Studios, recently acquired by the largest audio postproduction group in the world, Liberty Livewire Audio, has installed Genelec 1034B active monitors for LCR duties in its 5.1 surround array in Studio A. The studio's history involves Bruce Springsteen, The Jacksons, Bette Midler and Stevie Nicks, as well as major motion pictures.
Net: www.genelec.com

London's Abbey Road Studios has chosen the SADiE Artemis and CEDAR plug-ins for its new 5.1 digital audio restoration suite and DVD preparation facility. The 5.1 restoration suite is configured for surround mixing conforming to DVD and is expected to undertake projects including the restoration, editing and pre-mastering of music for CD and soundtrack restoration of archive TV broadcast material.
SADiE, UK. Tel: +44 1353 648888.

American webcasters world are using Aphex Model 2020 dynamics-control units ahead of data-compression systems that provide high-quality Real Media and Windows Media(tm) streaming audio and video content. These include Westwind Media.com, ChristianPirateRadio.com, GratefulDead.com and Comedyworld.com.
Aphex, US. Tel: +1 818 767 2929.

France: Claude Sahakian's Paris-based Plus XXX Studios has placed the first order for the AMS Neve 88R analogue console. The 88R will be installed in Plus XXX's Studio 1 replacing a Neve VR and is part of the studio's refurbishment that includes improving the acoustics and adding a 5.1 monitor system.

Major mergers

UK-Europe: Preco Broadcast Systems, one of the UK's largest broadcast audio distributors has acquired the Studio Products division of Harris Systems. As part of the agreement, Preco now represents Harris audio in the UK, Ireland and France, ENCO Systems, manufacturers of the DADpro32 hard-disk automation system and audio processing manufacturer, Orban. The Harris sales office in France is part of Preco's expanded organisation. Preco takes on the commitment for after-sales support of existing Harris customers, and it is hoped that the transition will be as seamless as possible. Preco has also expanded the management of the company with two new board appointments: Tony Costello remains the managing director of the extended company and is joined on the board by Stewart Trussler as financial director and Sergio Dalmazzo-Auckland as sales director.

Meanwhile, the French Dalet Digital Media Systems and Coolpit (a subsidiary of the Denmark based E-Group) today announced the formation of a partnership to provide end-to-end audio solutions for web broadcasters, and to further develop the world-wide market for Dalet's multimedia content management applications. The partnership claims to combine Dalet's 'back-office expertise in the field of digital audio broadcasting and multimedia publishing, together with Coolpit's front-office know-how in the customisation of websites and the development of site management solutions'.

CEDAR Awards

World: The recent LA AES Convention saw the presentation of the 2000 CEDAR Awards, recognising achievement in the field of audio restoration for CD Remastering from a Modern Recording (post 1949), CD Remastering from a Vintage Recording (pre 1950), Remastering of a Film Soundtrack, Audio Restoration for Broadcast Use and Audio Restoration for Forensic Use. In sequence, this year's winners are Marina and Victor Ledin & Stuart A Rosenthal (US) with Tchaikovsky's *Piano Concerto No.2*. Recording (1944) using the Decrackle and NR-3 Noise Reduction processes on CEDAR for Windows; Simon Gibson at Abbey Road (UK) Studios with Franz Lehar's *The Merry Widow* (1962–63) using Decrackle and NR-3 Noise Reduction modules running as part of a CEDAR for Windows System; Novastar Digital Sound Services

Latest Classic

CLASSIC SOUND, the predominantly classical music editing and mixing facility based in the former Decca Recording Centre in North-West London, is relocating to purpose-designed premises a few miles away. Now under construction, the Munro Associates-designed HQ will transfer Classic Sound's main mixing and editing equipment wholesale, based around an AMS Neve Logic II console, B&W 801/802 monitoring and SADiE DAWs.

Founded by Decca engineers Neil Hutchinson and Jonathan Stokes via a management buyout following Decca's decision to close its technical facility in 1997, Classic Sound has built up a strong clientele not only for its mixing and editing rooms but also its considerable mobile recording operation. Using London's remaining orchestral recording spaces, the mobile operation makes use of two analogue Neve consoles, four Genex MO recorders and a battery of classic microphones and signal processors. The transfer of the company to its new location effectively breaks the last tie between the Kilburn site and its long heritage of Decca recording. Directors Hutchinson and Stokes spoke exclusively to *Studio Sound*.

Q: *The lease is up at Kilburn, so is it a straightforward swap, one building for another?*

NH: Definitely not. Not only are we expanding considerably in floor space, but we also have a firm intention to adapt and broaden our services. We always knew we'd outgrow the old building. Half of it is unsuitable for a floating floor, for example.

JS: The great thing about the new place is that the acoustic designer and the architect have got a clean slate - a big, flat floor. The old place is a little bit of a rabbit warren...

Q: *Are the new premises custom built?*

NH: No, but they're completely customisable. It's a very modern, adaptable industrial unit. When we first spoke to Andy Munro about what we wanted to do internally he immediately suggested an industrial unit—a big, open space, generally with a good solid floor, where you can basically do whatever you want.

The problem with most industrial units is that they're on great big industrial estates, but this one happens to be on a very small estate in a good residential area. The neighbours aren't typically industrial, either. There's a wine company right next door, for example...

JS: It's got parking, too, which is a precious commodity in London. There's also very good access from the tube and the North Circular Road.

Q: *What's going to change?*

JS: Phase one is a mix suite and two edit suites, just as we have now, but the mix control room is going to be much bigger. And there's room for a live recording area, which we'll phase in later on.

Q: *How much room, exactly?*

NH: Not full orchestral size: about 35-40 musicians. But the postproduction facility is the priority. We may put in a mezzanine to add more suites. There's a lot of space to expand into.

Q: *Will the equipment stay the same?*

NH: Basically yes, except that both edit suites will become 5.1 with plasma screens. At the moment the main mix room is 5.1 while the edit suites are stereo. The monitoring will remain the same, as in B&W 801 and 802s, but will upgrade to surround.

JS: We think 5.1 is now getting properly established. We'll encode to Dolby AC3, and we asked Andy to design specifically for 5.1 and picture work from the outset. So we're not just offering surround in a room adapted from stereo.

Q: *Are you sticking to classical music?*

NH: Not exclusively; the rooms are being designed for acoustic music, so that could include acoustic pop and jazz. And the picture work will include TV soundtracks, for example, which can be in any style.

APPOINTMENTS

Ron Bradshaw, former general manager of Euphonix's Spectral Division has joined Symetrix as product development manager. He will develop the Lucid and Symetrix brands, conduct product research and manage product beta testing. One of the original founders of Spectral, he has held a variety of executive positions during 14 years with the company, most recently he directed the teams on the R-1 recorder.

Amek has named Barry Sanders as American national sales manager of its Rack Products. Sanders will oversee

distribution of the Pure Path and Amek 9098 lines of outboard signal processing equipment, manage Amek's national dealer network for the products and participate in sales efforts for Amek's Media 51. Previously, Sanders was studio manager at Nashville's Sixteenth Avenue Sound.

DPA Microphones has recruited Steffen Leth Möller to its European team as area manager, working across the continent. A keen pianist and musician, Möller has collected qualifications in marketing, law and economics as well as fluency in English and German in addition to his native Danish. Prior to joining DPA Microphones, he worked as an export technician and spent five years with the Michelin Group in Denmark.

Tascam has appointed Stan Kaneshige Webmaster on its in-house marketing team. Kaneshige will be responsible for the ongoing development and maintenance of the site (www.tascam.com) as well as new satellite sites which will be developed in the future. Kaneshige joins after two years as Webmaster at Alesis, having previously been an independent media designer.

Graham Bell has joined Mediaspec UK as regional director of the company's operation for Scotland and the North of England. Mediaspec specialises in digital audio solutions for the music, broadcast and film industries in the UK.

(US) with Cary Grant's *Once Upon a Time* (1944) restored for Sony Picture Entertainment using CEDAR Series X units; Andy Pearce at Masterpiece (UK) with Bing Crosby & Louis Armstrong's *Lazy Bones* (1951); and Axel Thiel at The German Air Accident Investigation Bureau using NR-3 Noise Reduction to reduce noise and increase the intelligibility of Black Box, and other forensic audio recordings.

German demands

Germany: Dolby Laboratories has announced that German Video-on-Demand service provider, media[netCom] AG, will supply premium feature film content with Dolby Digital 5.1-channel sound in the near future. The Cinema-on-Demand system is a patented design that allows digital content to be streamed into set-top boxes with built-in hard-drives. Once stored locally in its entire length, the feature film can be played out by any set-top box carrying the Cinema-on-Demand trademark. The delivery path can either be a digital cable (DVB-C) or DSL network.

'True Video-On-Demand brings real progress for the consumer, and adding Dolby Digital 5.1-channel sound to our service completes the picture and gives the consumer what he expects from a system such as ours,' said Frank Hackenbuchner, CEO of media[netCom] AG. 'Market figures for home entertainment equipment clearly demonstrate that consumers like multichannel sound, and the demand is growing fast. With the rapid acceptance of DVD with Dolby Digital, the public will start to expect new digital television services to deliver digital multichannel audio,' commented Dolby's Tony Spath, marketing director, technology.

Dolby Digital regarded by Dolby as the backbone of home theatre—for DVD. Dolby Digital is a standard audio format world wide and it is the audio format for the ATSC digital television standard, the digital cable standard SCTE and is now included in the DVB digital television standard. The Dolby Digital soundtrack is encoded together with MPEG2-quality picture using the latest Dolby E technology, stored on media[netCom]'s central Cinema-on-Demand server or the local network provider's C-o-D server from which it is then distributed to the consumer's household over broadband ip-networks.

The Virgin's tale

UK: Virgin Radio is teaming up with Manx Telecom (a wholly owned subsidiary of BT) to operate a trial Digital Radio licence on the Isle of Man, using a Digital Radio technology service provided by Crown Castle. Discussions have already commenced with the Isle of Man Communications Commission which licenses and regulates broadcasting on the Island, and the UK Radiocommunications Agency about necessary approval. Scottish Media Group, Virgin Radio's parent company, will present a new digital-only service—currently named Manx Choice. This will be the world's first interactive radio station: radio programmes playing out will be selected by the public on a daily basis, using third-generation technology. Listeners will be able to vote for and influence a daily playlist of music and introduce additions to the playlist, using 3G devices and the station's website. The most popular tracks will be played at the highest rotation, with users being able to interact using the Internet. In addition, it is hoped that the multiplex will broadcast a range of digital radio services, including Manx Radio, Virgin Radio in their existing formats, as well as new variants of Virgin Radio that will be soon launching on the Internet.

In the future it is anticipated that Digital Radio reception and 3G capability will be combined into Personal Digital Assistants (PDAs). Manx Telecom is also intending to launch the world's first third-generation mobile service in Spring 2001, ahead of the rest of the UK and other international plans.

Net: www.virginradio.co.uk

Japan: As part of its refurbishment, Tokyo TV has installed a AMS Neve Digital Film Console in its Studio 407. The 2-operator desk is lined up for a wide variety of television programmes and feature film work including IMAX, foreign films and multichannel work for special events. Executive VP and senior engineer Shuji Inoue commented: 'The DFC enables our engineers to concentrate on creative sound production and gives us a very efficient studio environment'.

THE MOTION PICTURE

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SONOLUX STUDIOS

Mixing drinks and music, Colombia's Sonolux Studios serves local needs with an eye on the international picture. **George Shilling** drops in for a session

SONOLUX HAS EXISTED in its present location in Bogotá, Colombia since September 1995. The company started as a record label in Medellin, with two MCI-equipped studios, before moving to Bogotá about ten years ago. The majority of the work in the studio is for the Sonolux label, but studio manager Jorge Diaz strives to fill the rest of the time with work from major record labels such as EMI, Sony and Warner Music. The studio building is within a complex that also houses RCN television. The owner of the whole outfit is the charismatic businessman Carlos Ardilla Lulle, who also owns major beer and soft drinks companies. This is not to be sniffed at, as Colombians drink more soft drinks per capita than any other country in the world. In the current Colombian political situation, he wisely travels with 60 personal bodyguards.

Entering and leaving the complex involves negotiating tightly organised security arrangements—although it is entirely possible that they only searched me to make sure I didn't have any rival soft drinks manufacturer's products with me. An in-house soft drinks machine is available at all times without charge to clients, although, sadly, there is no beer dispenser.

The two studios were constructed inside a former warehouse, under an overhead walkway, which slightly limited the available height, although the ceilings are by no means low. The design was by West Coast consultant George Augspurger of Acoustic Integrated Design, with acoustic construction by Horacio Malviccino.

Most of the music recorded at Sonolux is either 'Ballenatos' from Columbia, or Mexican 'Rancheras', which are both percussion-led rhythmic South American styles. According to Diaz, strings are recorded with increasing frequency, and there are a number of sessions recording and mixing what he describes as ballads and romantic music: 'Very soft music, drums, bass, guitar, piano and vocals and that's it. It doesn't get too crowded. Salsa used to be popular, and Merengué, but that is not a Colombian speciality.'

There are a couple of vintage items from the old Medellin studio, but mostly the equipment dates from 1995. Diaz explains: 'In Colombia you have to buy, thinking about what will be good for ten years, because we cannot afford to change consoles every four years.' For Estudio A the choice was an SSL 4048 G+ with Total Recall with a Sony 3348 multitrack. Estudio B houses a Sony MXP 3036 console with JL Cooper automation, with a Sony APR24 2-inch machine. Each studio has its own recording area but the larger Estudio A room is also wired to the Estudio B control room, and both studios have a window to this area. There are also 24 tie-lines between the control rooms, enabling transfers between the two multitrack formats and sharing of outboard. There is a comprehensive selection of toys, including Lexicon 480L, 300, PCM70 and multiple PCM80s. There are also Teletronix, Urei and dbx com-

Left: Jorge Diaz in Studio A. Below: Carlos Ardilla Lulle.

pressors, Tube Tech and GML EQs, and a good selection of mics including several valve models. Estudio A also houses a Kawai baby grand piano.

'Because this is almost a third-world country, we don't have the budget to re-equip very often, so we have to buy good equipment that we know is going to last a long time,' Diaz explains. 'I would like to buy some more compressors and multi-FX—after five years, if you use the same stuff, the sound is very similar. Also more microphones and mic-preamps. But the SSL sounds great. When you use the Sony, you notice the difference. I like to experiment. But I can only buy what is necessary. We have a small Pro Tools, I call it *ProToolito*—eight tracks, and two-in, two-out. We use it for commercials; we have a client using it from 6am–10am every day. We also have a selection of MIDI equipment, such as E-mu, Akai-Linn, but we are not really a MIDI studio.'

Diaz explains the staff situation: 'Usually both studios work from 9am–6pm, and 6pm–12 or 1am, so we need four engineers, but sometimes there is a lockout session that is up to 15 hours, so I try to share that between two engineers—to do the whole day for a few weeks, one engineer would be dead by the end of the session.'

Diaz himself is one of the engineers, as well as being studio manager. He went to college in Miami and Ohio and studied music and sound engineering, before working in Studios in Miami and Colombia, including the John Storyk designed Audio Vision, also in Bogotá. There are three further engineers on the staff, Rafael Henriquez and Marco Silva, and enthusiastic assistant engineer Raoul Ortiz. There is also a full- time maintenance engineer, Jorge Caballero. Diaz explains the need for this... 'We charge by the hour normally, and the rate is far less than a comparable US studio, but there are no hire companies here. I think we are the only studio in Bogotá that has a [Lexicon] 480. So if it's broken you have to use something else. But Jorge Caballero is very good at fixing things.'

I was at Sonolux for two weeks to finish off recording and mixing an album for EMI in Colombia by a new artist, Cabas, and enjoyed the experience enormously. The in-house engineers are enthusiastic, helpful and knowledgeable, and sensitive to clients' needs. I can't wait to return...

Diaz: 'The music business is different in Colombia. Engineers don't get paid very well. But I like it, I always wanted to get back to Colombia and work with my people. But the situation in Colombia is tough. We plan to keep working hard and doing our job. It is difficult to persuade international artistes to come here. If they happened to be here doing a concert, I think this would be the studio they would choose. But we don't have pizza delivery in the middle of the night, like in the States. To me, every client is important. We have had Greg Ladanyi here, Tony Harris from England and John Fausti from New York, and they loved it.

Contact:

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Tel: + 571 41 40077.

Tascam DA-98HR

Combining high-resolution recording and unexpected flexibility with improved operation, Tascam's new MDM is one to watch writes **Dave Foister**

TASCAM'S BID TO LIFT the MDM format out of the project studio league began some time back. Following Alesis' move to 20 bits with the ADAT XT, Tascam responded with the 24-bit DA-78HR, managing to fit eight full 24-bit tracks on to a standard DTRS tape running at normal speed. This removed the need for the various bit-splitting devices that had previously been available to spread longer words across the tracks, reducing the number of available tracks in the process, but did nothing to address the issue of higher sampling rates. Indeed most attempts to increase the resolution of digital audio on existing media have only been able to improve in one area—Pioneer fitted 96kHz on to DAT while Tascam got 24 bits on, but nobody has done both.

Now Tascam has produced a machine that addresses both parameters in a familiar format. The DA-98 is already familiar as the grown-up replacement for the venerable DA-88, adding built-in time code and synchronisation functions along with a new user-interface and menu structure, with many improvements in functionality and operational ease. Now it is joined by the HR version, that not only offers eight tracks of 24-bit recording at standard sample rates, but has the facility to double or even quadruple the sample rates, with reduced numbers of tracks. It even has the facility to have some tracks at standard rates and some at double, opening up a very wide range of possible applications.

The machine itself is essentially familiar in layout, but the sophistication of its front panel is a far cry from the relative simplicity of the original 88. Such is the increase in features since the early days of DTRS that getting at everything through menus all the time is simply too cumbersome, and many more dedicated front panel buttons are now fitted to allow direct access to more functions. Having said that, the menu structure itself continues to grow in complexity, and the display screen and layout of the menu-driven functions have improved immensely, making it far easier to get at the nuts and bolts of the system without recourse to the manual. Note that the differences between this machine and the existing DA-98 are too extensive to allow a simple relabelling; the front panel differs in many important respects to allow access to the new functions and proper display of what's going on. In fact the appearance has moved further away from the 'semi-pro' look of the early machines and is becoming ever more functional and professional.

Round the back the changes reflect the applications in which the machine is likely to get used. For the first time, analogue inputs and outputs are an optional extra, filling two long horizontal slots in the rear panel. Gone are the domestic phonos (although some used to prefer using them as there was less electronics in the way) and the cards have the traditional 25-pin D-type connectors for balanced in and out. The same connector type carries Tascam's TDIF digital I-O, which presumably is intended to be the default connection, and there is now a further D handling AES-EBU directly, with a full four stereo inputs and outputs. One of the many new menu functions determines whether these work in 1-wire or 2-wire mode for high sample-rate audio, offering unusual flexibility.

There's a host of other more conventional connectors, and it's interesting that the only XLRs on this professional machine are for the time code in and out. Besides these are BNCs for video and wordclock sync, MIDI connections, a 9-pin control interface, and Ds for Tascam remotes, sync to and from other DAs and external metering.

The front is clearly related to the DA-98, but has several ergonomic improvements. The area around the time display is bigger, in order to accommodate extra LEDs to show the sample rate(s) in use. Below it is the same basic set of dedicated function keys, only bigger and more clearly labelled. They all have a second shifted function and can double up as numeric keys for direct data entry of things like locator times. While these cover the most commonly needed features like input patching, location, autopunch, and so on, all are duplicated in the extensive menu system that lies alongside and covers everything the machine is capable of doing.

The screen for the menu system is now 20 characters by four lines, and divides the system into 15 basic menu screens. The top layer allows the screens to be selected with a cursor, and as the cursor moves around, the bottom line shows a few heavily abbreviated ideas of what may be found within. The ENTER key selects a menu, whereupon better labels allow functions to be adjusted either directly or by entering a further layer below. That's about as deep as it goes; everything's reasonably accessible and it's also refreshingly intuitive. I've always found the Tascam DA menus a little frustrating as the routes around them can be difficult to remember without the manual; perhaps that has happened simply as a result of the added features outstripping an old system's ability to handle them, and the 98HR is a complete rethink with all the benefits that brings. Of course everybody will have different ideas as to which functions should have their own dedicated keys; my own gripe here is that the continuous peak hold on the meters can only be reset by going through the menus, deselecting continuous hold and selecting it again—this is a function I tend to use a lot and here it's cumbersome. But it's a small gripe about what is essentially a major step forward in handling the power available in these machines.

The main news though on the DA-98HR is the addition of the higher sampling rates. Typically, a wide range of options is available to suit different applications. Plain vanilla 16-bit 44.1/48kHz across the eight tracks is of course available to maintain compatibility with all the existing DTRS machines out there, and Tascam refers to these sample rates as the base frequencies. The requirement to format tapes before use remains, and it is at this point that the configuration of tracks and sample rates is chosen. The FORMAT key selects between 44.1 and 48 as usual, and then there are two other decisions to make. First is whether to engage High Resolution (HR) mode, which is on a dedicated key next to the FORMAT key; this by default gives eight 24-bit tracks at base frequency, compatible with the DA-78HR. Once HR mode is on, a menu pops up to allow the track layout to be altered. The obvious selections are

- Selectable 16-bit or 24-bit high resolution audio
- High Resolution 48kHz mode (8 tracks), 96kHz mode (4 tracks) and192kHz mode (2 tracks)
- Confidence monitoring for layback, mastering, and live recording applications
- TDIF and AES-EBU digital I-O
- >104dB Dynamic range
- 20Hz–20kHz frequency response ±.5dB
- 1 hr. 48 min. recording time on a single 120 tape
- Built-in synchronizer for SMPTE, MIDI and Sony 9-Pin sync
- Individual track input monitor select switch facilitates easier checking of source-tape levels
- Selectable reference levels for integration into a variety of recording environments, with internal tone generator
- Track slip from -200 to +7200 samples
- Expandable up to 128 tracks (16 machines)
- Word Sync In-Out-Thru
- Serial Control (9-pin) port and parallel control port
- Optional 96kHz analogue I-O on balanced DB25

four tracks at 2x base frequency and two tracks at 4x base, but it doesn't end there. Two more options allow combinations of rates at the same time; one gives two tracks at double rate and four at base, while another gives three tracks at double and two at base. This ability to compromise and mix-and-match has obvious potential; a 5-channel stem with 24-bit 48kHz on the surrounds and 24-bit 96kHz on the fronts is quite appealing. Note that none of these options alters the running speed, meaning that all configurations retain the full tape capacity of almost two hours.

Once the tape is formatted, the recording process is exactly the same as on previous machines. In fact the same facility to record and format simultaneously is available, with the usual proviso that this can only be used for continuous recording with no stops and starts, such as a live concert recording. Multicolour LEDs show at all times how many tracks are available and what rates they are running at, and the sample rate LEDs beside the time display show what rates are in use. When a prerecorded tape is put in the machine, it immediately detects what configuration it carries and shows the same information on the LEDs.

Tascam's developments here partly reflect the fact that DTRS machines have been used with the previously mentioned bit-splitting boxes for classical recording for some time, where the requirements of the job mean that resolution is more important than numbers of tracks. In fact straight 2-channel stereo is the norm for much of this kind of work, and the 98HR can now handle this, with yet higher resolution, without external devices. The exception to this is that the machine does not carry its own 192kHz convertors, which will not be a problem if the default digital inputs are used with appropriate outboard convertors. If the analogue boards are fitted, the convertors do a very impressive job of the double rates, and there is one further feature that makes it easy to check on just how good they are.

This is another function clearly provided with the classical recording industry in mind, as when it's in use it becomes impossible to do conventional multitrack recording with overdubs and drop-ins. It is a Confidence mode, providing—almost incredibly—off-tape monitoring in all recording configurations. The off-tape signal is delayed by around a quarter of a second, and the available tracks are armed and made safe all as a block with no individual control, rendering studio multitrack work impossible. It does however offer the all-in-one-take engineer the opportunity to check what's going on, and it gave me the chance to compare the performance at the different sampling rates simultaneously.

On a straight-to-stereo session I was able to set the machine up with three tracks of double rate, two of base, all at 24 bits and fed from the analogue inputs, and also run a four-head DAT at standard 44.1/16-bit. The results could all be compared on the fly, and the differences emerged straight away. The increase in detail and low-level clarity going from 16 to 24 bits was as expected, and the improvement going to the high sampled tracks was of a similar order of magnitude. There's something about pushing the limits back like this that makes the resulting sound somehow more solid and robust, with a reassuring completeness that our old standards lacked. I always feel that the impact of the lows and mids benefits as much from the increased sampling rate as the HF clarity, and this was certainly the case here—bear in mind still that this was using the onboard convertors. With analogue it was always the case that as increased tape speed opened up the HF ceiling it brought the penalty of lumpy LF response —30ips is notorious for it—but there is no such trade-off in digital. The performance of the DA-98HR as a complete stand-alone analogue-in machine will stand comparison, I reckon, with anything.

It looks as though Tascam has got an awful lot of things right on this machine. The format already has a strong presence in a surprising range of environments, and this will reinforce its appeal in every one of them. 24-bit base rate is still the best tape MDM configuration available, and the combinations of higher rates will have a host of applications. It also looks as though the classical fraternity will at last have an off-the-shelf affordable solution. Anybody for 96kHz 24-bit Ambisonics? Here, at last, is the machine for it, together with state-of-the-art stereo and real high resolution sound for pictures. Get one—if you can. □

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Tascam, UK. Tel: +44 1923 819 630.

Marantz PMD680

Marantz' latest portable PC card recorder makes good company for its earlier MiniDisc machine. **Neil Hillman** explores its similarities and developments

AS THE NEW ACADEMIC YEAR STARTS and children return to their studies, younger siblings of the privileged few being schooled at our great academic institutions will hear their names resonating around historical school halls. The imposing spectacle of mortar-boarded, black-caped school masters will call morning register: 'Aachen Major, Aachen Minor; Botticelli Major, Botticelli Minor; Conquestador-Smythe Dunstable Major, Conquestador-Smythe Dunstable Minor'. And so it goes, as these young charges are introduced to the less forgiving world of big school.

One of the new names being called this Autumn is that of Marantz Minor. Marantz Major—also known at the big school as the PMD650 portable MiniDisk recorder, Ginger to its chums—has been around for a year or more now. Marantz Minor, the PMD680 portable PC card recorder—Squirrel to its chums—is the new boy on the dorm. The PMD680 heralds a new generation of portable audio recorder for Marantz, as it enters the crowded market of solid-state recorders. Marantz acknowledges that the excellent PMD650 MiniDisk recorder almost did not appear due to the development and imminent release of this solid-state device.

The 680 is designed firmly as a news journalist's tool. In part this is due to Marantz' curious decision to offer a single input, mono device—it acts purely as a recorder, while its nearest-price competition (Sonifex' Courier and Maycom's Easycorder) will edit material with varying degrees of finesse, and both of them in stereo. The assumption is that price was the determining factor, as the Marantz remains just below the psychological £1,000 (UK) barrier; but how soon will it be before the stereo version is released?

PMD680 recordings are stored directly onto a removable PCMCIA PC card that can be inserted directly into a suitably equipped laptop for editing, mailing or further processing. The recorder accepts both Type II and III PC cards giving access to the largest 400Mb cards available on the market—a creditable one hours worth of PCM recording. The machine will record in two formats, either compressed using MPEG1 Layer 2 (MP2) with selectable bit rates for MPEG recording at 32kbps–192kbps @48kHz or uncompressed using 16-bit, 48kHz Pulse Code Modulation (PCM) broadcast WAV (BWF) and .WAV file formats. Special file handling by the PMD680 also avoids the risk of losing all the disk contents when a power failure occurs during operation.

The top face of the slim, black machine is the main control surface housing 22 individual switches, yet maintaining the neat, uncluttered and logical layout of its brother PMD650. Both the MiniDisk PMD650 and the PMD680 PC Card recorder are wonderfully understated in styling, inspiring confidence from first touch. The bottom-right quarter of the machine is taken by the in-built 300mW speaker grille. Just above it is the inbuilt microphone. The remaining left half of the top face houses the slightly recessed bank of switches and buttons given over to the task of controlling recording and input parameters and dynamics. Three rows of three slider switches are dedicated to the functions of SPEAKER ON-OFF; microphone input selection: between Mic 1 (the XLR input) or Mic 2/Internal (the ¼-inch jack input), or in the absence of an inserted plug the internal microphone; and Record mode—switchable between short, medium and long play settings and denoted. The middle row of three switches govern REC LEVEL with the options to record using the ALC, with a limiter in circuit or in Manual mode; input source—switchable between the Line input or the two mic configurations, with the third position 'Ext' being redundant on the test model although this would normally access the telephone line input to the machine. The last switch of the middle row, MIC ATTEN, is switchable between attenuation presets of 0dB, -15dB or -30dB. The bottom row of switches cover the functions of a KEYLOCK slider; PRE REC OFF or ON—allowing for 2s of pre-hear before the RECORD button is pressed; and ANC—Ambient Noise Control. This switches between a 150Hz bass cut, flat or band-pass filters with roll-over frequencies of 150Hz and 3kHz. Each setting is denoted by a line diagram of its characteristic.

Below these switches is a row of five push buttons associated with record management: REPEAT (used in conjunction with the play key to repeat either a single track or the whole disk), RENUMBER (used in Stop mode to renumber recorded tracks), ERASE (used in conjunction with the forward and reverse transport keys to select a track to be erased), SILENT SKIP (a means of maintaining the machine in record-pause until a preset level is seen on the input, at which time the machine restarts and continues to record until the input drops below the threshold). This threshold is set in the pre-set menu and operation of this function is shown by an icon on the display panel. The last of the five buttons is AUTOMARK—when selected, an EDL mark is placed at the point at which the machine drops out of Record and enters Pause mode. An EDL mark is automatically placed at the beginning of each track regardless of whether this function is selected.

The right-hand side of the top panel contains the normal transport five switches of PLAY-PAUSE; STOP; FORWARD-FAST FORWARD; REVERSE-FAST REVERSE and RECORD. These last three switches also double as keys for the EDL-Mark functions with the FORWARD and REVERSE keys providing a Jump-to-Mark facility during EDL Play mode, while the record key—labelled REC-MARK—inserts an EDL mark during recording every time it is subsequently pressed. Two small circular push buttons above this are used in the 'mark' process. The first is TOTAL MARK, used to display the total number of marks used with the limit being 255, the second labelled MARK SELECT, allows the selected EDL mark to be one of four types. These are a Play

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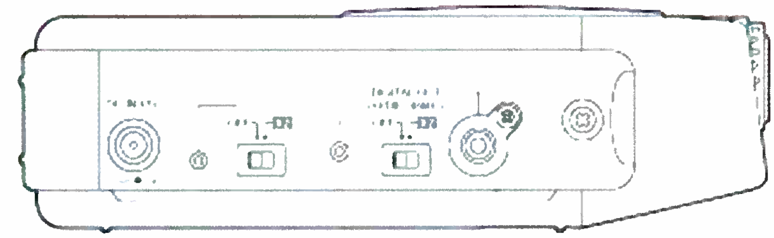

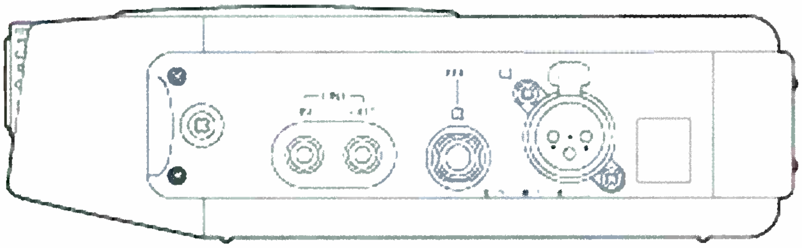

Mark 'P' (the audio after the Mark is to be played until another mark is reached), a Skip Mark 'S' (during EDL playback the audio after this Mark should be skipped until the next Mark is reached), A-point Mark 'A@ (the start point for a repeating loop-play) or B-point Mark 'B' (the end point for the loop-play). All new EDL marks are initially placed as Play marks, but the definition of all of the Marks may be changed subsequently.

The right-hand side panel houses the microphone inputs on a female 3-pin XLR connector for Mic 1 and on a ¼-inch socket for Mic 2, with Line In and Line Out connections appearing on twin RCA sockets. There is disappointingly no microphone powering available. The input sensitivities for the microphone inputs are -68dBu at an impedance of 9kΩ and -20dBu at 47kΩ for the line input.

The left-hand side panel is home to the 3V DC input socket, a slider switch to switch in the charger circuit for the internal battery, a sub-miniature remote jack socket, the SPDIF-48kHz Digital Out RCA socket and an ON-OFF switch for the digital output.

The front face of the PMD650 carries a large display window that shows track number, whether SP or LP recording is selected, battery-level as an icon either full, half-full or empty and a single bar-graph level meter calibrated between infinity and 0dB, with steps marked at -40dB, -20dB, -12dB, -6dB and -2dB. An Over level is set to the right of 0dB. The metering is disappointing, and could be bigger scale than it is, with the majority of available space being given over to track information. A small red LED next to the metering panel illuminates to indicate Record mode. To the far left of the front panel is the 32Ω, ¼-inch headphone socket that disables the speaker, and above this is the volume pot for both the headphones and speaker; the speaker is also defeated when the machine is in Record.

The card slot is to the left of the front panel, with the eject mechanism centrally located. There is no cover for the card slot when a card is inserted, relying on the weatherproofing of the optional carry case; a flap prevents the ingress of dust and dirt when a card is not in the slot. There are two push buttons above EJECT, the first marked LIGHT that either momentarily back-lights the display screen, or if held for a couple of seconds, remains lit. The second button alongside the light is marked DISPLAY and toggles through a loop of track number and total time, track number and total remaining time, a time of day clock and the current date. At the right-hand end of the front panel are the dual-concentric Record Level pots, graduated 0-10 and, as with the PMD 650 MiniDisk recorder in my opinion in need of either a heavy friction pad or a locking device to prevent movement inside a mixer bag taking the record level either up or down at random.

Strong and handsome like its older brother, and equally easy to use, the family likeness between the 650 and the 680 is striking. But as is so often the case with younger brothers, its promise appears to be potentially greater than that of its elder; a little more time should allow its character to be developed, allowing it to mature into a fully rounded product that will soon outstrip its older sibling. And before you ask —no, I am an only child. □

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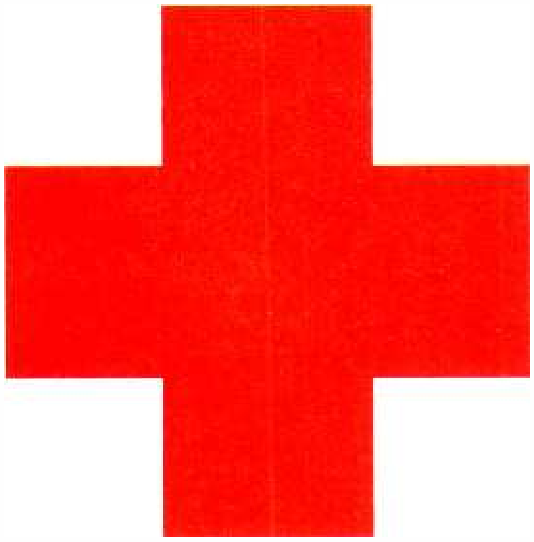

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Amek Driver in a Box

Boldly bringing one of the least glamourous aspects of console design into the open, Rupert Neve's Driver signals better audio. **Dave Foister** gets in tune

IT SEEMS THE NAME OF MR RUPERT NEVE can sell anything these days. On paper the Amek Driver in a Box (DIB) could possibly qualify as the dullest piece of equipment ever offered for our attention; DI boxes are more interesting than this. Yet in many applications what it does is highly desirable, and the fact that what it does is an area of special interest to the great design maestro means that it should do it remarkably well.

What we have here is essentially eight line drivers in a single 1U-low box. The manual gives an excellent account, apparently written by Rupert himself, of what long cables can do to audio signals, balanced or otherwise; the DIB's purpose in life is to minimise these effects. But because of the properties it acquires as a result, it can turn its hand to several other jobs as well, making it more exciting than it might at first appear. Well a bit.

The primary requirement for an output stage capable of delivering a signal down a long length of cable is a low-source impedance. This minimises the effect of the impedances inherent in the length of cable itself, which, as they are complex impedances comprising inductance and capacitance as well as the simple resistance of the wire, will manifest itself as a steady rolling off of the upper frequencies as the length increases. Detailed diagrams, graphs and figures are given to support this, with the sobering thought that 100m of typical installation cable driven by an output impedance of 600Ω will be 5dB down at 21kHz by the time it reaches the other end, while dropping the output impedance to 100Ω reduces the loss to 1dB. This is the output impedance of the DIB.

Also crucial to the design requirements for Neve is the use of transformers for balancing both inputs and outputs, for their unbeatable common mode rejection ratio and complete electrical isolation of circuit components. Any disadvantages of transformers reduce greatly as size and quality increase, and Neve specifies transformers that are the equal in performance of any electronic circuit. The consequence of this is that although the small number of controls means the DIB

NEW TECHNOLOGIES

New Tascam digital

Tascam's DM-24 digital mixer (US$2,999) is a small format desk with flexible routeing, built-in automation, and built-in effects, dynamics processing and parametric EQ. It is 24-96 capable and has full transport control, LED ring encoders, hierarchical grouping and 100mm touch

sensitive motor faders which work in conjunction with on board automation that requires no external computer or software. Up to eight mixes can be stored and automation data can be offloaded via MIDI.

In addition to 16 analogue inputs with XLR mic, ¼-inch line inputs, and analogue inserts on each input channel, the console includes 24 channels of TDIF and eight channels of ADAT optical, one stereo AES-EBU and two SPDIFs. Two option slots are provided for additional 8-channel analog, digital and cascade interface modules. Tascam has teamed with tc electronic and Antares to offer reverb, mic modelling and speaker modelling for the desk. The DM-24 also provides eight

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auxiliary channels for effects returns and each channel has a delay with a design that compensates for fixed latencies within a digital studio.

MX-OS Version 2.0 for the MX2424, scheduled for release later this year, allows recording and playback at 24-96 for 12 tracks of 24-bit and 24 tracks at 24-bit, 48kHz. Audio file transfer via the Ethernet port will be possible along with HFS+ Mac drive format compatibility. Tascam is to develop a version of the DA-98HR that supports Sony DSD format. The DSD Recorder will be configurable as a DSD 2-track with Sony's SDIF-3 digital I-O and as a PCM multitrack. Recordings made on the DSD Recorder will be backward compatible with existing DTRS recordings.
Tascam, US. Tel: +1 323 726 0303.

Genex convertors

HHB has launched two affordable 8-channel convertors from Genex Research. Drawing on technology from the GX8500 multiformat recorder, the GXA8 and GXD8 are

the first commercially available devices to support 8-channel, 24-bit/192kHz and DSD conversion.
The GXA8 A-D translates eight channels of balanced analogue into 24-bit digital with outputs available in AES3 fitted as standard, supporting four channels of

fits eight drivers in 1U of panel space, the depth and weight of the box come as something of a surprise when you try to get it out of the carton.

The other downside of the slim chassis is that there is not room on the back for XLRs for inputs and outputs, never mind for the insert points. The answer is to put all the connections on 25-pin D-connectors, wired to the same convention as Tascam multitracks, which is fine once you've bought one and made up the bits, or for installation, but not so great for quick setup. However it does leave room for push buttons for various functions: enabling the insert points in pairs, and linking outputs to form various combinations of distribution amplifier. The insert points are unbalanced and all eight appear on a third 25-pin connector. Suggested uses for these include adding external faders to help use the DIB as a set of virtual earth mix amps, and splitting each signal path into two amplifier blocks. A hidden bonus is that use of multiple channels as a DA leaves input stages unused, and these can be accessed via the insert sends to give even more signal paths.

The front panel makes a valiant attempt to be interesting, with illuminated badges on the rack ears saying 'Amek' and 'Pure Path' (the range of equipment the DIB forms part of, along with the Channel in a Box and the Stem Compressor). Each channel has a level meter, a centre-detented gain trim pot and three illuminated buttons for MUTE, PHASE INVERT (potentially very useful in the kind of setups the unit is likely to find itself in) and SILK. This last introduces the characteristic Neve sonic quality, and undoubtedly adds a subtle smoothness to the overall sound.

Inside are jumpers and further gain trims to establish custom gain structures through the unit, to help with the wide range of applications Amek suggests. These include using a single channel to operate as balanced inputs and outputs for an unbalanced digital mixer via the DIB's insert points, and deriving clean-feeds using the mix amplifier mode. The manual is very helpful in describing exactly how to go about incorporating the DIB in these various setups.

My attempts to determine the benefits of the DIB over long lengths of cable only served to show that my building's wiring is not as bad as I thought. I ran signals round as long a length of tie-lines—cheap, single-overall-screen multicore—as I could find, one version direct from my console output and one via the Amek. Measurement showed there to be no difference at 20kHz; neither was attenuating compared with the source signal. Consequently this defeated any attempt to identify a sonic benefit, so clearly the cable was too short for the detrimental effects to manifest themselves—either that or my console has a similarly low output impedance. On the other hand, previous experience of 25-year old Neve distribution amplifiers (still in use in my facility) suggests that these circuits will do the job, and the SILK button is a worthwhile bonus.

The Driver in a Box is not something every studio will want or need to rush out and buy, but there will be many in the live sound arena, the location recording industry, outside broadcast and elsewhere who will have recognised that it is just the tool they need to solve old familiar problems. □

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AKG C 4500B-BC

AKG's new mic promises to offer condenser performance in the presenter's environment without the drawbacks **Dave Foister** reports

LOOK AT OLD PICTURES of DJs in their radio studios and the chances are they'll be captivating their audiences via an AKG D202. This once-ubiquitous dynamic microphone was one of the most successful all-round dynamics ever, although in more recent times the field has opened up considerably. Even so, dynamics are still popular as close-up presenters' microphones, despite the potential difficulties caused by the close proximity of multiple computers and monitors in the modern on-air suite.

Now AKG is back, this time with a condenser intended to defeat the computers' fields, yet retaining some of the advantages of the traditional dynamic. The C4500B-BC is the latest in a cosmetically similar range of affordable condensers that also includes the C4000 and C2000, and almost certainly shares some components. The main difference is that the 4500 operates in end-fire orientation, allowing a couple of extra features to suit it to its chosen role.

The housing of the previous CX000 microphones could almost have been designed with this in mind, as its cylindrical grille includes a circular top mesh where other grilles have the frame components. On the 4500 this becomes the front, and the length of the cylinder itself helps with fitting the microphone to its function.

The two most difficult things to deal with in the typical presenter's situation are the threat of the popping plosive and the control of the inevitable proximity effect. The orientation of the housing in the 4500 allows both of these problems to be addressed internally. First, there is a new multilayer pop filter within the grille, with a design AKG has applied for a patent for. Second, the physical size means that the mouth can never be closer than 2–3 inches from the diaphragm, reducing the worst of the proximity effect and leaving the desirable presence. A switchable 120Hz filter deals with anything that's left, along with structure-borne noise and rumble. Finally, a foam windshield, supplied as standard, increases the protection for particularly difficult voices.

The stand mount is also familiar from the rest of the family. Proper elastic suspension cords hang the microphone securely within a plastic ring, while a locking ring grips the microphone body. The supporting ring carries the stand mounting swivel, secured with a locking bolt. This arrangement is more than capable of holding the weight of the 4500, which is not particularly heavy despite having a properly screened metal housing. The end-fire configuration also means that this fairly elaborate mount can be used in front of the presenter while still offering a relatively small frontal area to avoid clutter.

The screening of the internal electronics is all part of the business of making the microphone immune to outside interfering influences, as is the use of a transformerless output stage. The acoustically open sections of the housing also have double screening to protect the capsule, and the result of all this is a self-noise level around 8dBA. Coupled with an overload point of 145dBA without the pad, this gives a very impressive dynamic range indeed. If you need to go further, the pad adds 20dB of attenuation; AKG suggests that this should reduce the output level to something like that of a dynamic microphone, allowing the 4500 to be substituted without radically changing the gain structure of the signal path.

Although the main suggested use is the on-air presenter, the instructions also reckon it is suitable for studio vocals and instruments. With this in mind, and because it might present even more of a test than speech alone, I used it on a singer with little recording experience working very close. Some presenters' microphones might have complained at this as their tonal flavour is very much coloured for the specialised role, but the graph for the 4500 suggested that this would not be so much the case here. The bottom end is shown as very flat, and only a couple of humps in the upper mid—neither with more than 4dB excursion from flat—deviate from the neutral. In fact it's flatter than many general-purpose condensers.

What little coloration there is turns out to be very flattering to the voice either singing or speaking. It adds considerable presence and impact when working up close, without the extremes of EQ that often seem to be required on the radio and can be added later if desired. The pop filtering turned out to be impressively effective, cutting out all the plosives from what was not a well-controlled voice. In addition, the suspension mount coped well with the singer's inability to stop handling the microphone stand; although I could see the thing swaying from side to side I could hear no consequential effects whatever.

The switches are not as accessible as they might be when the microphone is clutched in its mount, but the idea that the pad is mainly provided to match the level to that of a dynamic was borne out by its behaviour without attenuation even working very close. With all of this in mind, the C4500 is really much more than a presenter's microphone; it does that job very well, but has the flexibility to do a lot else besides. □

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Arbiter Group, UK. **Tel:** + 44 20 8202 11 99

NEW TECHNOLOGIES

24-bit/192kHz in dual-wire mode or eight channels of 24-bit/96kHz in single-wire mode. An optional AES expansion card provides eight channels of 24-bit /192kHz in dual-wire mode, with cards also available to support eight channels of SDIF2, ADAT and TDIF output.

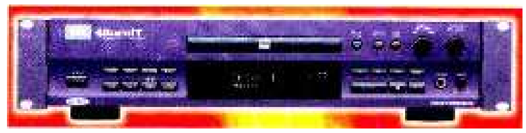
As an option, the GXA8 can operate in DSD mode. The GXD8 D-A operates in PCM formats up to 24-bit /192kHz and, with an optional card, in DSD mode. Digital audio can be input in a variety of formats with AES3 available as standard. Optional cards are available for AES expansion, SDIF2, ADAT, TDIF and, in the near future, IEEE1394. The high-stability crystal oscillator clock used for the conversion process is buffered and brought out to the rear panel allowing the digital source device to lock to the GXD8's D–A convertors.
HHB, UK. Tel: +44 208 962 5000.

Digidesign does POW-r

Digidesign has licensed the POW-r digital audio word length reduction (WLR) for introduction on Pro Tools. POW-r (Psychoacoustically Optimised Wordlength Reduction) is a patent-applied-for algorithm, that reduces longer word lengths up to 32 bits to CD 16-bit format while retaining perceived dynamic efficiency and very low noise. POW-r is a scalable algorithm and ready for sample rates to 192kHz.
POW-r Consortium, US. Tel: +1 530 647 0751.

HHB's BurnIT

HHB's CDR830 BurnIT is a low-cost professional audio CD recorder with 24-bit A–D and 24-bit, multilevel Delta Sigma D–A that uses pro CD and CDRW media. Phono I-Os are complemented by coaxial and optical SPDIF I-Os, with an on-board sample rate convertor



accepting frequencies from 32-48kHz. Auto track increment combines with four synchro-recording modes and a 2x finalise feature means that the discs are 'fixed' in two minutes. CD Text support is included along with digital record gain and balance control, input monitoring with track increment rehearsal, fade in-out adjustable from 1-12 seconds, five CDRW erase modes, track skip ID recording, track index search on playback and program, random and repeat playback modes.
HHB, UK. Tel:+44 208 962 5000.

Sony CD-Rs and MDs

Sony has revealed two CD-R machines with the higher-end CDR-W66 available at the beginning of next year featuring DSP functions, parametric EQ, limiter and SBM and the CDR-W33 available immediately (US$799).. The CDR-W33 incorporates SRC, coaxial digital, optical

digital, analogue unbalanced jacks while the CDR-W66 adds Word Clock interface, AES-EBU I-O, balanced analogue I-O, RS232C and parallel (GPI) control ports, a 2X speed duplication link for dubbing using two

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Tascam CD-RW2000

Regrouping from previous assaults on the audio CD-R market, Tascam has combined elements of predecessors into a new machine. **Zenon Schoepe** believes they have created their own Best of...

TASCAM HAS BEEN BUSY releasing CD-R machines of late and the cynical out there might say that the moves have been made to improve the breed. Without doubt each generation of machine has indeed added features and operational ease and convenience which makes the newest CD-RW2000 the most advanced machine to come from the company.

It looks identical to a CD-RW700 in the presentation, casing and button count, although closer inspection reveals that some of the buttons have been relabelled to reflect additional functionality. However, basic operation is as per the CD-RW700 (*Studio Sound*, April 2000) so I'll concentrate on the differences.

Most important things first then, the timer has been removed and the Record mute function of the CD-RW700, which was alarmingly lacking on the original CD-RW5000, has been preserved.

The relatively meagre back-panel connections of the CD-RW700 have been boosted with some proper possibilities. Analogue I-Os are available on phonos and balanced XLRs the latter with useful grub screw output trimmers. Digital I-Os are presented on XLRs for AES-EBU with SPDIF in coax and optical flavours. Digital control of level is available on all inputs and the SRC is defeated along with this when direct digital cloning is being performed.

There's also a switchable Word sync input that would become relevant in larger digital setups and this commanderes the front-panel space occupied by the infrared sensor on the CD-RW700. On the CD-RW2000 this is replaced by a Word indicator, something that is made possible because the new machine comes with that rarest of things these days—a wired remote control. This is a pleasure to use, once I stopped trying to force the battery compartment open which is located beneath a label that states that no batteries are required as the remote draws its power from the machine. Freedom of movement is returned and I was reminded at just how inconvenient staying in line of sight with an infra-red remote can be. From memory, the functionality of the remote seems very similar to that delivered with the CD-RW700, but the reassurance of being able to press a button and know that the command has gone down the cable cannot be overemphasised. It really helps to take some of the edge off dealing with write-once media.

The back panel is completed by a 15-pin D-sub for pulling out control I-Os and enabling such things as fader start, for example.

In among the few front-panel switch changes is the replacement of the MONITOR button of the CD-RW700 with a FADE IN-FADE OUT button for firing the programmable ramp in and out of recording. Activation of the monitoring function, which allows you to use the box as a straight-through convertor when no disc is loaded, is performed from the remote. The switch allocated to this fade function on the CD-RW700 is now marked Call on the new machine. Pressing this in playback modes returns the transport to the last point at which play was instigated from pause. It's a minor trick, but it does underline this machine's suitability to applications where it will serve as recorder and playback machine.

Copy protection can be defeated, activated or activated for single generation use.

Highlights remain the rehearsal mode for pin-pointing the precise start of a recording, Auto Cueing of tracks to the beginning of the audio (set by threshold) where they are paused for a type of instant start, and a good selection of recording modes.

Menu functions are accessed and adjusted using the MENU button and a dial with push to make. It works in conjunction with a screen that is still clear and bright, despite a large number of characters that appear to signify status, and good peak hold meters.

It also sounds good and I found it easy to operate, although how much this has to do with my own familiarisation now with the Tascam Method—RECORD button presses increment track IDs manually, for example—and the confidence inspiring wired remote is difficult to quantify. However, I have to say that this really is the most complete CD-R from the company as it has the full complement of I-Os, word input, control I-O breakout on the back of a really decent CD-R-RW with the bells and whistles that are now expected.

What it doesn't have, which the new generation of CD-R machines from HHB, Marantz and Sony have, is the ability to create CD Text. This may or may not be an issue for you as I for one have managed to survive without text info on CDs and can remember feeling surprised when I was reminded that text was an option on the platter. There is also the issue of compatibility and it's worth bearing in mind that while you may now be able to enter CD Text on to your disc your standard studio CD players may not be able to interpret it.

On balance this is by far the best CD-R/RW that Tascam has made. You could regard it as a combination of the features of the CD-RW700, and some, with the back panel connections, and some, of the CD-RW5000. It's a very strong package and is priced right in to the thick of the accessible end of the market. Certainly worth investigating. □

Contact:

Tascam, UK. **Tel:** +44 1923 819 630

Tascam, US **Tel:** +1 213 726 0303

NEW TECHNOLOGIES

machines and DSP functions available on digital and analogue inputs.

Sony has introduced two 1U-high, rackmount, MD recorders. The MDS-E10 and MDS-E12 incorporate ATRAC type R algorithm and replace the 2U-high MDS-E58 and the MDS-E11. They have 10 Instant Start memories. SPDIF coaxial and optical digital I-O, phono I-O; long record-play using ATRAC3, and menu control of hot start, auto cue, auto pause, sound start pause, varispeed, next track reserve, and digital record level adjust. There's also RAM edit, A-B erase, and front panel PS/2 port.

Sony, US. Tel: +1 201 930 6342.

Surround reverb

Yamaha's SREV1 sampling reverberator is described as a high-end, multichannel, sampling effects system, that includes the SREV (24-bit/48kHz 3U-high rackmount mainframe), RC-SREV1 remote controller and

DB-SREV1 DSP Expansion Board. Shipping is scheduled for the beginning of next year.

A selection of editable reverb programs simulates environments ranging from room ambiences to stadiums, with each offering control of pre-EQ, post-EQ and Reverb parameters. The unit operates in 2-channel (up to 5.46s/channel), 4-channel (up to 2.73s/channel) or 2-channel x 2 (up to 2.73s/channel for each processor) modes. With DSP expansion board, reverb time in each mode is doubled.

SREV1 uses impulse response samples of actual acoustic environments and by using a Time Stretch Pulse to measure impulse response via bundled PC software, one, two or four channels may be measured to capture and create custom reverb.

The box has two mini YGDAI card slots, two AES-EBU I-Os, two 'to host' serial ports and external word clock input. Options include a CD-ROM drive for data and upgrades plus a PCMCIA card slot.

Yamaha, US. Tel: +1 714 522 9011.

Crane Song's latest HEDD

HEDD 192, is a 24-bit stereo A–D, D–A convertor with DSP emulation of tube and tape sounds from Crane Song with adjustable triode, pentode and tape sounds. In addition to tape emulation, an analogue dither source has been added to the device. In the A–D mode the dithering options are 20 or 16 bits while in the digital I-O mode redithering to 16 or 20 bits can be accomplished by selecting the appropriate function. HEDD 192 will be upgradable to 192kHz when the components become available. Interface options include AES, SPDIF, Toslink and ADAT optical. HEDD 192 can be used as a word clock source or can sync to an external word clock. Release scheduled for early 2001.

Crane Song, US. Tel: +1 715 398 3627.

Tube pre

GT Electronics' Vipre is an all-valve mic preamp with

Used by thousands
Heard by millions

Soundcraft Broadcast Consoles

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Audio-Technica 815ST and 835ST

Packing Audio-Technica's latest shotguns, **Neil Hillman** takes to the streets to test reputations and form

IT IS NOT BEYOND the realms of possibility that 1962 may become a particular favourite among 'nostalgia' years. After all, this was the year that John Glenn became the first American to orbit the earth in his spacecraft Friendship 7. It was also the year that fellow American aeronaut Gary Powers returned to earth with a bump after being shot down in his U-2 spy-plane deep inside Russian airspace. A year earlier had seen the Berlin Wall built, and a year hence would see the release of the first Beatles album. All in all, this has proved to be rather fine timing for Audio-Technica—a company founded in 1962—as its opening product to a burgeoning market was a phonographic cartridge. Things have moved on, unsurprisingly, after nearly 40 years, and now A-T's product range is almost as long as the wait that was regularly endured by cuban-heeled Beatles fans queuing for an evening's entertainment in the company of

Helen Shapiro and the mop-topped Fab Four; those were indeed the days my friend, when bands played in variety shows and theatres.

Audio-Technica's product line still includes phono cartridges, but it also encompasses wireless systems, mixers and headphones, although just recently it is microphone products that have been drawing attention from discerning buyers who might previously have felt that A-T products were more 'budget' than top-end. And if you are one of those newly re-acquainted with Audio-Technica through the seriously sensual, despicably desirable adaptive-array AT895 microphone, there is strength in numbers; two new A-T stereo shotguns have just been launched that could well give the existing players a run for their money. But the marketplace is tough; sale volumes to this specialist sector of location recording are by definition not enormous, and the competition is well positioned in terms of either price or quality, and sometimes both.

The competition comes mainly in four guises: the Neumann RMS191—offering a smooth, full sound, but with a bulky control and matrix box to lug around, that enables adjustment of the pickup angle by varying the gain of the 'S' relative to the 'M' in six 3dB steps. The Sennheiser MKH30-50 piggy-backed coupling, with its faithful, clean sound, due in no small part to Sennheiser's use of a super-cardioid for the 'M' element and securing a greater than 6dB of rear rejection compared with other hyper-cardioid arrangements. The Sanken CMS-9—'baby's rattle' shape—that has a unique 'M' element design that while only cardioid offers a 6dB advantage over other cardioid designs, but which seems to bring with it an annoying degree of self-noise. And finally, my choice of compromise between budget and quality, the Pearl MS-8 with its rugged build, crisp sound and its ability to withstand rustle, rattle and general handling noise inside it's Rycote windshield.

The 2-pronged A-T attack is by models that operationally are identical, but use differing lengths of interference tube to achieve their objective. The AT835ST employs a 236mm long tube, giving a more super-cardioid pickup pattern, while the AT815ST sucks like a Hoover with its 380mm long tube providing a higher degree of hyper-cardioid receptiveness. Both mics are a slim 21mm in diameter, and weigh very little at 103g and 142g respectively. Speckle-grey, the bodies thoughtfully have a 'this way UP' label to ensure LR compliance, and house two miniature slider switches to allow bass roll-off selection (100Hz) and the choice between an M-S or an XY output, with the addition that the XY signal may be chosen to be 'wide' or 'narrow' stereo. There is a noticeable difference between these two settings and a quick look at the polar diagram's bears this out, with Narrow operating between 240° and 120° and Wide opening up to between 240° and 150°. Both the middle and side elements are back-electret condensers, giving a frequency response between 20Hz–20kHz, and show a considerable resilience to RF, something that can prove to be embarrassing with the otherwise capable Pearl MS-8. The microphones are phantom powered—11V–52V—and there is little self-noise, an irritation with the Sanken CMS-9 on low-level signals. The impedance is stated as 200Ω and the signals are output via a 5-pin XLR-M connector into a supplied 'Y' cord of two standard XLR-M connectors. The Mid and Side signal to noise ratio is given as 72dB and 68dB respectively; XY corresponding figures are stated as 70dB.

Priced at comfortably under £1,000 (UK) each, the AT835ST and AT815ST models are sent in to do battle at the budget end of the market; and in fairness they do lack that ultimate degree of sophistication sourced by either the Neumann RMS191 or the Sennheiser MKH30-50. They are, however, not so very far away. In the less than perfect surroundings of a life on the road, they present a truly capable and credible option for stereo location recording. □

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variable impedance and selectable rise-time. Altering the impedance load pushing against a mic alters tone and character while the switch-selectable rise time alters the speed at which the sound is amplified. The combination of circuitry and variable rise-time is said to provide intricate, and until now-unheard of, sonic capabilities. Fully floating differential inputs and outputs are switch-selectable and gain settings in 1dB steps from 15 to 75dB are complemented by audio metering. Alesis, US. Tel: +1 310 255 3400.

Genelec sub

Genelec's 1093A active subwoofer boasts a frequency response of 18Hz–85Hz ±3dB from a single 10-inch magnetically shielded driver in a 41ltr cabinet. Able to create SPL's of 112dB @ 1m, it claims a THD of less than 2% at 2nd and 3rd harmonics at 100dB @ 1m. Flexibility is afforded by the Loop Thru for daisy-chaining multiple units when higher SPLs are required.

Other features include a new 6.1 bass management

system within the cabinet, this provides XLR I-Os for combined LCR front and LCR rear configurations as well as a separate LFE input for Dolby EX and DTS ES. A built-in calibration tone generator allows phase to be set correctly using an SPL meter. Other features include an LF response control, and driver protection. Genelec, Finland. Tel: +358 17 813 311.

Digital Echelon

Otari has introduced the Echelon Series of digital audio products. The ND-20 is the core of the new series and offers A–D and D–A conversion as a stand-alone device or combines with multiple ND-20s to create a 96kHz digital audio network. Each ND-20 has 32-channel

capacity (16-channel with 96kHz sample-rate) and has four slots for I-0 cards. Options include mic inputs (remotely controlled), line inputs, line outputs, and AES I-O. All I-Os are capable of 96kHz, 48kHz, and 44.1kHz sample-rates, plus pull-up/pull-down rates. MADI, TDIF I-O and mLAN formats are available.
The Echelon FS-96 format and sample-rate convertor

Line6 POD Pro

The 'red bean' processor that blew the minds of guitarists the world over gains pro status with the addition of pro connectors. **Zenon Schoepe** decides it's all in the fingers

YOU EITHER NEVER WORK with guitarists or you're an unreceptive sort of an individual if the word POD means nothing to you. It was inevitable that a piece of high-tech machinery would eventually capture the heart of the generally technophobe guitarist in a big way and a small bean-shaped contraption from Line6 has been the one to do it.

Dedicated guitar preamps haven't made it in to the racks of all studios because most specifiers of the gear are not guitarists themselves. This is where I believe that the rackmount POD Pro will clean up.

The first thing you need to know is that POD uses modelling to simulate a variety of well-respected guitar amps, a variety of similarly respected cabinets and the interaction of both with a mic when recording them. Throw in the ability to mix and match these according to taste and it promises a lot and in each case you get a set of traditional guitar amp controls from the unit's front panel. We're talking about emulations of things like Marshalls, Fenders, Vox's, Mesa Boogies, and a few other more modern and esoteric models forced together with stacks and combo cabinets.

The second thing you need to know is that guitarists on the whole love the POD because it sounds great. Whether or not the models represent true approximations of the real things depends entirely on how familiar you are with them, but I have heard enough opinions from more knowledgeable guitarists to suggest that there is something meaningful in this. I think it's safe to say that the less acquainted you are with the originals in question the more amazing the POD will sound to you. It's an incredible feat of modern technology that POD sounds more like a real amp-speaker combination—forgetting which combination in particular—in terms of response, attack, timbre and playability than just about every other dedicated guitar recording preamp. It really feels like you're playing through an amp (and a good one) because of the way dynamics are preserved and how vibrant, live and unforgiving the experience is. When played, POD encourages you to forget its existence and get on with the job and if you want to change settings you can do so quickly via front panel dials for amp simulation, cabs and effects (thankfully restricted to effects in a guitar vein such as tremolo, chorus, flanger, rotary and delay). Reverb, tone and drive controls are provided where they should be on a string of pots. There are 36 memories available, but one of the smart things about POD is that you can upload your own presets in to a box via MIDI. There's an active sound-patch page on the company's website and extended editing of the POD's parameters is afforded through Emagic SoundDiver.

How it works precisely is probably not an essential for session work as the guitarist is likely to want to get involved with the box directly—put the box in Manual mode and you operate it just as you would an ordinary amp which is the guitarist's business anyway.

What you have is unbalanced line input and a unprocessed guitar output, unbalanced left and right outputs and balanced XLRs with switchable speaker simulation and a ground lift for the XLRs. There's also an AES-EBU XLR output and phono SPDIF (both track the speaker sim selection) and the unheard of inclusion on a box of this type of an external clock input (a front panel toggle selects 44.1, 48 or external sync). MIDI in and Out are accompanied by unbalanced stereo effects loop and a socket for a pedalboard that pertains only to live oriented functions.

The strength of the POD Pro in a professional environment aside from the back-panel sockets is its familiarity to incoming guitarists and the speed with which a result can be achieved. It goes without saying that if you don't deal with guitars that often then it's a good insurance measure for when you have to.

By it's very nature it's not a 3-noise sort of box, like so many of the pres around are, and it's capable of an enormous variety of extremely good workable sounds. If it has a down side then it's that it cannot muster super clinical cliched 'modern' guitar pre clean sounds no matter what you do to it. It always sounds like you're dealing with an amp—again forgetting which combination in particular—and it's very, very convincing. If you get solace and inspiration from dialling in a Marshall stack (basketweave cab, of course) then so much the better. An overlooked application is that of sending keyboards through the box (excellent for Hammond simulations) and while a dedicated Bass POD is now available you can get respectable results through the POD Pro. The line input also means you can reprocess recorded guitar tracks. It depends on the character of the recorded signal, but you can easily roughen up a clean guitar or send it to planet Jimi.

It may originally have been billed as a solution for the bedroom 4-track guitarist but POD has broadened its appeal with the Pro and a great idea has grown up. ☐

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NEW TECHNOLOGIES

makes multitrack transfers between different digital audio platforms fast and simple. It supports AES3, TDIF-1, ADAT optical and SDIF-2 with optional MADI and IEEE1394 for future networking capability. The FS-96 converts up to 24 channels of 24-bit at sample rates up to 24-bit/96kHz. A built-in digital router offers 10 preset routeing maps.

Based on the original Status, the Status 2 digitally controlled analogue console is designed to be an affordable 5.1 surround capable desk. The console is now 24-bus, has a 5-way panner on each input module, an M-Mon option available and an automated joystick panner for surround panning.
Otari, Germany. Tel: +49 2159 508 61.

Steinberg's Houston remote*
Steinberg's Houston remote controller for its Nuendo and Cubase VST production systems has nine 100mm touch-sensitive motor faders, eight rotary encoders with LED position indicators and a matrix of buttons. It also

has a large LCD display, transport controls and a jog and scrub wheel, a numeric keypad for entering values, and for selecting setups and marker positions.

Designed for hands-on mixing without having to use the mouse or PC keyboard it provides access to potentially every parameter available in VST windows. Houston allows the display and related rotary encoders to show the same parameter for all eight channels or eight related parameters for one channel. Plug-in parameters for VST audio effects and VST Instruments can be displayed, edited and automated.

TC Works has announced a surround-reverb for Nuendo. TC Surroundverb supplies 5.1 reverb and features a fast and easy to use interface, with graphic displays visualising the reverb components and a algorithm core.
Steinberg, Germany. Tel: +49 40 210 330.

Hafler's M5 compact*
Hafler has launched the M5 passive nearfield monitor which is scheduled to ship this spring for US$299. The magnetically-shielded M5 is a 5.3litre, 4th order Butterworth, vented, 2-way loudspeaker designed to

EB Promotion Mystere

Investigating a new Swiss ENG mixer, **Terry Nelson** puts on weight and narrowly misses a ducking. The Mystere comes through smiling

THE ELECT CLUB of ENG, or location, mixers has recently welcomed a new member—the Mystere from EB Promotion in Switzerland. The French 'mystere' means mystery the unit is very much to the contrary.

ENG mixers have to be simple and easy to use, rugged, provide high audio quality and yet have enough features to get the job done. The unit comes in a case that gives easy access to all controls and connectors and provides adequate protection in adverse weather. The unit is compact with the operating controls on the front panels, connectors on the side panels and supplementary controls on the top.

The front panel features on-off status LED; BATTERYCHECK push button; two 20-segment LED displays

for the left and right outputs and two LEDs indicating limiting. Underneath the meters are a headphone level control; 3-position reference tone switch; output-input monitor select switch, and 5-position rotary monitor switch (Stereo, Mono, M+S, Left or Right). There are four SQN-style rotary knobs for the four input channels with an adjacent overload LED and four smaller knobs for input gain. The panel is completed by left and right output master knobs and a switch for uncalibrated and calibrated operation. The calibrated position disables the level controls and corresponds to 0 level on the controls.

The top panel houses a centre section with four panpots together with associated 3-position LOW CUT switches and two LINK switches for operation with channels 1/2 and 3/4. In Link mode, the odd numbered channel level control becomes the master. A Master Out section to the right features a 3-position LIMITER switch and a MASTER LINK switch for linked operation of the main outputs (as per the input channels). The limiter is for in-out plus linked operation.

The question of where to put the connectors on a portable mixer has caused many designers sleepless nights. The Mystere carries inputs on the left and outputs on the right, with the exception of the Neutrik

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work alone or in conjunction with its TRM10s active subwoofer. The tweeter uses a proprietary 25mm silk dome and exponential horn waveguide combination and is the same found in the TRM6 active monitor. The 5.25-inch mid-bass driver is specific to this new monitor and uses a 32mm diameter voice coil for a 200-watt system power rating.
Hafler, US. Tel: +1 888 423 5371.

Evolution condenser

The latest addition to its evolution line of microphones, the e 865 is the first electret condenser in the Series.
The wired e 865 claims a 40Hz–20kHz response, a maximum SPL of 150dB and good pop suppression.
The mic is rugged with a metal housing and ships with a pouch and mic clip.
Sennheiser, Germany.
Tel: +49 5130 6000.

AT's condenser wireless

Audio-Technica has introduced the ATW-7373 multichannel UHF wireless hand-held condenser system which combines the condenser element from the AT4033 studio mic with the RF performance of the 7000 Series frequency-agile UHF Wireless System. The system includes the ATW-T73 hand-held condenser microphone-transmitter and the ATW-R73 true diversity receiver. The receiver is half-rack size with balanced and

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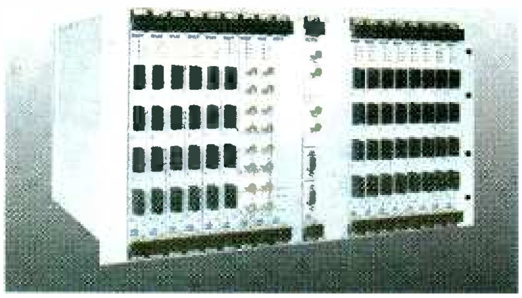
unbalanced audio output jacks, a ground-lift switch, adjustable volume and squelch controls and RF/AF/A-B front panel indicators.
AT, UK. Tel: +44 113 277 1441.

Star routeing
Stagetec's star connect routing element, the Nexus Star, has been in operation at SWR in Stuttgart since September. It can be fitted with up to 16 I-O cards, each with 256 inputs and 256 outputs, for routing of 4,096 inputs to 4,096 outputs. If more capacity is needed, the system can be cascaded. Nexus Star is a switching node that serves as a star hub for an entire Nexus decentralised audio routing system. Traditional Nexus base devices, each fitted with various I-O cards, are connected by fibre optic. Two different connection cards are supported: the Fibre Optic Connection card used to link Nexus Star to base devices and the RMF card that provides four MADI interfaces for connecting directly to Nexus Star and from other I-Os in the system.
Stagetec, Germany. Tel: +49 951 9722515.



locking headphone jack that appears with the inputs.

The input panel features a 4-pin XLR for external DC power supply, with associated toggle switch for external or internal powering, four XLR female connectors for the mic-line inputs with associated toggle switches and the headphone output jack. Each input also features two toggle switches for microphone characteristics—the first selects 12V or 48V phantom powering while the second toggles between Tonader (parallel) powering, phantom and dynamic (power off).

The output panel features two XLR male connectors for the left and right outputs and referenced to +6dB, two RCA (phono) connectors for secondary unbalanced outputs and switchable between -40dBu and 0dBu and a 9-pin D-connector that provides an auxiliary balanced stereo output and unbalanced input for an external source such as tape return. Twelve volt powering is also provided. The panel is completed with the housing for all NP1-type batteries and a flap that can be easily opened and closed.

The unit works with a voltage range of 11V–18V and when the battery is reaching a critical condition, the last segments for each channel flicker alternately. At this point, you will have about five minutes before the power drops below the critical level and the mixer 'dies'.

The meter display is bright and can be easily seen in sunshine. However, it is not overpowering in dim light. The LEDs to 0 are yellow, with six LEDs indicating to +5dB. Levels from -9dB to +5dB are indicated in 1dB increments. The REFERENCE switch provides either a continuous tone of 1kHz in the left position, with the right position providing momentary tone while the switch is held. The tone level can be set via internal DIP switches to -9dB, -12dB, -18dB, with the factory setting being -9dB. When tone is output, one LED on the meter display indicates reference level, with a second LED indicating deviation from this in steps of 0.2dB. The second LED should be adjusted to 0 level using the respective output master pot. The Mystere was tested with a Shure VPS-88 stereo microphone, Sennheiser MKH-416 short rifle and EV N/D 750 dynamic. In all cases, the gain range was more than sufficient for all applications, the audio quality was excellent and no untoward background noise was noted.

The limiters worked extremely well and were unobtrusive even under hard limiting conditions. The same can be said for the high-pass filters, which gave subtle contouring in the 80Hz position and 'traffic effectiveness' in the 120Hz position. The icing on the cake is the provision of a holder (optional) for a radio microphone transmitter, together with D-sub connector for powering and audio out. This feature allows a transmitter to be neatly interfaced to the mixer and avoid yet another bit of gear flapping around. Most popular makes can be accommodated.

The Mystere would appear to fulfil all of the criteria demanded by an ENG mixer though personal preferences will dictate an individual test. The unit is not heavy and weight, here, is a small sacrifice for the audio quality and features of a unit whose field-worthiness was demonstrated when an operator fell into a Swiss lake and the mixer carried on working. While this is not recommended review procedure, on dry land Mystere is worth your attention. ☐

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CreamWare Luna 2496 DSP

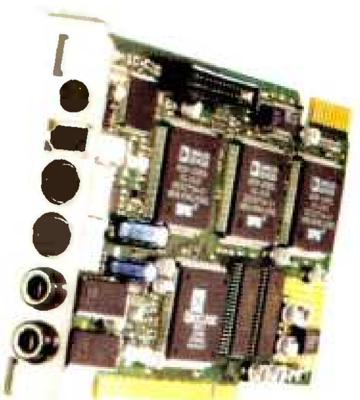
There is already such a wide choice of computer audio interfaces available that for many it would take really something special to raise a flicker of interest. **Rob James** finds it

AT FIRST GLANCE, Luna could be seen as being 'just another interface and break-out box'. In reality it is part of a family of products which together come closer than most to realising the 'Complete Studio in a PC' goal.

German-based CreamWare was among the first to make multitrack recording and editing an affordable reality for PC users. I ought to declare an interest at this point: one of the first reviews I ever wrote was for TripleDAT. The CreamWare approach to both hardware and software was and remains unusual, but I have considerable affection for the CreamWare way of doing things and the original card is still in regular use. Luna builds on the foundations laid by TripleDAT and the later Scope and Pulsar systems. Anyone familiar with either of these will quickly be at home with Luna.

All the CreamWare cards apart from TripleDAT are PCI types, built around Analogue Devices SHARC DSP chips, and all have S-TDM connectors to enable multiple cards to be integrated. This all begs an obvious question—why would you need the ancient legacy ISA TripleDAT card now Luna has appeared? In effect the TripleDAT card is now simply a repository for the key numbers required to run the TripleDAT software and plug-ins. I hope Creamware will make TripleDAT or some new recorder-editor software available for Luna.

Luna uses three SHARCs on a small PCI card. The connectors employed are familiar enough, but their configuration is less so. SPDIF I-O is taken care of by a single 3.5mm, 3-pole ('stereo') jack. MIDI In and Out are on mini DINs (short adaptor cables are supplied) and stereo analogue In and Out are unbalanced using stereo ¼-inch jacks. The remaining connector is the most interesting. Although it is pure 6-pin FireWire in hardware terms, CreamWare has used a proprietary protocol dubbed Z-LINK to produce a multichannel digital interface. This connects to the Luna 2496 convertor unit, providing power and eight channels of input and output.



The convertor is a neat metal box, 1U-high and half rack width. Sixteen phono sockets on the rear do the unbalanced analogue I-O at a nominal -10dBv. The front panel has a power LED and eight tricolour Input LEDs. These light red at -1dBFS, yellow at -12dBFS and green at -60dBFS. There is also an infrared receiver that may allow remote control of some sort in the future. The convertors are all 24-bit, 96kHz capable with a quoted THD+N figure of -94dB on the A–Ds and -101dB on the D–As. Subjective performance is in line with these figures—adequate yet not outstanding. To put this into perspective, consider the price point. As recently as a year ago 24-bit, 96kHz convertors were not available at anything approaching this level let alone complete with interface card.

The package includes software for both PC and Mac users. Copy protection is, like other CreamWare products, in the form of an alphanumeric key. Each software package requires its own key. The key is specific to the card and only has to be entered once on installation.

I think this is the least annoying form of protection to date. No dongle, no reliance on the machine configuration remaining constant and no key disk, while still providing excellent protection. Without the correct key for the card the software doesn't work so it can be made freely available.

ASIO, EASI and Windows drivers are supplied along with the Luna application. Like Pulsar and Scope, Luna uses CreamWare's own interface. Although things work in a Windows-like manner there are differences. While the graphics are pretty, I still don't see the point of reinventing the wheel.

Opening the application produces a toolbar with pull-down menus for filing and system settings and three large icons that open the Environment, 24-channel Recording Mixer or 16-channel Surround Mixer windows. The Environment window is where virtual modules are placed and connections made. The modules are divided into two groups, Hardware I-Os and Software I-Os. As the names imply there are modules for each of the hardware inputs and outputs on the card and for sources and destinations to other applications within the PC. Connections between modules are made by clicking on appropriate source and destination nodes. If multiple connections are required simply pressing the X key adds further wires. The 'hot spots' where you can click are not very large, at least in 1,028 x 764 resolution, calling for accurate rodent work. CreamWare has sensibly made illogical connections—like audio to MIDI—illegal.

A default Environment is loaded when the computer boots. If Luna is to be used simply as a hardware interface for other applications all you need do is design a suitable Environment and save it using the Save As Startup option. Other Environments can be designed, saved and recalled for specific purposes and you can also save your own default environment. In effect, all this is the equivalent of router snapshots.

Luna also has mixing capabilities and so double-clicking on either of the mixer icons places a mixer module into the environment and opens its control window. Both mixers are familiar graphical representations of hardware mixing surfaces. There are four aux sends and returns and the usual complement of mutes, solos and meters. The surround mixer is in 5.1 format—the output buses are LCR Rs Ls and LFE. Each channel has a surround panner with a horizontal and vertical slider for left, right and front, rear panning. A red ball in a square indicates current position. The ball can also be dragged with the mouse to pan.

A form of divergence is provided, as is a separate, post-fader, LFE send control. The divergence operates in a peculiar manner. If a source is panned hard left or right front, no signal is sent to the bus unless the divergence control is increased, while a source panned to left or right rear does not exhibit the same effect. There is no direct way to send a source to the LFE bus without sending it to one or more of the other buses. However, the curious divergence provides a workaround. If you want to send a source to LFE only, pan hard left (or right) set the LFE send to the desired amount and leave the divergence at zero. All mixer functions may be remote controlled via MIDI and Mixer presets can be saved and recalled.

The DSP confers the advantage of near-instant responses. There is absolutely no rubber-band effect when moving controls. I would like to see some more Luna modules to exploit the DSP power. A mastering mixer springs to mind.

The CreamWare manuals suggest the user should carefully consider whether the increase in storage requirements and loss of processing capability inherent to 96kHz working is essential to the project. Luna performs perfectly adequately at 96kHz, 24 bits, should a DVD or other project with these requirements come to hand.

Like many other PC products, Luna ships with a slim 'get you started' manual with the full manual as an Adobe Acrobat PDF file. While this saves on trees (at least the manufacturers trees, since many people print out the manual) in the case of Luna, there is a snag. The PDF manual is well organised but there is no overall index. The individual sections are well indexed and easy to navigate but I frequently found myself bouncing around through several sections to find the index entry I was looking for.

An optional I-O expansion board is available which adds ADAT I-O. A further Sync-plate option physically connects to this to add wordclock I-O and ADAT sync. With all the expansion options fitted a maximum of 72 audio channels are possible

With these options the Luna package might well be all the I-O many people will ever need. For example it works well with the Yamaha 03D which is limited to a single, 8-way digital interface card in addition to its analogue I-O. Fully expanded, it would work equally well with much larger mixers or as a patchbay for a complete studio.

It is also useful expansion for anyone with a Pulsar or Scope. When used in conjunction with either of these the Luna modules are available for use in projects in exactly the same manner as Pulsar or Scope modules.

Luna is a good way in and out of a PC or Mac in its own right, especially for people using an external hardware mixer. The convertor unit is small and especially convenient since it does not require a separate power supply. Driver support is particularly good with low latency ASIO, EASI, Windows MultiMedia and DirectSound all supplied.

But Luna's real strength is as a cost-effective starting point for building the Complete Production Environment in a PC. □

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TC Works VoiceTools

Tailored specifically to vocal processing, VoiceTools brings comprehensive pitch processing, dynamics and EQ to your Pro Tools system. **Jon Thornton** is on song

INSTALLING TC WORKS' VoiceTools plug-in package onto Digidesign's Pro Tools editor is a straightforward process. The installer automatically locates the correct plug-ins folder on the application hard drive and copies the requisite files. As the software is copy protected, it requires an authorisation code to be entered, which you get by registering. This means that you can easily re-authorise the software in the event of a terminal system crash. Usefully, the software will also run for 21 days after installation without the code, so that you can get down to work straight away.

The package comprises of the TC Intonator pitch processor and TC VoiceStrip EQ and dynamics processors. Inserting the Intonator plug-in across a vocal track in Pro Tools presents you with a number of options. Perhaps the easiest is to set up an auto-correction mode. The fundamental pitch of the vocal is measured by Intonator and compared with its tuning reference. The default for this is A=440Hz, but this is adjustable if, for example, the track was originally recorded to a different reference. Large buttons, arranged as a musical keyboard, allow you to then select the musical key of the song, and a pop-up menu allows the selection of a variety of musical scales. The incoming signal is measured against the resulting musical intervals, and it's deviation from the closest interval is displayed in red on a large bar graph.

A useful feature here is the ability to define a window around the intervals into which a note has to fall in order to be automatically corrected. This allows you to selectively correct massively out of tune notes or to be more selective. With a generally good vocal performance, this means that reasonably natural results can be obtained, even when used in a 'fire and forget' fashion.

The amount of pitch correction applied is variable, and controlled by a single parameter called Amount. At maximum setting, this corrects exactly to the correct interval, but lower settings allow partial correction. An Attack parameter sets the time that Intonator will take to apply the specified degree of correction. Fast settings give Cher's 'I Believe' repeat playing while slower settings 'bend' the vocal to the correct pitch.

Again, adjustments to these parameters yield quite natural sounding results relatively quickly. As with most pitch shifting devices, large shifts start to sound

Pro Tools Mix System
Pro Tools Software v4.x/5.x

Pro Tools Mix Plus, software v5
Macintosh G4/500

unnatural but smaller shifts lack the artificial quality found in some devices. Incoming pitches that fall between two intervals can sometimes cause the auto-correction to warble between the two, but this can be cured by hitting the NOTE HOLD button. This forces Intonator to hold the note to the currently corrected pitch. In conjunction with Pro Tools plug-in automation, tidying up a vocal track becomes a very quick process.

If globally applying automatic pitch correction across an entire track isn't required or desired, there are a couple of other ways to use Intonator. In Manual mode, hitting a note on the keyboard buttons will correct the incoming signal to that note as long as it is within the correction window mentioned earlier. This works well but it's a shame that a connected MIDI keyboard can't activate these controls.

The other option is to use Intonator in Automatic mode, but to define a custom scale which only applies correction to specified notes, useful if a singer repeatedly struggles with a particular note or notes.

Finally, global pitch adjustment is available in both Automatic and Manual modes, which applies a uniform amount of correction on top of everything else. Another useful feature is a post-correction high-pass filter, which can either be fixed at a certain frequency, or put in an 'adaptive' mode. The latter means that the filter is applied at the specified frequency but as the fundamental pitch of the incoming note approaches the cut-off frequency, the cut-off adapts and moves downwards. This is really effective, and lets you tighten up the low end of a vocal track without fear of making it sound too thin.

VoiceStrip is a separate plug-in that features a high-pass filter, noise gate, EQ, compressor and de-esser—in short nearly all of the outboard tools needed to process vocals. Again, inserting the plug-in across a Pro Tools track (or tracks—if inserted across a pair of tracks the plug-in works in linked stereo) presents you with all of the parameters on one screen. Its use is speeded up by the exclusion of certain parameters. For example, the gate has no user definable attack and release settings, and the EQ section is 3-band, but with a fixed high frequency shelving filter and no Q settings available on the other two bands. Some people might bemoan this lack of flexibility, but in practice you don't need this level of control to achieve extremely good results.

The signal chain is ordered as outlined above —HPF, gate, compressor, EQ, de-esser, although

the order of compression and EQ can be reversed if required. The filter isn't adaptive, as with Intonator, but does allow the selection of either a DC removal mode, or a traditional 12dB/octave filter with a selectable range between 60Hz and 120Hz.

The gate has two controls, threshold and intensity, the latter apparently a complex mixture of range, attack and release. In practice, lower settings initiate a gentle attenuation of level when the gate is closed, and higher settings result in a much more obvious removal of signal.

The EQ section has three bands. The low band has a shelving response, with a range between 100Hz and 350Hz. Although the slope is not adjustable, this increases with the amount of cut or boost. Similarly, the mid band is a peaking filter with a range between 700Hz and 7kHz, and the bandwidth of this filter narrows as the gain increases. The high band is a shelving filter with a turnover frequency fixed at 2.5kHz. Although the specified bands and responses might at first appear restrictive, in use they are just about perfect for most vocal work. The EQ itself sounds quite warm for a digital unit, with a slight graininess in its character. A unique feature that adds to this character is something TC refers to as SoftSat. In practice this means that high levels of boost in a frequency band are soft-limited, while at the same time adding subtle harmonic distortion to signals that would otherwise be clipped. This feature can be enabled or disabled on the EQ section as required.

This same algorithm is permanently applied to the compressor section, which has the usual controls for ratio, attack and release, and a 'drive' parameter that appears to be a combination of input gain and threshold level. Lower settings give transparent levelling to a vocal and higher settings coupled with slightly higher ratios start to bring the SoftSat algorithm into play, again giving a slight graininess to the sound reminiscent of vintage devices and their current reissues.

The de-esser section is separate from the compressor, and again is easy to set up. A side-chain monitor helps to identify the cut-off frequency, and a threshold level sets the onset of de-essing. Usefully, the threshold level is set relative to the average signal level of the material. This means that if the overall signal level drops, the threshold level will also drop proportionately, allowing de-essing to occur over material with a wide dynamic range.

Taken together, Intonator and VoiceStrip are extremely useful tools for working with vocals. While the individual sections may not be the absolute best in their class, or have the flexibility in control that some users may want, as a package they do work extremely well together. The sound of VoiceStrip might not be to everyone's taste, so its worth auditioning first, but if you do a lot of vocal work with Pro Tools, it could pay for itself in the time it saves you alone. ☐

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THE X FILE

Flamboyant American John X talks about his work as a producer, engineer, mixer, musician, composer and programmer to **Richard Buskin**, who learns that men in baggy trousers can command respect

MUSICALLY, JOHN X is at the cutting edge. A producer, engineer, mixer and remixer, as well as a composer, keyboard player and programmer, he has lent his multifarious talents to the recorded efforts of Garbage, U2, David Bowie, The Rolling Stones, Marilyn Manson, Michael Hutchence, Black Grape, Black Sabbath, Ice Cube and Shonen Knife, to name but a few, as well as several movie soundtracks. Hell, he even has a cameo role as a super-cool, psychedelic deejay in the new film *Boys and Girls*. Sartorially, his style is less easily quantified.

'People are always saying, "Dude, you're the mixer who wears the dresses",' he laughs. 'I have to tell them they're culottes. I like to keep things flamboyant and bright, if not just downright silly.'

As for the name, John X fits more easily onto a CD than John Volaitis...

'The "X" was given to me by someone called Joe X,' he recalls, 'a dear friend of mine and a great drummer. We were in bands together and he was probably one of the first supportive and inspirational people I knew. Well, he named me John X—whether by accident or design—back in the early eighties, and it stuck.'

Born in Flushing, New York, John X took piano lessons at an early age, but by his own admission he was always 'More of a nerd. In fact, I was definitely a supernerd,' he asserts. 'I was into really geeky stuff.'

This amounted to taking apart and reassembling virtually all of the electronic items around the house, and consequently learning how they worked. After a stint as a general gofer at Manhattan film postproduction facility Sound One Corporation, he landed a summer job building 949s for Eventide prior to attending college in Pittsburgh. Working as a part-time college deejay, Mr X had his license revoked for lewd and lascivious on-air acts and subsequently made a speculative move to the West Coast.

'I came out in search of bikinis,' he admits. 'I told my parents I was coming for an interview at Cal Art, but I left after five minutes on campus and started working for George Tobin who had just finished producing Smokey Robinson's 'Being With You'.'

A chance meeting led to a spot as an assistant at Tobin's San Fernando Valley studio. Here he worked with Smokey Robinson, Natalie Cole, Herb Alpert, Thelma Houston and 'a million Latin artists'. R&B work came John X's way, but he didn't delve into rock until moving to the Malibu-based Shangri-La facility then owned by Bob Dylan and members of The Band. Here he learned multi-miking, and a succession of different jobs introduced the X man to an eclectic assortment of projects and musical genres: heavy metal at Prairie Sun Recording in Cotati, Northern

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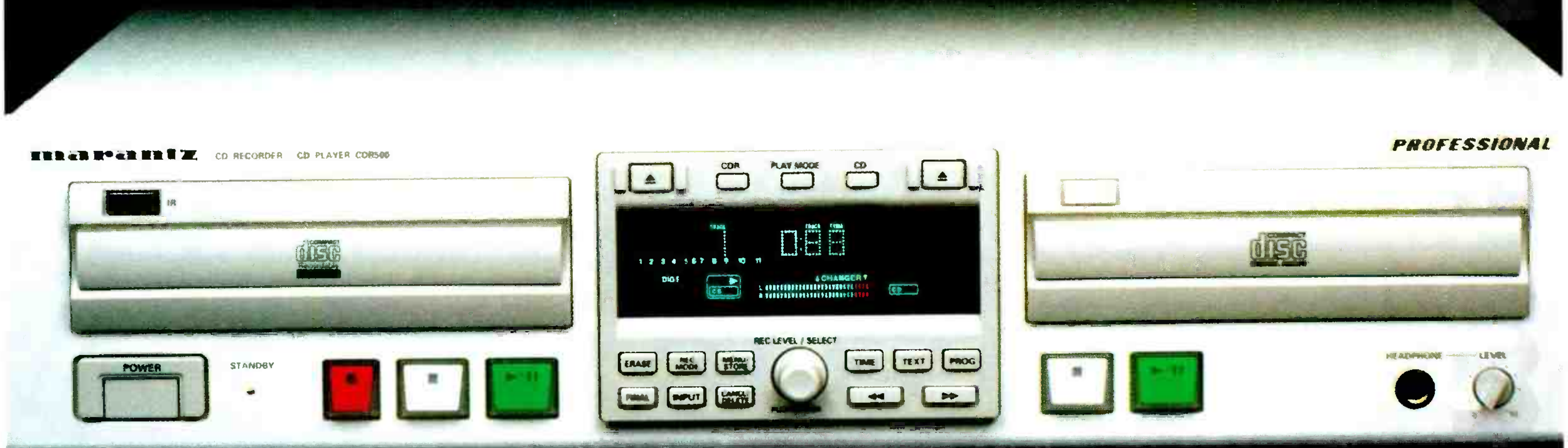

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California; low-budget hip-hop, metal and R&B at Fiddlers Studio in Hollywood; and the more mainstream efforts of Tracy Chapman, Iggy Pop, Tanya Tucker, Peter Frampton and Tom Jones & The Art of Noise at Powertrax, also in Hollywood. All this experience has contributed to John X's style and so we began by discussing how he applies what he has learned.

What did the rock-work at Shangri-La teach you after your initial experiences at George Tobin's facility?

Rob Firboni was the producer-engineer in charge at Shangri-La, and the girl who engineered for him, Terry Becker, taught me tons of stuff. It had been classic late seventies R&B-style recording in the way that everything was super-close-miked and super-controlled. There were a lot of different rhythm sections, but as far as I was concerned it all sounded pretty much the same except for the different singers. Everybody was listening to Steely Dan records and comparing whatever they were doing to that. It was a real production line—you could practically leave the same mics up for every session and it would work until somebody bumped into something.

With Terry Becker, on the other hand, there was way more experimentation. She and Rob Firboni were pulling me back into an older place, showing me more about ambient miking and the purist style of recording, with a signal path taken down to where you just move the microphones. It was at Shangri-La that I started diving into effects. In fact, I still have tapes that I made back then, which I use now for people's remixes. There are these insane feedback loops which were created when I had no idea as to what I was plugging in. I'd get things right on the verge of where they were going to go insane and then start modulating them. I use those tapes all the time.

Anyway, I got to work with some great people there. I remember Mick Fleetwood, who I don't think ever spoke to me, but who would just sit down and point at me as if to say, 'Okay, I'm ready to record now'. I'd look over and he'd be drumming with towels and tennis balls on his tom-toms while he was doing these fills. He'd do that just to mute the toms or get a different sound, and that was the first time that I saw anybody do that kind of thing. He wasn't just a regular cat.

After you left Shangri-La, you got more involved in songwriting. Did that teach you how to structure a band within the context of a mix?

Absolutely. I would start everything with a piano and a click. The piano was in the other room, and so all of the songs would start with me running to it and you could hear me landing on the bench. Back then, in about 1982 or 1983, there were no samplers around. There was an AMS delay, but it wasn't that quick. It was okay for some stuff, but not for getting real fancy in the way that we've grown used to now. I'm sure I initially used that for my kicks and snares, and then I'd just tap the stuff in; it was easier.

At the end of the eighties you started life as a freelance by working for three months at a Tokyo studio. Did the wealth of equipment there have any significant impact in terms of how you worked?

The first week you don't realise it, but then after a while it's like, 'Alright, I'm on another planet'. We were at Victor Studios and there were five almost-matching SSL rooms. I always go into the gear closet to see what else they'd got and I discovered five DMP 251s—those reverbs with the stick shifts on them—just sitting there. I looked at the assistant and said, 'What's up with these, dude? Why don't you have these out in the control room?' and he said something like, 'Oh, we don't know how to use them'. I said, 'Wheel one in, dude, I'll show you'. Because there's that one thing where if you don't hit the Set button it doesn't change anything, so you sit there all day moving the shift and, although it looks good, nothing really happens. I showed the assistant how it works and I ended up having three of those machines in the room. I was doing discreet verbs on each side of the toms and so on.

What is in your home setup?

I have a Pro Tools rig and I use E-Magic Logic for sequencing. I go back and forth between both, as there are things that are good about each one in terms of what the other one doesn't do. My first sequencing rig consisted of an Atari 520, which you plugged into your TV set and was really minimal, along with a couple of keyboards like an Oberheim Matrix, a CZ101 and an Alesis drum machine. My second rig had an Atari 1040.

Most of my keyboard stuff is made by Roland. Obviously I must like it, even though I never realised that I had such a corporate obsession. I've got their EG101, the Groove keyboard; an XP30 expandable synth, which I use more for the normal orchestral kind of stuff, but which also has some great, crazy sounds; the JP8080; and the Doctor Groove 202.

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A talented musician, successful composer and recognized producer, Lenny White is considered one of music's real innovators. He's played with such greats as Miles Davis, Chick Corea and Stanley Clark, and has also collaborated on some of the most recognized and influential music of the past three decades, including the new CD *On the One*, with Sammy Peralta.

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Sammy Peralta loves music. That pure and simple fact comes through strikingly clear as he sits at his keyboard tinkering with half-written tunes. Sammy's background includes work with talents including Tito Puente and Willie Bermudez. "I have to be careful because I can get so lost in the music, I sometimes forget I have a family that would like a little of my attention too".

One last point: Sammy Peralta's new CD ***On the One*** featuring Lenny White was mixed entirely with LSR monitors.

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There's also a Kawai K1 from the old days, a Matrix 6, and my Akai 612 sampler.

I've got the tc electronic FireworX which I love, a Sony M7 which I also love, and the Electrix MO-FX which is almost like a deejay toy—I can blast the MIDI clock into it coming from Pro Tools and it automatically sets delay times. It's got delay, flanger, tremolo, *phaser, auto pan and distortion*, and switches to engage or disengage each one which are all MIDI recordable. So, you can do super-tweaky stuff. It's a total remix mutilator toy and it's really fun, I love it.

I don't have much in the way of mics. The only one I've been using forever is the Sony SM7—I've used it on Hutchence, I've used it on Mick Jagger, mostly because frontmen like it. Jagger had never used it before, but I stuck it out there when I was faced with this live scenario and he was open to trying it. The first response I got was Don Was coming into the room going, 'Dude, what are you doing with Mick's vocal?' I said, 'I've got an SM7 and a bit of compression...'—I think it was an LA2A—and he said, 'Well, it sounds fantastic. It sounds wide open'. And it did. It sounded really good.

In terms of speakers, I've got Mackie HR824 powered monitors, Dynaudios and Genelecs, but I prefer the Mackies in that they can handle the bottom end that I'm pumping into them without shutting down. I got used to the ones that Danny has up in his room and I felt like they were good, rock-worthy speakers.

I've also got the Mackie CFX20 mixer, but I see that Roland's got a digital mixer so I've got to look into that.

On the recordings of Garbage, Bowie, Marilyn Manson and U2, would you say that you mixed the remix or remixed the mix?

I mixed remixes for each of them; 'Staring At The Sun' and 'Happiness is a Warm Gun' for U2, 'Queer' and 'Stupid Girl' for Garbage, 'Little Wonder', 'Dead Man Walking' and 'Fun' for David Bowie, and 'The Beautiful People' for Marilyn Manson. Danny [Saber] often does a remix and then hires me to mix it, so I consider my work to be a mix. Still, that causes so much confusion—'What do you mean? You mixed the remix or you remixed the mix?' Let's just say that I'm the last one who touched it.

When something I've worked on is listed as a Danny Saber remix, 90% of the time I don't play on it, and I don't do very much production of any kind. It's all about finishing and making it sound awesome. One thing I will always do is vary the levels of vocal mutilation, either adding harmonies or vocoders. That's because with a remix I feel it's okay to go over the top with my vocal presentation—even on the work of established artists.

The Rolling Stones have been known to take their time in the studio. What was it like tracking 'Out of Control' 'Already Over Me' and 'Gun face' on their Bridges to Babylon *album?*

That was all done at Ocean Way in Los Angeles, and then I mixed 'Gun Face' at Sarm (West) in London with Danny and Mick Jagger. At Ocean Way I recorded the entire band live, and that was just about the most high-pressure gig I've ever walked into. So I went out shopping and bought a bunch of new, hideous outfits, just absolutely beyond anything I'd worn before. Good or bad, I was going to make an impression.

I was told to show up at five, and when I got there someone had set up everything for me. My assistant hadn't done it so I asked him to tear it all down, but then another engineer walked in and said, 'Dude, if these guys walk in and you're setting it up, you're fired.' I was like, 'Well dude, let me hang myself.'

I didn't know what to expect. I didn't know how heavy or loud or hard the vibe would be, so I decided to stick to my old approach and it worked just fine. There were some rough spots getting the cue together and there was a lot of people in the room — there were two percussionists and everybody was playing live in one room. It didn't take long before Charlie Watts' drums sounded just like him, and it was even easier when I turned up Keith's mic. I've heard 80 million people try to sound like that guy over the last 20 years, and it was really nice to hear the real thing.

Of course, the board hadn't been zeroed out, so I turned up Keith's channel and there was this reverb. I was thinking, 'Keith plays through reverb?' I didn't realise it was coming from the board and then I heard from the back of the room, 'Turn off that bloody fucking reverb'. I was thinking, 'Who did this to me? How can you sabotage me this way?'

Anyway, the bottom line is 95% of the stuff was first take. Mick Jagger was in the room with the rest of the band, playing acoustic guitar and singing live. I kept the leakage to a minimum, switching off to a lot of dynamic mics and trying to reduce the leakage to the point where it was pleasant, controllable and workable. When I had first walked in, every mic in the room was a condenser, which is nice in theory, but not nice when there are nine people playing live in a live room. So, I switched to dynamics and I don't think I compromised anything by doing that because the tracks sound really natural.

I was only supposed to do the one song, but they said, 'We have a good vibe tracking here. Can you stay?' I said, 'Yeah, I'd love to,' and the best thing was that I got to leave before they got sick of me... Before they got sick of those outfits.

Talking of which, what is it with those clothes?

I've seen the way that other people have dealt with stress and conflict, and I like to retain a bright and creative environment. Sometimes you've got to go nuts and sometimes people have to get pissed off, but it's really good to get a handle on things and reel them back in quickly. Still, that happens more with bands, and then there's the artist versus producer thing that used to happen more when I worked with other producers, but nowadays I don't really work with them as much. They often send me the tapes and so I've become protected from that. □

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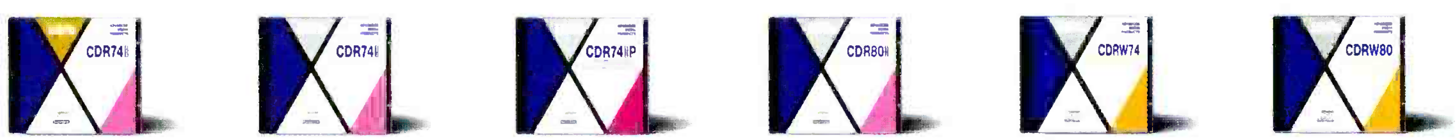

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FEAR OF HEARING

When you have nothing to fear except fear itself, a soundtrack to your nightmare might seem an indulgence. **Kevin Hilton** talks to the sound designers behind the BBC's Phobias series

AN AGORAPHOBIC knows how it feels to fear open spaces. An arachnophobic recognises only too well the creeping sense of dread when confronted by a spider. They may be able to describe this to someone who does not share the same terror, but simply stating the effects of a phobia is not enough for today's graphic, tabloid television; the viewer needs to be able to feel and share, by some means, the anxiety and trepidation.

Film and television producers have long known the emotional effect music and sound effects can have. Phobias, a 3-part series made by BBC Science, relies partly on this style of representation of the subject matter to achieve an effect; but takes the concept further by processing the soundtrack, introducing atonalities and 'morphing' the audio to unsettle and disturb.

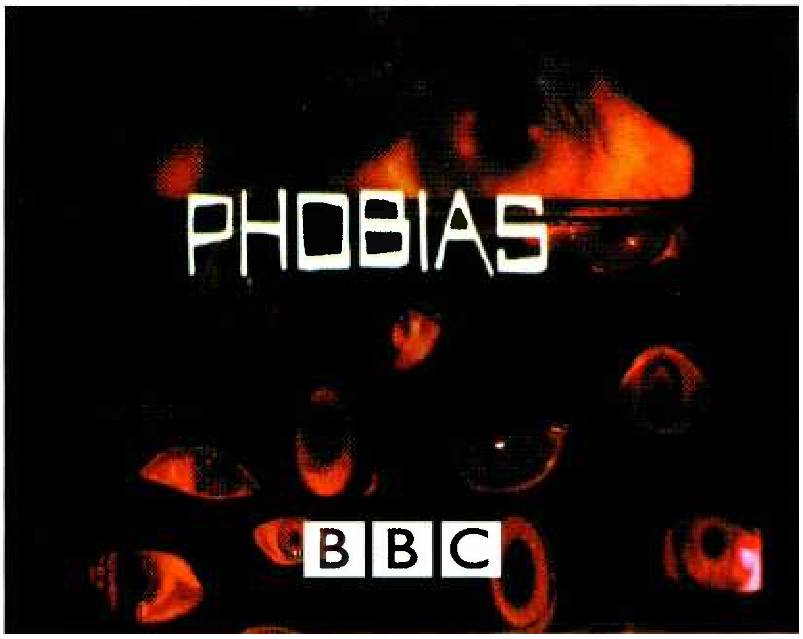
Such stylings can be traced back to the electronic treatments on *Forbidden Planet* and the work of composer Bernard Herrmann. The theme music of *Phobias* acknowledges Herrmann's influence and vision, as he was one of the first to treat music and effects in the same way to achieve the end of the film maker. It is a composite homage, featuring pastiches of music from Hitchcock's *Pyscho* and *Vertigo*. The incidental soundtrack is an eclectic mix of classical, angelic choral, strident New York jazz and *Godfather*-style Italian folk music.

Natasha Brody, the producer and director of the series, explains the thinking behind this approach: '*Phobias* is based around the most basic and extreme human fears and it was of ultimate importance to create a sound and music experience that reflects closely our exploration of what goes on in people's heads when they experience a phobia.'

To realise these aims, Brody approached Adelphoi Music, an independent music production and sound design house that had already won some attention for its work on another BBC series. *Walking With Dinosaurs* was the surprise hit of the autumn 1999 schedules for BBC1; Adelphoi's part in that production was to create the sounds and noises of creatures that had died out centuries ago. The thinking seems to have been that if its sound designers could give voice to a T-rex, then they could make viewers hear what it is like to have a deep seated fear of heights. Sophie Taylor, Adelphoi's producer on *Phobias*, observes, 'We approached the project very much as you would a film score, with sound designers and composers working in collaboration rather than isolation. The composers worked in a variety of styles; one of the key themes was setting beautiful classical music against subtle, unsettling sound design to create a strange atmosphere.'

Adelphoi was founded by musicians Charles Hodgkinson and Kirk Zavieh in 1994 as an independent record and music production company. The founders are at the heart of Adelphoi's work, but the company has a roster of other composers. These include Jamie Masters, who worked with Hodgkinson and Zavieh on the soundtrack of *Phobias*. The team was completed by sound designer Andy Sherriff.

'Among the three of us are those who consider their forte to be sound design and some who consider themselves better at classical music,' Hodgkinson says of the team's structure and demarcation. 'When one is writing a lot of music, you're working every day on a music project; but hopefully one has the ability to join in other areas.'



The legacy of Bernard Herrmann is that the music and effects making up a modern soundtrack usually cannot be separated; certainly their effect on the audience is much the same, whether it is realised consciously or subconsciously. Hodgkinson acknowledges this, saying, 'Music and sound design are so inextricably linked—it's all about creating atmospheres.'

The crossover that Hodgkinson identities is illustrated by Andy Sherriff, who is also a musician, albeit one who is very interested in sound design. 'I started getting into dance music,' he explains, 'and that got me into the idea of music being sound unto itself. You look at what is being done in the clubs and it's all about creating sounds to go with the light show.'

Sherriff continues that the work on *Phobias* was not sound design in terms of spot effects, but creating drones and eerie music sequences, often morphing and filtering the sounds. 'There are sections that are just straightforward music, but that's okay for me because I enjoy the engineering side of things as well.'

Adelphoi's Covent Garden studio centre is organised on communal lines, the idea being that musicians, operators and projects can be swapped between the various suites. Hodgkinson worked in a room equipped with a Mackie D8b console, a 12-output Soundscape digital audio workstation and an Akai S6000. 'The 8-bus is very useful in a situation like this where you're working very closely with picture,' Hodgkinson says. 'It is also good at dealing with sudden dynamic changes, spot effects and changes in equalisation. There is a large number of channels and the routeing is flexible; the automation also gives a reasonable amount of flexibility.'

Hodgkinson explains that the desk's recall facility enabled mixes to be called up for the programme directors to see how the work was progressing. The 8-bus' channels were configured with 24 on the first bank and 24 on the second. Of the Soundscape's 12-outputs, eight fed digitally into the desk; two of the inputs and four of the outs were either analogue or SPDIF, although the device was originally configured as eight digital ins and outs on TDIF. 'It was expanded as far as the system will go without moving up to the red, although we will make that move in the future,' Hodgkinson says.

Studios were divided between sound design elements and straight music composition and recording, all the sections being brought together at a later date. Logic Audio is Adelphoi's common sequencing package, while the visuals were run on Soundscape's vision system. 'Usually we're digitising video footage for commercials work,' Hodgkinson comments, 'so digitising 40 minutes of footage was quite a tall order. But it gave us perfect sync: as the Soundcape run's, the visuals automatically play and drive Logic, which in turn drives the desk using time code.'

Adelphoi's other studios house a Yamaha 02R and two Mackie 32:8 analogue consoles, although there are plans to make the facility a totally 8-bus zone. Alongside the S6000, other processing equipment included a Lexicon PCM80 for the morphing effects, a Studio Electronics SE1 analogue modulator, a Roland JV1080 (used to supplement the S6000 and provide a warmer sound, particularly for some of the classical sounds), an Emu Proteus 2000, a Sherman filter band, Meatball pedal effects unit, Colour Sound distortion device and a Kurtzweil Micro piano, used for the synthesised piano and string sounds.

Purists may still balk at the thought of electronic recreations of acoustic instruments, but Hodgkinson maintains some realistic classical orchestral effects were achieved. 'We also combined our sample

PHOBIAS IMAGE DESIGN BY SAM BURFORD OF TRANSIENT

Adelphoi *Phobias* Team. Left to right: Charles Hodgkinson, composer; Kirk Zavieh, composer; Sophie Taylor, Producer; Andy Sheriff, sound designer; and Jamie Masters, composer

generators with real instruments,' he says. 'Wherever possible it helps to include something real, so there are a few violins, a flute, saxophone, harmonica and vocalist [although the voices are used purely as another instrument]. Voices add a very human element, which is necessary because these are very human stories of psychological conditions, even though the voices may be processed and elongated.'

Among the sequences specially 'scored' and designed by Hodgkinson, Zavieh, Masters and Sherriff was of a woman who is afraid of riding in cars. The story begins with slow-motion footage of crash test dummies in action, accompanied by what Hodgkinson describes as 'very calm pseudo classical music'. The scene suddenly jumps to busy traffic in New York, underscored by jabbing, very modern jazz-style saxophone work. 'By using atonal music and sounds we're trying to create the same sort of feelings that the patients experience.'

Andy Sherriff adds, 'We started with natural sound and processed it. When we arranged the tracks ready for dubbing, both the original recordings and the "warped" versions were there, so the dubbing engineering could switch between them. This creates an effect of moving from reality to an altered state and back again. We added all sorts of things over the location recordings, like scrapping cymbals and so on, just to create mayhem.'

When the Adelphoi team first received the film footage, the narration had not been recorded and so they were not certain where the commentary would be and where they could place music and effects. 'When we went through it, we thought there were certain bits that were good for sound design,' says Sherriff, 'so we did it as we went along, almost improvising in some cases.'

The current trend among sound designers is to shy away from sound-effects libraries, either because the sounds are too familiar or too limited. Adelphoi has built up its own library of effects and also records its own Foley. One sequence that does feature library sounds is of a man who obsessively rides stomach-churning roller-coasters, but will not board an aeroplane to visit what he sees as the ultimate ride in America. Sherriff took the sound of train tracks and warped them, producing a surreal result. 'By juxtaposing the visuals with the sound the viewer is hearing, we can create a different type of reality,' he says.

Creating eerie or unsettling noises in isolation may be fine for horror movies, but *Phobias* is a documentary with a scientific basis, no matter how populist the presentation. Consequently, the sound design had to have some relation to what the phobics are experiencing. 'The directors were very good,' says Hodgkinson, 'they were very clear in their background briefings to us. They described to us as well as they could how these people felt and how they reacted to individual circumstances. We also tried to get a sense of the issues involved for the people. Our designs came out of a combination of spending time with the director and watching all the footage.'

With four people working on different elements of the soundtrack, it was important for one person to oversee the whole process. This role was taken on by Hodgkinson. 'There had to be some continuity,' he says. 'For example, it is effective if the signature theme reoccurs occasionally during the incidental music, so it was good to have some cross-fertilisation between the different studios. It helps that all the suites are next door to each other.'

BBC1 has screened the series as three 40-minute programmes. For the US market two hour-long versions were compiled from the footage, although, due to the amount of advertising on American television, these are not full hours. The finished soundtrack was laid onto the eight tracks of a Tascam DA-88, generally arranged as four tracks of music and four of sound design. These tracks were 'chequered'—one music cue on Tracks 1 and 2, the next on Tracks 3 and 4—to give the dubbing engineer some flexibility and control when it came to crossfades, rather than predetermining them. The dub was carried out at the Edit Store.

Work on the sound design stretched out over three months, although it was not a constant process. 'It was quite time consuming,' says Sherriff, 'but it was a lot of a little bit here and there. We did an episode at a time and it just seemed to take a long time.' Sherriff admits that trying to differentiate between the different phobias and give each one its own 'personality' was often difficult, but feels that a good effect was ultimately achieved.

Phobias has been a critical success on UK television, with writers commenting on how the inner panic of the sufferers was conveyed. It can only be hoped that Adelphoi's work in creating aural representations of such as claustrophobia does not induce telephobia (fear of television) or acousticaphobia (fear of noise). If there is a second series, one wonders if the sound designers will be called upon to interpret verbophobia (fear of words), pogonophobia (fear of beards) or philophobia (fear of falling in love).

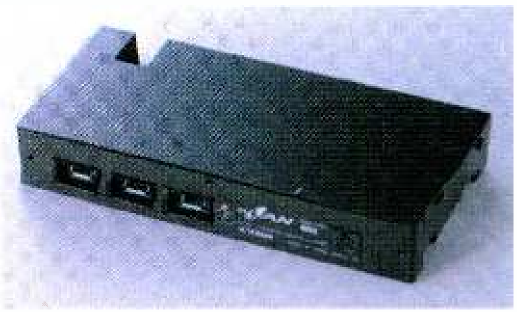
PHOTOGRAPH BY JIM NAUGHTEN

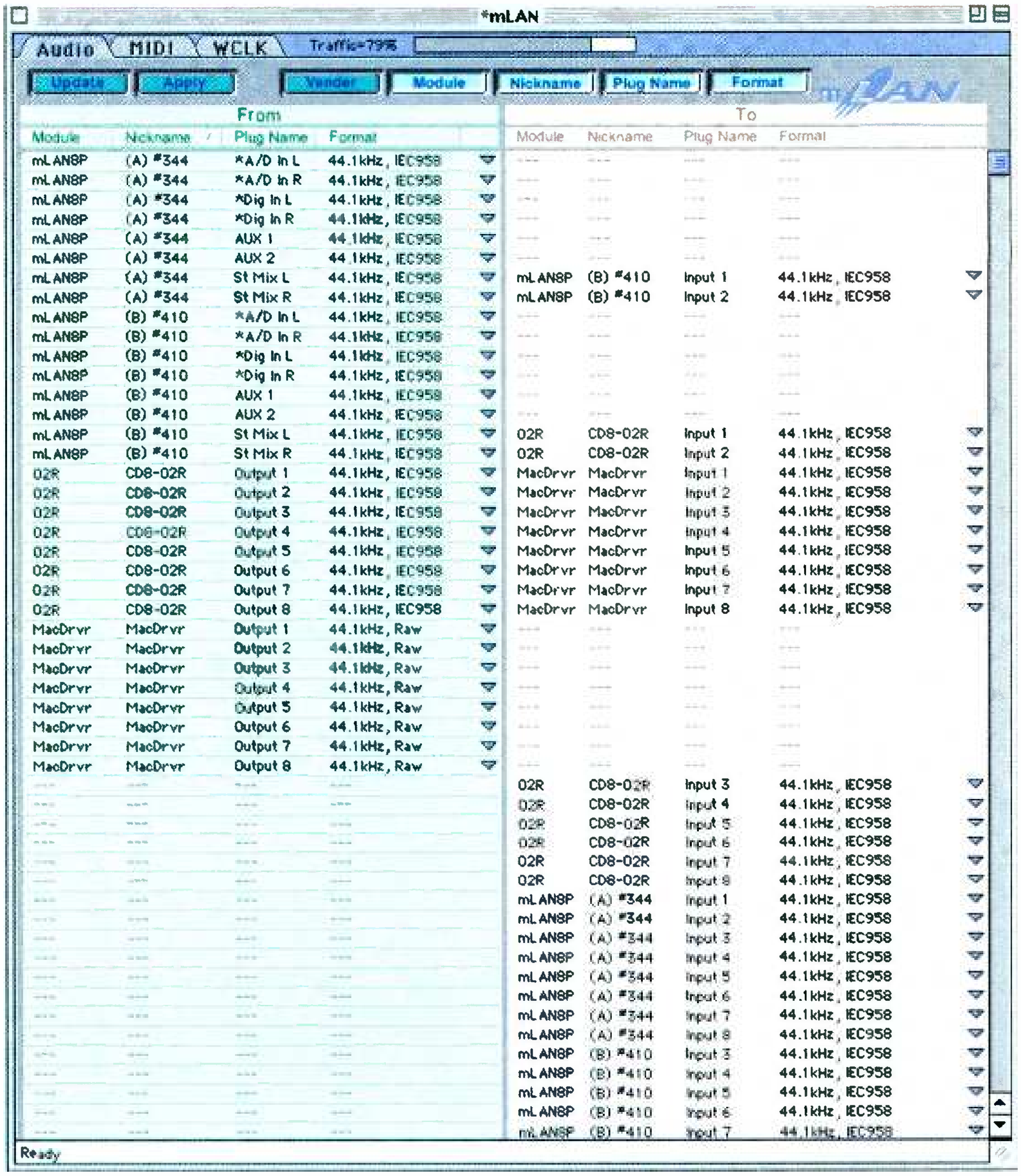
FIRESTARTER

Will Yamaha's mLAN audio and MIDI networking protocol fulfill it's promise and revolutionise the very concept of the studio? **Simon Trask** brings news of the nodes

THE EMERGING WORLD of converging digital technologies is itself converging on a networkable interconnect standard that will help make convergence a reality. Known formally as IEEE 1394 and informally and popularly as FireWire or i.LINK—the names given it by Apple and Sony respectively. This high-speed serial interface is finding its way into more and more areas and applications of computing and consumer electronics. What's more, its functionality is such that it can straddle professional and amateur applications with ease—and now, through a Yamaha-developed extension to IEEE 1394 known as mLAN, it's set to infiltrate the world of professional audio and music production.

FireWire was developed in the early 1990s by Apple as a desktop LAN and was taken onboard by the IEEE 1394 Working Group, which turned it into a standard in 1995—the full name of the core document is actually IEEE 1394-1995. Today there are over 70 standards documents relating to it, around seven of which concern the core standard while the remainder cover device—or application—specific uses and extensions. Broadly speaking these latter break down into consumer electronics (AV/C—audio-video consumer or compatibility), computing, content protection and other categories, the last encompassing industrial, instrumentation and automotive applications. Meanwhile, IEEE 1394 is being adopted as the interface for digital video cameras, digital VCRs and digital TV, as well as for storage media. Home networking is also in its sights, while yet another extension—IP over 1394—hooks it up to the Internet. This year has also seen Microsoft integrate support into the various iterations of the Windows OS (include its portable device OS, Windows CE).





Following on from the likes of MIDI, ADAT and TDIF, mLAN is the latest 4-letter digital interface acronym for audio and music professionals to get to grips with. Short for music Local Area Network, mLAN is the most ambitious interfacing protocol to date, encompassing bidirectional multitrack audio and multiport MIDI over a single cable within a flexible local area networking infrastructure. Yamaha says that at the currently supported IEEE 1394 data rate of 200Mbps it's theoretically possible for mLAN to handle more than 100 channels of CD-quality digital audio data and more than 4096 (256 x 16) MIDI channels.

IEEE 1394 supports both asynchronous and isochronous (real-time) transfer of data, so it can handle both streamed and file-based transfer (simultaneously, if required—up to 80% of bandwidth is available for isochronous transmissions). Isochronous transfer within IEEE 1394 is a method that guarantees the right to transmit or receive data at a fixed interval of 125µs.

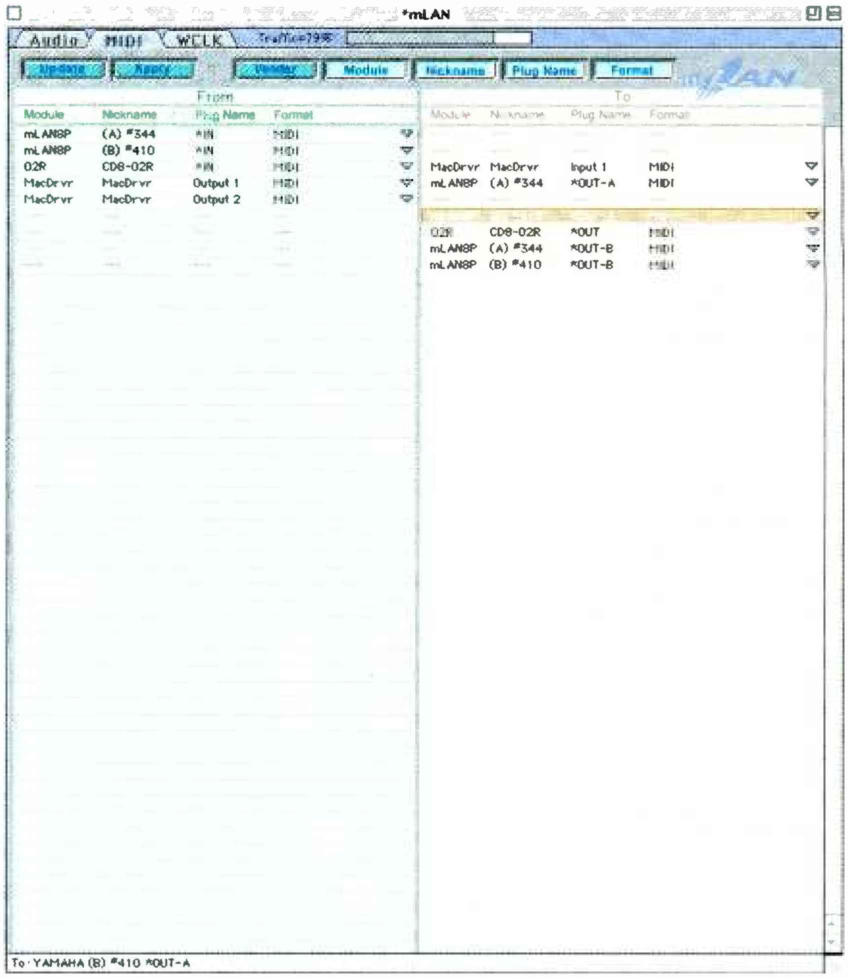
Also, consider that as a networking protocol IEEE 1394 allows up to 63 devices (nodes) to be connected, all being individually addressable on the network —what's more, any device can address any device (peer-to-peer networking). Within the network of devices, one node has to be selected as the 'root' node. This is also known as the 'cycle master' node as it provides the master clock for timing synchronisation of all the nodes. In addition it arbitrates bus access rights.

If you have 16 or fewer devices you can daisy-chain them together; more than 16 requires a more sophisticated 'branching' topology. Devices connected on branches are known as 'leaf' nodes. Another feature of IEEE 1394 is that devices are 'hot-pluggable', meaning that you can plug and unplug cables at any time, even if the power is on. When you add in a device it will be automatically configured into the network. However, (un)plugging or switching on-off a non-leaf device, including the selected root node device itself, will induce a 'long bus reset'; this initialises and reconfigures the bus, which obviously interrupts real-time data flow such as audio and MIDI data. If it's a 'leaf' device, though, (un)plugging the device or turning its power on or off will induce what's called a 'short bus reset', that doesn't interrupt real-time data on the network.

IEEE 1394 also has a capability for powering devices via the cable itself, although with current mLAN devices it will be the case that in a system consisting solely of mLAN devices the system will only function if (mains) power to all the devices is turned on.

mLAN itself has had a long gestation. Yamaha began to consider developing a music networking protocol prior to the introduction of IEEE 1394, then after deciding that 1394 would provide the best hardware interface it set to work developing extensions to the protocol and a proprietary chipset to handle the audio and MIDI requirements. A lot of work went into what became the Audio and Music Data Transmission Protocol v1.0, which was released in May 1997 under the auspices of the Audio-Video Working Group of the 1394 Trade Association. Subsequently it has passed a PAS (Publicly Available Specification) vote by the IEC, which has given it the official designation IEC PAS 61883-6. Fortunately it's also known as the A-M or, latterly, A&M protocol.

In essence, A-M defines an instance of a real-time data transmission protocol describing isochronous transmission of audio and music data over IEEE 1394-1995. This includes the transport of IEC60958-conformant digital audio data, raw audio samples, and MIDI-conformant data as defined in the official MIDI spec, along with 32-bit floating-point data as defined in IEEE 754-1985. The protocol is defined as being applicable to 'all modules or devices which have any kind of audio and-or music data processing, generation and conversion

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This year saw the release of TA Document 1999014, 'Enhancement to Audio and Music Data Transmission Protocol 1.0' introducing a new generic model of packetisation with adaptation and application layers, and interpreting the original, 1.0 model as a special instance of this model. Through this development it introduces support for DVD-Audio and SACD, as well as SMPTE time code and sample count transmission (TA 1999024, 'SMPTE Time Code and Sample Count Transmission Protocol v1.0').

Although popular conception will no doubt encompass this audio and MIDI data transport functionality within 'mLAN', technically speaking mLAN describes and defines the higher-level command and connection-management functionality that allows the A&M-defined audio and MIDI streams to be used within an equipment setup. It also defines the functionality that needs to be implemented in devices that want to carry the mLAN designation. The use of the mLAN logo on a product will indicate that the full capabilities of the mLAN protocol are supported. It will be possible for an mLAN device to interface with a device that only supports the A&M protocol, in which case the connection management capabilities afford by mLAN won't be operative between the two devices.

In July, Yamaha announced an mLAN licensing program, with licenses offered by the company as from September on a royalty-free basis to stimulate adoption of mLAN by pro-audio and MI industries (no Apple-style slip-ups here). There are three licensing options. Technology Transfer (Example A) has Yamaha providing both mLAN adaptor boards or driver software and their interface specifications; in this case licensees need only design the client interface. This would be, for example, sequencing software developers wishing to integrate mLAN support into their programs. For Technology Transfer (Example B) Yamaha provides circuit diagrams, firmware and other design information, allowing licensees to build mLAN products of their own design. Finally, with Implementation Contract (Example C) Yamaha provides all intellectual property usage rights and technical data necessary for implementation of specific mLAN functions, allowing licensees to develop mLAN-compliant products. As it happens, at the MacWorld show in January of this year Yamaha showed an S80 synthesiser plugged into an 02R desk plugged into a G3 Mac running Cubase, operating via mLAN with mLAN OMS and ASIO drivers for Cubase. Emagic showed Logic Audio interfaced to a Yamaha 03D mixer at Frankfurt this year, while Steinberg showed Nuendo on the Mac interfaced to an 03D at the recent AES in LA, in both cases using mLAN ASIO drivers. Though apparently, as at this writing, neither company has officially signed a licensing agreement. Nor have there been any other official announcements of licensees or products, although Korg has become the first third-party manufacturer to bring out an mLAN-capable product, in the form of its new Triton-Rack synthesiser and sampler expander module. This features support for an add-in mLAN board (basically a rebadged Yamaha mLAN8E board). However, there's no pricing or availability information as yet. Yamaha itself has several mLAN-ready devices, in the form of the CS6x, CS6R and S80 synthesisers and the A4000 and A5000 samplers, and the 02R and 03D mixers, having internal support for mLAN and ready for a board or card that adds mLAN functionality and ports.

The aforementioned capability to transfer multiple channels of audio and MIDI data bidirectionally over a single cable, coupled with the support for any-to-any device communication, is going to have a major impact on cabling and cable suppliers. Yet while mLAN has the potential to create cabling heaven for users, by removing the physical traceability of existing routeing it could introduce a conceptual confusion all its own. Much will depend on the interface(s) that facilitate what is in effect virtual cabling (the physical IEEE 1394/mLAN cables can be thought of as conduits carrying these virtual audio and MIDI cables). The mLAN8E, mLAN8P and Korg's EXB-mLAN all come with patchbay and mixer software. However, going by available screenshots, the patchbay software is lacking a graphical display of devices and cabling. With its support for the FireWire bus, and the Mac's strong presence in the creative community, perhaps Apple will even implement native support for mLAN in the Mac OS. As mentioned, mLAN replaces physical audio and MIDI cables with virtual cables. So fittingly mLAN's Connection Manager capability uses virtual connectors called 'mLAN plugs', and establishes virtual, or logical, routes between these input and output

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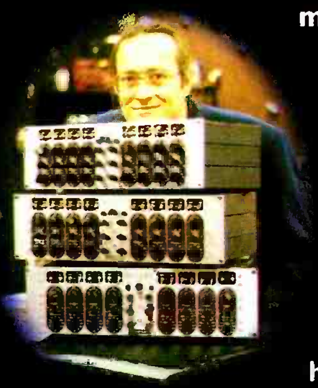
GOT THE PICTURE?



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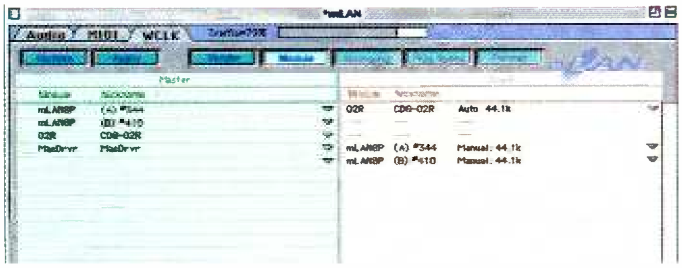
plugs called 'mLAN connections'. Ever mLAN device has Connection Manager functionality, and can create mLAN connections in response to a request from another node. It can also provide its connection data on request, and when the bus is reconfigured due to a bus reset or power-off an mLAN device can automatically restore connections. The Connection Manager functionality can even restore connections to devices which are disconnected and then reconnected to the network, providing another logical connection hasn't been made in the mean time, and can identify unique devices by ID, so two units of the same equipment model will be seen as different devices.



Another aspect of mLAN is the FS Manager, which manages the master-slave relationship between word clocks on the various connected nodes. The master can be assigned manually or automatically (the latter on the basis of output stream connections), with options for group assign in each case (the non-master nodes are automatically set as slaves). If a master device is removed from the network, audio on the slave devices will be muted, as they can no longer receive clock data, so a new master device will need to be defined.

Korg's Triton-Rack provides an example of what we can perhaps expect from 'first generation' mLAN instruments, in that it will replicate the analogue output functionality of the Rack—left and right stereo and four individual outs, plus 16-channel MIDI In-Out. The Triton-Rack is actually 16-part multitimbral in Multi mode, and while physical and cost constraints limit the number of separate analogue outputs the carrying capacity of mLAN knows no such constraints. However, to send up to 16 audio channels over mLAN from the Rack would require a change in architecture along with additional processing power that apparently isn't available. The other issue is that the current mLAN Packet Handler chip, the YTS-434-F or PH1, handles only 8-in/8-out audio channels at up to 24-bit 48kHz (although by cascading two or four chips in a device 16-in/out and 32-in/out can be achieved). PH2-generation chips, due in Spring 2001, will be able to handle 32-in/32-out channels at up to 24-bit 48kHz and 16-in/16-out at 24-bit 96kHz (cascading chips within a device enabling 64-in/64-out and 128-in/128-out). On the MIDI front, the YTS-434-F handles two MIDI-conformant sequences (two sets of 16 MIDI channels, or two MIDI ports). The next-generation NC1 (Node Controller) chip will handle 8-in/8-out audio channels and 4-in/4-out MIDI ports-cables if used by itself. However, the NC1 will also be usable in conjunction (linked) with the PH2, and if only the PH2 is used for audio then the NC1 will be able to support 8-in/8-out MIDI ports/cables. Like the PH2, the NC1 chip will also be cascadable for more ins and outs.

Another issue is data transfer rate. The IEEE 1394 protocol currently supports 100Mbps, 200Mbps and 400Mbps transfer rates, also known as S100, S200 and S400. First-generation mLAN supports S200. Second-generation mLAN (the NC1 and PH2 chips) will support S400, and proportionately more audio and MIDI channels. However, IEEE 1394 is headed for even greater data transfer rates—800Mbps (S800) on up to 1.6Gbps (S1600), with even S3200 envisaged in the future. This means that over time mLAN will be able to offer greater and greater multichannel capability as the IEEE 1394 capability grows. The P1394b working group is in charge of developing the spec in this direction, also increasing the supported cable length, and as such is implementing IEEE 1394 on a variety of physical media.

The recommended cable length with the current standard is 4.5m. For the S800 and S1600 rates, new Beta mode copper and Glass Optical Fibre (GOF) cabling will be used, offering cable lengths of up to 4.5 metres and 100 metres respectively. Cheaper 100m options at S100 and S200 rates will be Plastic Optical Fibre (POF) and Hard Polymer Clad Fibre (HPCF) cabling. Another option will be Category 5 Unshielded Twisted Pair (UTP) supporting S100 over a distance of at least 100 metres. Home-networking applications have obviously been a driver of the demand for longer cable runs, but of course there are potentially advantages for studio and stage applications, not to mention sound reinforcement installations. However, only the Beta mode option with its so-called 'bilingual' capability will provide compatibity with current connectors, hence bridging capability will be required, and this apparently is planned. Bridging will also allow existing networks to be linked, increasing the number of nodes from the current maximum of 63 to a possible 1023 x 63.

The fact that mLAN is built on IEEE 1394, and all that that brings with it, is a major point in its favour, not least because over time it will benefit from developments going on in other aspects of the IEEE 1394 protocol, such as data rate and distance increases. Longer-term developments involving hooking up IEEE 1394 and mLAN to the Internet or to ATM networks also hold out promise, and there could be a fruitful connection between Yamaha and the fast-emerging Rocket Network on this.

mLAN's long development history and the fact that its underlying audio and MIDI data transmission protocol is now an IEC specification also stand it in good stead. And the fact that it's a networkable interface gives it plenty of sophistication and flexibility. Yamaha's wise decision to provide royalty-free mLAN licensing will also weigh in its favour.

The real test of mLAN will come once it gets into everyday working environments, which of course means mLAN-enabled hardware and software being available. As mentioned earlier, this should start to happen before the end of the year. Only time will tell if mLAN can handle whatever gets thrown at it, and if the advantages in physical cabling simplicity won't be offset by disadvantages in virtual cabling complexity. Other major real-world factors which will determine mLAN's success or otherwise are of course the extent to which it's adopted by other manufacturers (which, as always, has an industry-political dimension to it) and the extent to which support for legacy devices is made available.

It's tempting to think of this time as a defining moment, like the introduction of MIDI 17 years ago, in which case we can only hope that manufacturers show the same farsightedness that they did back then in collectively supporting a new standard. □

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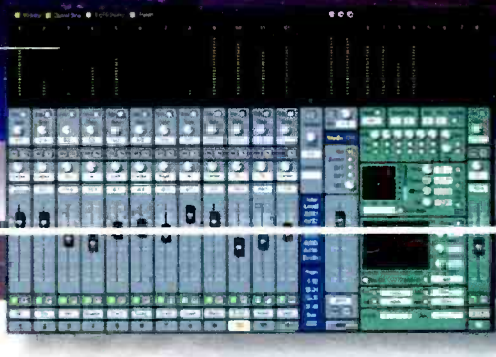

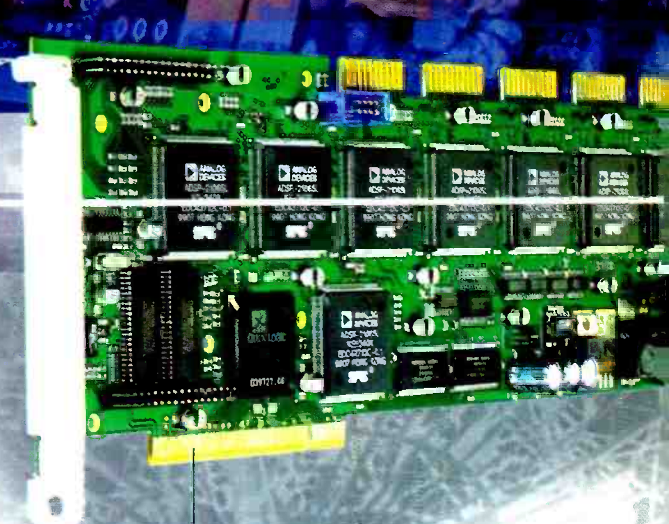

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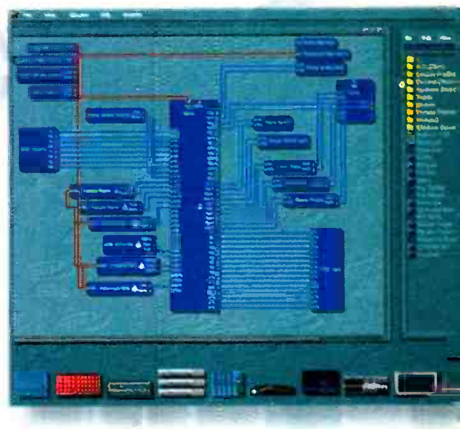

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REALITY BITES

With 'reality' television programmes commanding exceptional viewing figures, the teams behind *Big Brother* and *Jail Break* are in demand. **Neil Hillman** talks surveillance and success

BY THE 50TH YEAR following his death, the fictional masterwork of British author George Orwell had missed its predicted fruition date by some 16 years. For some, the summer of 2000 was when the nightmarish scenario of Orwell's *1984* was not only witnessed but embraced and taken into the shops and houses, hearts and minds of a previously-believed discriminating populace, in the guise of the television programme *Big Brother*. By others it has been asserted—fairly, judging by the viewing figures—as the time when the power of television was restated. Orwell's description of these viewing masses would no doubt be 'proles' and his images of life in a 1984 London leap from the page following the programme's nationwide poster campaign suitably synchronous with an Orwellian account of 'Pictures and posters on every corner, reminding citizens that Big Brother is always watching them'; and a prediction by the author that 'The Lottery, with its weekly payout of enormous prizes, was the one public event to which the proles paid serious attention'.

Big Brother is the bastard son of a game show, and one of several in the new-genre of 'reality television'; spawned from a desire to alchemise a new formulation from the established or known elements of television programming but at a fraction of the cost and content of previous offspring. The premise is this: 10 selected contestants are thrown together in a house for nine weeks, covered by 24 cameras and denied contact with the outside world. Inside the windowless house—a specially constructed bungalow in East London—the occupants were under constant scrutiny. Through the coverage of microphones and cameras within the building, every part of their incarceration could be monitored and broadcast six nights a week on television or live on what became the web site with the most hits in the UK. Each week, the inhabitants would vote to evict two of their fellows. These two candidates were then placed into a television ballot with viewers voting for their choice of loser. Each week the numbers reduced by one, until the final three contestants were put to a viewers' vote as to who would collect the £70,000 (UK) prize. The formula was not unique to the UK. The programme format first aired in Holland in the Autumn of 1999, gathering 70% of the viewing total along the way to the final. The Dutch people tuned in in greater numbers to only one other programme: the declaration of war in Kosovo.

The *Big Brother* format has also been sold to most European countries—retaining its anglicised *Big Brother* name—as well as to the US, New Zealand and Australia; although the Spanish have chosen to rename the show *Gran Hermano*.

In the UK, the programme was produced by Bazal Productions, through its association with its Dutch parent company Endemol. Pronounced not to rhyme with banal, Bazal is the UK's largest independent producer of factual entertainment. In its associated publicity, Bazal states, 'The year 1984 has come and gone and, fortunately, the world bears no resemblance to his nightmarish vision of life under the microscope'. Well, some might refer back to the scenario painted by Orwell in his novel, where the ordinary people '...were only concerned with the care of home and children, petty quarrels with neighbours, films, football, beer and, above all, gambling'.

Interestingly, Bazal's other premier credits include: *Changing Rooms* (home), *Ground Force* (home), *Ready, Steady, Cook* (home), and the prime-time Saturday evening slot for *The National Lottery Show* (gambling). The programme then, in the words of the producers, offered viewers a great insight into the human condition. Or as it's modestly put, 'The result was part documentary, part soap opera, part game show and part nature programme. Altogether, these elements combined to make *Big Brother* the most fascinating, compelling and ground-breaking series to hit British television in years'.

Louisa Knight and Robin Delwiche seated at the Soundtracs DPC-II 48-fader console

What was undeniably fascinating, and possibly ground-breaking, was the way in which the sound coverage was achieved within the house. The Outside Broadcast company, Roll To Record, supplied all the production technical facilities and crew for *Big Brother*—24 cameras, 10 remotely controlled hotheads, six Beta SX recorders, AKG, Sony and Sennheiser fixed mics, the '*Big Brother* voice' PA system in all rooms and garden, and three Tascam DA-88 multitrack recorders; all fed via their Calrec 48-input audio mixer. Peter Webber, Supervising Engineer from Roll To Record, was there at the start.

'The whole project was planned over a very short time scale, further truncated by the late completion of the house, giving us just two weeks to install the entire technical rig within the house and the gallery, 200m away'. He says that much thought was given to the sound and particularly the radio mics: 'We took advice from the production team responsible for the Dutch version of *Big Brother*. They initially used a bespoke system that caused quite a few problems before switching to a Sennheiser system, which worked much better.'

Webber also called in London hire company Cine-Video to assist in the equipping of the radio microphone channels. Although the *Big Brother* house was equipped with static AKG, Sony and Sennheiser microphones, actual conversation was very hard to pick up unless each contestant wore a radio microphone at all times; in fact the only time the microphones were allowed not to be worn was when they slept.

Robert Miles, head of audio at Cine-Video and himself an ex-sound supervisor, speaks the same language as his clients: '*Big Brother* was an interesting one for us. We knew the Sennheiser radio microphones would be fine—it was just a question of where to put aerials. Because the site was quite large, we supplied a number of different ones, ranging from co-linears to ground planes, all via distribution units to the receiver racks. The Sennheiser SK50 transmitter batteries were changed twice a day by a nominated contestant'.

Webber's instructions were clear: 'All contestants should be heard without interruption wherever they went within the house or garden'. He specified an aerial triax pickup system and a high number of channels—one for each of the 10 contestants plus spares in case of breakdowns. 'An important choice was that of the microphones, because it was apparent that distinct, clear sound would be vital to the overall success of the show.'

The mics used with the Sennheiser SK50s were DPA 4060s, a favourite choice of Cine-Video's Robert Miles. 'The 4060 is a lovely sounding mic, often used on classical instruments,' he confirms. 'In fact, the 4060

Above: In front of the winning target. Left to right: Robin Delwiche, Steve Williams, Phil Gibbins Andrew Ferrans and Keith Nixon. Right: Left to right: Steve Williams and Keith Nixon (Sound Moves)

capsules are increasingly being used with the Sennheiser radio microphone system because they are small and of very high quality. Their ability to withstand rough treatment was particularly important on the *Big Brother* project because they were being used by untrained people who had no previous experience of handling mics—we saw transmitters being swung around by their mic cables. In all we had to have a total of 27 repaired during the course of the show.'

All of the Roll To Record gear faced a punishing schedule. All of the radio mics were recorded clean onto DA-88s 24-hours a day for postproduction, with two audio TX mixes and two ISO cuts running throughout onto Beta SX recorders and monitoring on Genelec active speakers. The 48-input Calrec M-series desk was refurbished by Calrec specifically for the show, and all of the audio put down on tape was mixed through it. The radio mics predominated—except when the contestants were in the house jacuzzi.

The audio personnel faced a daunting schedule too. Two sound supervisor's mixed for 18 hours a day with an overlapping third person, while another sound supervisor sat in for six hours nightly. During this time various isolated feeds were required either for editing or to accommodate the web streaming requirements, and these were fine-tuned progressively during the first couple of weeks. Once the number of contestants in the house started to go down, the desk itself had to be reconfigured every week.

As the *Big Brother* house was a closed environment, Peter Webber and his team had to rely on the contestants to attach their own microphones each morning, and to operate a rota system to ensure that their batteries were changed every day.

'They were given basic training before the series started, then they were asked to wear them at all times except when they were in the shower or in bed,' he explains. 'As a result we had quite a few breakages, usually caused by a contestant being too rough with the microphone cable. When that happened they didn't always realise that we'd lost the signal, so we alerted them via the *Big Brother* PA system and got them to swap the damaged mic for a new one.'

Oliver France, one of the team of Roll To Record sound engineers who worked on the programme, adds that the male contestants were much better than the girls at remembering to wear their microphones and keeping them in good working order. 'Two of the girls, Mel and Claire, were forever forgetting to put their microphones on, which meant we had to rely on the static spot mics to pick up conversation until we could alert them to the problem. The boys, however, devised a system of looping the cable around their necks, which helped reduce much of the damage.'

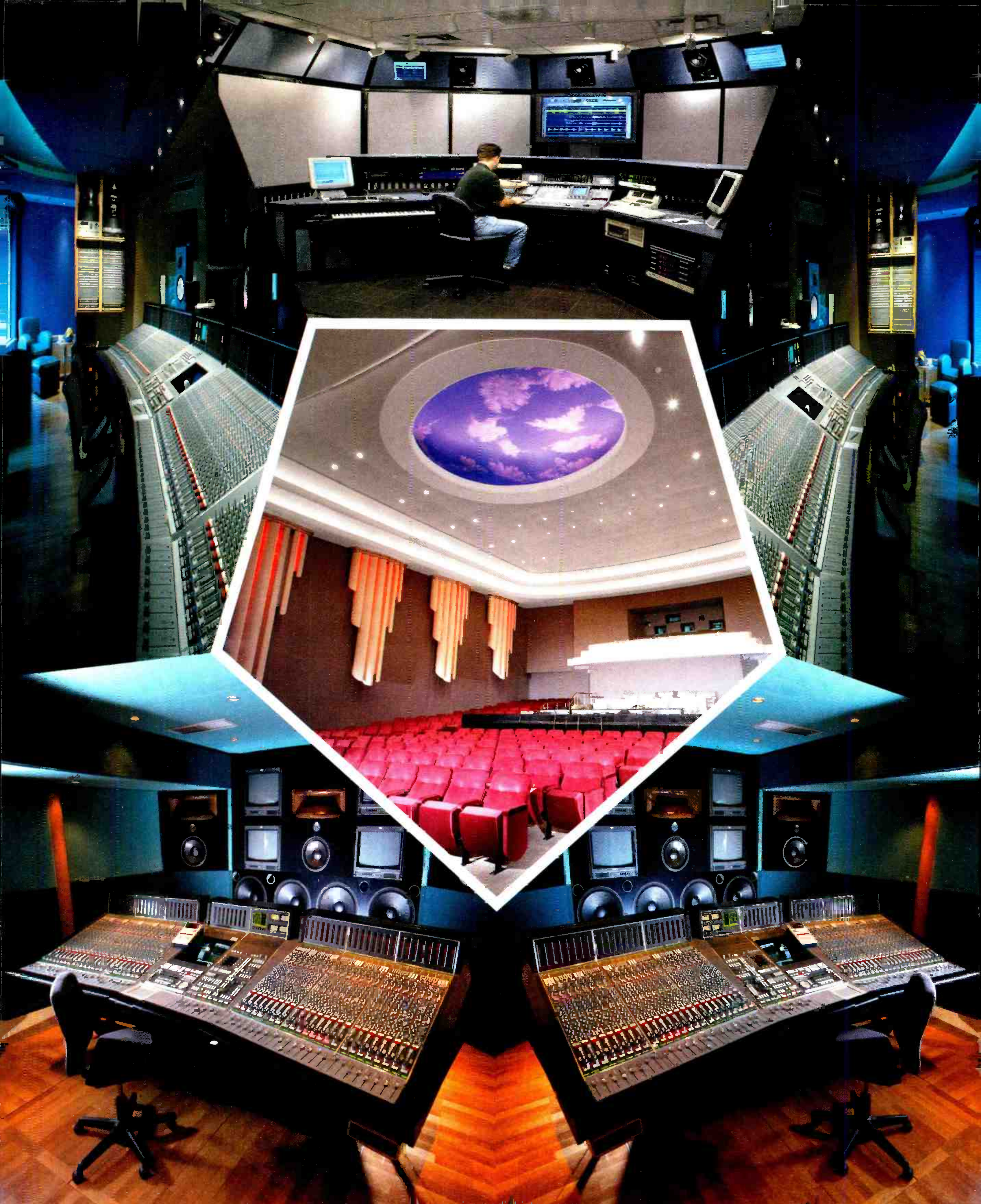
Due to the success of the *Big Brother* sound system, Cine-Video was asked to supply another multichannel radio microphone system—this time to another reality TV show called *Jail Break*. This attempts to emulate Channel 4's ratings success by drawing audiences to the fledgling Channel 5. Princess Production's *Jail Break* involves 10 contestants who are challenged to escape from a purpose-built high security jail. Their only contact with the outside world is through emails from the public who could send in suggested escape plans. The first contestant out wins a massive cash prize, and any member of the public who suggested a workable escape route also wins money.

Steve Williams, head of Sound Moves, handled the show's audio design, supply and personnel requirements. As for the *Big Brother* team of sound engineer's, his task was clean coverage.

'It was really important that we had good, reliable sound so we chose the Sennheiser system. Each of the 10 contestants was miked up—as were the presenters and prison staff. Of course with any project like this, the back up from the hire company is all important. The ability to replace damaged items quickly and efficiently is vital.'

Steve and his colleagues used Sound Move's new Soundtracs DPC-II 48-fader console for their programme sound requirements. He was suitably impressed: 'Our task has been to enable the viewer at home to hear all the inmate's and officer's conversations, both shouting and furtive; a real test for the dynamics of any system. Using the Soundtracs DPC-II we have been able to use the superb dynamic range offered by the 24-bit convertors to mix the 60 fixed microphones and 30 Sennheiser radio mics. With this number of sources the only way to have the correct microphone available under your fingers at a moment's notice is to use the snapshot facility. This is fired in turn by another computer that interrogates the video matrix and switches the correct mix to the two video-directed cut-streams and the five web-streams.'

With each incarnation of the reality genre, a new complexity of stealth and techniques evolve that push further that which is possible and erode that which is prestigious. And so I ask, Who watches the watchers, protects the protectors? □



THE MAGIC PENTAGON

As stereo challenged mono, surround sound raises the curtain on a new soundstage. **Steve Parr** shares the lessons learned from ten years of mixing music in surround

SHORTLY AFTER COMPLETING the building of London's Hear No Evil recording studio in 1990, I was called upon to mix the music for a German cigarette commercial in Dolby Stereo. Andy Day from Dolby came over with their SEU4 and SDU4 encode-decode units and helped me position my spare NS10s for the job. The final slice of the cake was the ritual of setting the levels with the infamous Tandy audio level meter. Once Andy had left, the real fun started, and within an impossibly short time I was hooked forever on surround. Ten years, and some 500 mixes later, I'm still in there and still learning as the goal posts keep moving from LCRS through 5.1, 6.1 and 7.1 to 10.2.

Although mixing music in surround formats is in many ways easier than mixing in conventional stereo, there are many issues to be aware of. First, there are the different surround formats.

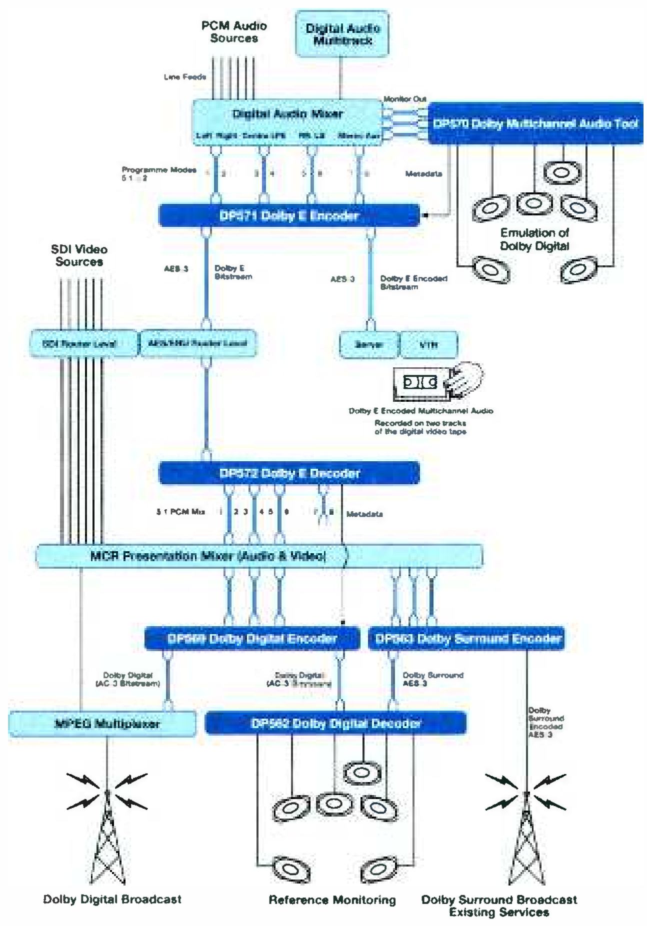
Dolby Stereo is a matrixing format that relies on phase differences to encode and decode channel information. When mixing for Dolby Stereo you will generally supply your mix as a 4-track LCRS music stem, but you must monitor through encode-decode hardware so that you can hear the unpleasant effects of the processing. Unless you mix very wide, the matrix will not see sufficient phase difference and will tend to collapse the audio into the centre. Loud bass-heavy signals placed in the centre will also have the same collapsing effect. The key factor is this: just because you place a signal in a specific channel does not mean it will be reproduced in the same position after the encode-decode process.

Even if you get a perfect surround music mix through the matrix, there may be additional sound effects and dialogue added at a later stage that will scupper your best efforts. The consumer version of the matrixing process is Dolby Pro-Logic; all home-theatre amplifiers have it and you should be aware that people rarely turn it off. If you are working in stereo, you should check your mix through a Pro-Logic decoder to hear how approximately 100m people will also hear it. And because Pro-Logic works on any stereo signal, this means ALL formats.

If you have mixed and monitored through the matrix, you should label your master Lt&Rt to signify that fact.

Dolby Digital uses perceptual data compression (about 12:1) to encapsulate the information into an AC3 digital datastream. The inputs and outputs are generally 16-bit, 48kHz and as such it is a very lossy codec and not ideal for situations where the music will be heard under a high degree of scrutiny. Quality aside, the advantage is that the surrounds are in stereo and placements will be maintained. And, of course, you have a separate sub-bass channel.

Digital Theatre Systems (DTS) is a 5.1 format that is similar to Dolby Digital, however, the data is stored on a synchronised CD and so the compression is far less severe (about 4:1). DTS applies to Red Book CDs, and so it is possible to play back DTS 5.1 20-bit encoded CDs on any CD deck with the help of a decoder that takes its feed from the digital output. You can buy a cheap software encoder from a number of manufacturers so at last you can mix in 5.1, encode it, burn it onto a CD, and take it home to play it through your home system. DTS CD releases number about 200,



Creation of metadata from post to home

ranging from Steely Dan and the Eagles to mainstream releases. Whatever their merits, I consider them to be the Bible of surround mixing, and an essential tool to anybody wishing to get involved in the practice.

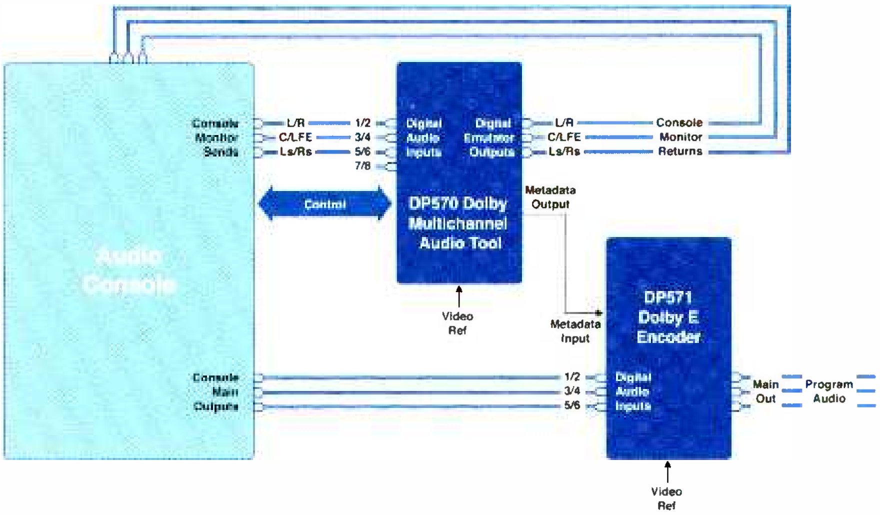
Sony Dynamic Digital Sound (SDDS) is Sony's 7.1 format, having two extra channels placed inside the left and right front. This is a cinema format and not really relevant to music mixing as yet. Finally, there are the Dolby and DTS 6.1 formats, where an extra rear channel provides centre surround, but these need not concern us for the moment.

Although most engineers leave mastering and final encoding to dedicated facilities, a knowledge of the issues involved is necessary. Decoders in home entertainment systems have the ability to read coded 'metatags' in the datastream which determine how the audio is interpreted by systems that, say, have no centre speaker, no sub bass, or just LCRS. The decoder downmixes the six channels according to the metatags. The algorithm can also apply dialogue normalisation (a type of floating reference level) and dynamic range compression for replaying film soundtracks at low levels. All these options are decided during encoding and have a great effect on how your final mix will sound on a domestic system. The home entertainment amplifier will also have an element of bass-end management for bandwidth-restricted monitors.

DVDs have a stereo LPCM soundtrack and an optional surround mix. You should be wary about letting an algorithm decide on how your 5.1 mix should sound in stereo—it is far better to do a specific stereo mix in the way that you always have in the past and you should do this if at all possible. If you can't go back to the original multitrack elements to mix in stereo, you can create a stereo mix from the six tracks on your 5.1 master by folding in the respective surrounds to the left and right and adding in the centre and sub until you get the right balance. However, you've probably spent many years perfecting the art of mixing for stereo and it would be unrealistic to expect a downmixed 5.1 to compete on any real artistic level. Many of the compression, equalisation and stereo placement techniques normally used become irrelevant because of the expanded soundfield. No longer are your elements jostling to be heard, they have space and dynamic range.

The bass management of home entertainment systems filter off a low-frequency element of all five channels, combine it with the '.1' channel, and then direct it to the sub-bass speaker. If a home system is without a sub, that information is generally ignored. It is therefore important that vital mix information is not placed solely in the sub or you risk losing it. The normal practice is for the bass, or maybe kick drum to be placed there, but never without having it in one or more of the main five channels. Historically, the sub-bass comes from cinema practice where all the front speakers are full-range and the sub was used for low-frequency enhancement (LFE) of explosions and high-energy effects.

I've tried many different combinations of monitors over the years, from having three full-range main monitors at the front with smaller monitors of the same type at the rear, to moving the rears to the side, and then substituting all the monitors to five of the same type. I've found that using large main monitors is too overwhelming; it feels like I'm drowning in black forest surround gateau. My current preference is for five close-field monitors such as Genelec 1031s, placed as recommended by the ITU, that is, at 30° and 110° from the front centre position. I occasionally move the rears further back for projects that require a more ambient mix, such as orchestral film scores that need envelopment without too much imaging. I'm lucky that I have a large control room and am able to have the monitors placed freely on stands. This also works for other engineers who have their own preferences for where they place the monitors. I've also set up a secondary 5.1 system in my live room with a domestic bass management system to get more insight into my mixes in the same way that you switch between different monitors when working in stereo. While a stereo mix has to sound good on any format, whether radio, CD, boombox, car or television, a 5.1 mix will only ever be heard on a home theatre system, or maybe a car with a central driving position.



A typical integration of a console and DP570 Dolby Multichannel Audio Tool

There is also a question of how you control the level of six speakers simultaneously so that they level track consistently. Although it is possible to set up a group of six faders on the returns from your 6-track master and then send the outputs of these to your monitors, it's both a clumsy solution and uses up too much console real estate. If you have three separate stereo monitoring buses that will each route to a separate pair of speakers you can do it this way, but calibration is awkward, and you're never quite sure if the three stereo outputs are tracking properly. The best way to tackle the problem is by the use of a surround monitoring device such as the Magtrax. The most basic units have six line-level inputs, a volume knob, and six amplifier outputs. More sophisticated units have one or more external inputs and switching so that you can choose to monitor either your console output buses or the outputs of your master 6-track recorder. Some form of output calibration for speaker amplifiers, and maybe a downmixing switch so that you can hear your 5.1 collapsed down to stereo are useful if only to prove to you that you're going to have to do a separate stereo mix anyway.

What is the function of the centre speaker in 5.1? Should you not use it at all, creating in effect a 4.1 mix? On the positive side, the centre channel can be used as an anchor in a way that you can never achieve with a phantom centre. If you move away from the sweet spot in a stereo mix, sounds placed in the phantom centre will also appear to move in relation to the L&R. By using a hard centre you can go a long way to stop this happening, making the image more stable when you shift away from an ideal listening position. It will also sound punchier and you will avoid the 2kHz dip that you get with a phantom centre due to non-coincident wave fronts reaching the ears. You can also go a long way to avoid the push and pull of conflicting frequencies in the same speakers—a kick drum could be placed as a hard image in the centre channel with a bass line as a phantom centre, equal in the left and right. On the down side, cheaper home systems rarely have three similar speakers in the front and a mismatched centre speaker could easily throw off your whole mix. It is far more likely that only the L & R will be balanced properly, thereby making it much safer to stick to phantoms. This is a situation where analysis of the mixes on the DTS CDs can give valuable insight to what works and what doesn't, but this is something that you really have to make up your own mind about.

You should probably be trying to create a large listening area with your surround mix, to provide a high degree of listener envelopment, and to provide a conductor's or an audience perspective so that the mix sounds great without you necessarily being aware that you're listening in surround until you hit that 'stereo' button.

Some music sounds best in a natural acoustic space. Classical and jazz are two good examples. To hear an instrument coming at you solely from the rear speakers in a classical recording is disconcerting, and in fact, sometimes downright annoying. What the mixer should be trying to recreate is a natural acoustic that envelopes the listener without distracting. The ideal perspective is that of the conductor himself, who has all the instruments wrapped around him in a semicircle; he hears most of the instruments directly, enhanced by early reflections and the general reverberation of the concert hall.

The other case is that of a pop record. Here, the instruments, if there are any, have generally been recorded on a piecemeal basis and there is little or no spatial information involved. The engineer has to create the illusion of space by use of reverbs, delays and processing. This has no basis in the real world and so the engineer has the freedom to be a lot more aggressive in the placement and use of dynamic panning in the 5.1 mix.

It is an interesting exercise to pan slowly from a front speaker to the equivalent rear. If you apply equal level to front and rear, the audio seems to be coming from 45° in front and not 90°, as you'd expect. If you keep panning towards the back, the sound then breaks up so that you can almost hear it as two discrete sources with slightly different frequency content, and then finally it will zip to the rear. This phenomenon is upheld by psychoacoustic research showing that the ear's frequency response to sounds coming from the rear is radically different from that of sounds coming from the front. This is in part due to the physical geometry of the outer ear and lobes affecting the frequency response of the ear canal, and it can be of help to get an assistant to gaffer back your lobes when mixing. I've also found that you can help this problem

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by slightly equalising the source as you pan from front to back.

How many times do you place source hard left or right when it isn't part of a stereo pair? Sources that are placed solely in one channel can sound obtrusive; you become aware of the positioning of the speaker rather than the positioning of the sound. It's more musical to pull instruments slightly into the room by placing small amounts of the signal in the other channels so that you almost feel you can walk behind them. This also helps to widen the critical listening position. The same principal extends to the use of reverb. It's good to use different reverbs in the different planes. For natural ambience, I tend to use the four outputs from a Lexicon as my master reverb. I pan the outputs to the four corner monitors, but bring in the front pair a small amount so that will be a very limited amount of return to the centre. Any close-miked instrument that I then send to this reverb will immediately get a context within the room. I then set up reverbs for the different planes according to the content of the mix. I've found that it's better to change the position of the reverb returns so they are in a different place from the source. It's also good to have different reverbs in the front and rears, and don't forget that there's nothing to stop you having a horn section, for example, panned between left and left surround, being fed to a stereo reverb that is panned hard right and right surround. A vocal in the centre could also have a delay in the surrounds, to create the feeling of the reflections of a large stadium.

Many engineers use their stereo mix as the basis for their surround mix either by starting to work on the surround while their mix is up on the board or by recalling their stereo mix subsequently. However, there are times when this is not possible. Sometimes it is impractical to mix the surround in the same session as the stereo, because of budgetary considerations or because the record company is not yet ready to commit to the format. In this case, a good solution is to lay off elements of the stereo mix to another multitrack format, preserving effects and dynamics that are integral to the sound of the stereo mix. Formats such as RADAR and Pro Tools are ideal for this purpose. The elements can then be archived for later retrieval when the time is right; in this way the integrity of the original mix can be preserved even if the 5.1 mix is undertaken at another studio by a different engineer. Typically, it will extend a stereo mix session by about an hour.

Another situation is where it is unfeasible to mix in the same room due to lack of suitable monitoring or limitations of the console itself. While its possible to mix in 5.1 on any professional desk, some lend themselves to the process far more easily than others. Digital consoles have a natural advantage because much of their functionality is software based, and it only needs the correct algorithms to mix in the various surround formats; even moderate priced digital desks like the Yamaha 02R and the Mackie D8b have a good implementation of surround. However, older analogue consoles are more difficult to configure without using up much valuable console real estate. On desks with more than one stereo bus it is possible to use one for front L&R and another for surround L&R with the centre and sub being addressed by auxiliary buses. But this is clumsy because it makes panning between front and back difficult and smooth pans through the centre speaker well nigh impossible. Dynamic panning is also tricky in all but a very basic way. I chose a Euphonix desk because as an analogue console under full digital control, it gives you full automation of every surround function and the ability to mix in over a dozen differing surround formats—useful for Imax and special event audio systems.

I mentioned earlier that in many ways a surround mix is easier than a stereo mix. You now have six speakers with the equivalent of a greater useful dynamic range so that you can get much better bass, more separation between instruments and a creation of space in the mix. When working in stereo, an engineer has to spend much of his time EQing and compressing to fit a large number of signal sources into a stereo perspective so that they can not only be heard, but that they are balanced without masking each other. And of course this perspective is in one plane only. Many of these problems evaporate when mixing in 5.1 because you now have a 3-D perspective in which to place your sounds. Although you now have four planes between adjacent pairs of speakers (front, rear, left side, right side) you can also bring sounds forward into the room so you can literally think of your space as a stage on which you can place the various instruments. You now don't have to EQ and compress just to pull something through on a mix, and strangely, even balance becomes slightly less critical.

Surround mixing is still in its infancy and there are limited outlets for surround mixes; to whit, music for film, concert remixes for DVD, and albums released on the DTS CD format. However, consumers who have bought their home entertainment centres with a wide-screen television and all the associated paraphernalia will quickly become accustomed to listening in surround and will expect to hear the same quality and aural spaciousness from music albums. Lately, after several weeks of mixing solely in 5.1, I had to go back to stereo and my sense of loss was palpable. Moving back to 5.1 was like coming home. □


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HOW TO SURROUND

The move from stereo to surround sound challenges many practices and conceptions. Studer's **Stefan Ledergerber** discusses modern mixing and virtual environments

SURROUND SOUND VASTLY INCREASES the spectrum of creative mixdown options. The extra channels can also be put to very different uses, some of which highlight limitations of the mixing technology that severely hinder efforts to create a high-quality listening experience.

There are presently two approaches to using the additional surround channels. The first uses sound source imaging to the rear and sides and may be considered an 'effect', to be creatively deployed by the recording engineer. The second is intended to create the impression of a genuine acoustic event, with so-called 'envelopment' giving the listener a heightened sense of involvement. Besides any other challenges found here, the recording engineer must establish a realistic-sounding envelopment.

This is achieved by feeding the surround loudspeakers with signals corresponding to what would be heard from a given direction—this includes crucial side reflections generated within the perceived space. The most natural way to create an envelopment is to use an array of five microphones positioned similarly to the five reproduction loudspeakers and route their playback signals directly to the corresponding speaker channels. The rear microphone pair picks up the surround portion, while the front pair picks up mainly frontal sound. This technique has proved successful for classical music recording in rooms with extremely clear and transparent acoustics. In most cases, though, the recording engineer wants —or is compelled—to create an artificial mix incorporating sonic corrections or even to overemphasise certain aspects. The extreme would be to generate a surround image from a multitrack recording of purely mono sources, where the challenge is to create the envelopment as non-artificial sounding as possible, and establish a genuine surround impression.

Let us take a closer look at the sounds reaching our ears. In order of arrival, these may be classified into: direct sound, early reflections and late reflections (reverberation).

The direct sound and early reflections are most affected when a sound source changes its location relative to a fixed listening position. The reverberative element remains virtually unchanged, since late reflections are already highly diffuse within the acoustic space. Generating all three components as faithfully as possible for each of the loudspeakers requires knowledge of the sound source position for the direct and early reflection elements.

Looked at another way, realistically integrating a monophonic sound source (like a single spot microphone or one track of a multitrack recording) in a surround image with envelopment requires generating the first two parts dependent on the panner position. The simplest place to achieve this is in the panner itself. Reverberation may be generated using an external surround reverb unit. Integrating the reverb unit with the mixing desk brings increased operational and automation convenience.

Surround mixes may be roughly classified as follows. The surround channels are used simply as effects. Although arbitrary and flexible, this technique is unlikely to deliver long-term listening satisfaction. Impressive surround effects may be generated using a battery of delay lines, reverbs and other effects units, with their outputs routed to the various playback channels but mixing is very time consuming. Surround music mixes are preferably made from material that already contains dedicated surround signals. In pop music, these are frequently derived from ambience microphones positioned close to the live audience. Classical or jazz recordings frequently use main microphones with rear-facing capsules to acquire signals for the surround loudspeakers. At the start of the mix it is clear what signals are to be routed to the rear loudspeakers. The problem, having established a rough mix, is surround fall-off as more monophonic signals are added. This is caused by a lack of envelopment, particularly the type created by early reflections with correct directional and timing characteristics. Simply expressed, the fewer mono signals a surround mix contains, the better the surround image. This means compromising between acoustic balance and the overall surround impression.

Ultimately spatial perception hinges on positional reflections, there's nothing new there. But using surround to create a better image of 3-dimensional acoustic space clearly means paying more attention to these reflections. For the mixing desk panner to function as an effective positioning tool, it must also take account of position-dependent reflections—as the Studer Virtual Surround Panning system (see sidebar) is designed to do. □

Virtual Surround Panning

VSP is a parameterised positioning tool for imaging a mono source using a 2-channel to 8-channel playback system. It offers the following independent advantages over conventional panners:

Generating early reflections within a simulated acoustic space, depending on the pan position. These reflections are reproduced from the correct direction, at the correct time.

Better directional imaging (left-right panning) by adding phase and frequency spectrum information to the customary amplitude difference between left and right loudspeakers.

The newest VSP version also provides for late reflections (reverb). These are delivered in de-correlated form, independent of the pan position, to two (2-channel stereo) or four (surround) loudspeakers. VSP in conjunction with reverb is a complete room simulation tool built-in to the Studer D950S.

Recorded Highlights

Grand Mothers Funk

This mixdown of a live concert recording involved a 40-track recording, Studer D950S console, four ambience microphones (no main microphones, two ambience mics close to the stage, two further away), and 36 monophonic sources.

A sound engineer performed the mixdown twice, with and without Studer's Virtual Surround Panning system. The startling difference between these mixes lay not so much in the individual instruments, but the envelopment generated from the separate mono sources. VSP engaged the listener, putting him in the thick of the musical action and emphasising the groove. Although only light VSP processing was applied to individual tracks, the result was a convincing surround experience

The Proms, Royal Albert Hall, 1999

This mix of a live classical recording used 40 tracks on a Studer D827 DASH multitrack and a Studer D950S console, with the main microphones arranged as a Decca tree, additional ambience microphones and numerous spot microphones for tone and balance correction.

The sound engineers began by establishing an enveloping surround image derived from the main and ambience microphones. Surround was perfect but the balance and tone of individual instruments was unsatisfactory. Spot microphones were then added to the mix, mostly panned onto the soundstage between the left and right loudspeakers. Balance and tone were now right, but the good initial surround was swamped by the 2-channel mix between the front loudspeakers. The logical corrective step was to increase the level of the surround channels and add a touch of ambience from the side between the front and rear channels. The surround loudspeakers were audible again, but in place of seamless surround were two separate sound images emanating from the front and rear speakers. Even with additional reverb treatment, the new surround image was significantly inferior to the original.

Starting from the previous mix, Virtual Surround Panning was activated in the spot microphone channels. First, the simulated room model was tuned to match the Royal Albert Hall as closely as possible (early reflections). Then, these reflections were subtly added to the respective prepanned microphones. Despite the spot mics, the surround effect returned; the concert hall became apparent, suddenly we were back in the performance, totally involved and enveloped. Furthermore, it was no longer necessary to compromise between the front soundstage and the surround effect. Subtle early reflections brought another bonus: the spot microphones could be inserted at the correct distance-impression in the sound image. It also eliminated the need to use a few external effects units, making the mix even more clear and transparent. The recording engineers compared this result with the previous, traditional mix. Opinion was unanimously in favour of the Studer VSP.

Monitoring. Opinions differ as to whether all monitors should be identical or if smaller rear-fields are adequate

Convenient screen positioning becomes a real issue once a centre channel is installed and can cause conflict in sound-for-picture facilities. Similar conflicts occur with control room windows

Acoustic design and planning remains the single most important consideration when gearing up for 5.1. Acoustic designers, and some loudspeaker manufacturers, can offer experience in balancing the limitations of the available space against your requirements and ideas

Gearing up for 5.1

Multichannel production puts enormous strains on the traditional 'stereo' mixing console. The ability to monitor a variety of sources and destinations quickly for comparison is essential for convenient working in multichannel. Routeing and panning is more complicated and impacts on the automation. Multichannel has become the realm of the digital multichannel capable desk, for obvious reasons

Doing multichannel properly and well can involve a significant investment that can start with the wiring and infrastructure of a complex, for example, and work up. Investment in training is often overlocked. Short cuts are possible and do serve to allow experimentation and familiarisation with the potential, but they're no long-term solution. Investment in multichannel is an investment in your future business

Outboard. The effects rack is not spared the ravages of the multichannel upgrade process. Existing equipment does not become obsolete, but a variety of multichannel devices are now available that take advantage of the extra channels particularly with regard to dynamics and reverb control

MEDIEN HAUS, the brainchild of owner Alfred Huff (seated right), has been two years in the planning and over 18 months in the building. Located just outside the historic city of Mainz, its work catchment area includes the many broadcasters in the region, most notably ZDF. With seven studios, mostly 5.1 and all equipped with Genelec monitors from 1034Bs to 1031As the studio complex is divided into two sections. Studio Tonmeister has five control rooms and a large live room connected to a projection room above while Film Up which has two Avid suites and several production rooms. Most studios feature SSL desks including one with an Axiom MT. All studios in the Filmup section have natural light and are north-facing to minimise glare on monitors from sunlight.

SIMPLY SURROUNDING

While most talk of surround mixing assumes high-spec, high-profile facilities, there are simpler, cheaper alternatives. **Jim Betteridge** offers a pragmatist's guide to broadcast surround

MOST DOMESTIC CONVERSATIONS concerning surround sound are coming around to Digital 5.1. If you compare its performance with Pro Logic, its analogue counterpart, the reason is clear: 5.1 is vastly superior. Although initially somewhat slow on the uptake in Europe, sales of DVD players world wide are growing fast with Dolby stats showing over 15m DVD players currently in homes. Only about 5m of these are at present connected to 5.1 decoders, amps and speakers, however, compared with over 50m homes with Dolby Pro Logic systems.

From the broadcast point of view, there are very few people listening live in 5.1 (Dolby quotes around 104,000 satellite-TV decoders sold). So in the short term, if broadcast sound is to encourage and be part of this growing consumer interest in Surround, Pro Logic is the main tool.

The perception of Pro Logic as very much an interim surround format makes it less attractive to some postpro facilities to invest in the equipment necessary for its production. To buy an analogue encoder (SEU4) and decoder (SDU4) from Dolby will cost about US$4,000 and provide no 5.1 facilities at all. To buy the digital versions, the DP563 encoder and DP562 decoder, will cost nearly twice as much, but will at least offer the facility to decode an AC3 digital stream into its 5.1 discrete components (decoder) and downmix discrete 5.1 material to an Lt, Rt pair (encoder). It still won't encode 5.1 into an AC3 stream, however, so there's still a sense of investing in old technology. On top of that, there are the extra amps and speakers.

We're not talking about a fortune here, but coupled with stressed dubbing mixers inexperienced in the format, the extra work involved and the current context of dwindling budgets and time scales, the attraction is not always sufficient. Hence only a relatively small amount of TV and radio is officially mixed and badged as being intended for Pro Logic reception. As a result, a significant number of interested dubbing mixers with sufficient leeway for experimentation, have been mixing for Pro Logic using only a domestic decoder and a bit of creative know-how on the mixing console. But before going into detail, a quick refresher on the nuts and bolts of the Dolby Surround process might help things along.

The original Dolby multichannel system (until a few years back known as Dolby Stereo now rather confusingly called simply 'Dolby'), was designed to encode a 4-channel (LCRS) signal into a 2-channel system while still maintaining left-right stereo and mono compatibility. The answer was to leave the left and right channels untouched and add the centre channel equally into the LR channels (at -3dB). The mono surround channel was then fed equally into the LR channels (again at -3dB), but with one side +90° and the other -90° out of phase, producing an overall 180° phase shift. The resulting 2-channel 'matrix encoded' signal is referred to as 'left total' and 'right total' (Lt, Rt). If an LtRt mix is played though a standard LR stereo system, the LR channels will reproduce unaffected, the centre channel will be heard as a 'phantom' centre and the surround signal will spread itself 'wide' across the stereo image. Mono'ing this signal will be the same as mono'ing any other LR stereo in that the difference signal, in this case the surround component, will cancel itself out.

Apart from LR level alignment, the effective cancellation of the surround channel in the front speakers is also dependent on the phase performance of the replay system. To mitigate the HF phasing effects issue 26

Solid State Logic

Todd-AO scores immaculately with SL 9000

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Marc Gebauer, supervising engineer for the scoring stage, and Kirsten Smith, studio manager, pictured at Todd-AO Scoring, Los Angeles.

SL 9000 J scores for Todd-AO

Todd-AO Scoring, a division of Todd-AO Studios, a premier audio post-production facility for feature film and television in Los Angeles, has upgraded its scoring stage with a 96-input Solid State Logic SL 9000 J Series SuperAnalogue™ console. The console was immediately employed to record a massive 123-piece orchestra performing the music soundtrack for the upcoming Jim Carrey feature release *The Grinch*.

"After extensive research, Todd-AO Scoring chose the SL 9000 J Series because of its incredible sound and SSL's capability to customise the console and the recall automation," says Marc Gebauer, supervising engineer for the scoring stage at Todd-AO. "We are using it for one of the largest orchestras we have ever recorded to handle the tracks for *The Grinch*. This is an especially interesting project as the composer has invented a lot of instruments for the movie so the music tracks can almost be thought of as special effects. The 9000 came through with flying colours, elegantly capturing the orchestra. The sound quality was immaculate."

The SL 9000 J Series at Todd-AO is a 72-input mainframe with a 24-input sidecar, yielding a total of 96 inputs with full surround sound mixing capabilities.

'We worked closely with SSL on customising the 9000 with extra mix buses and patching facilities that we specified," describes Gebauer. "We needed to add wiring support for the four mix machine panels, which are 32 channels wide, to accommodate the 33-bus mixing scheme we devised to handle the way we work. SSL did a great job working with us at every step to come up with alterations while maintaining that great 9K quality."

Of particular importance to the engineering staff is the recall automation on the 9K. The policy at Todd-AO is to document every show that comes in, which used to mean long hours of manually writing down all the settings. "I can't tell you how happy the staff is not have to do that any more," adds Gebauer. "The 9K is really well thought out and, because of the custom features we put in giving us extra facilities, we already see that it will bring in a lot of business. I am glowing over this console. Everyone is very happy."

Rising stars ...

Chris Puram is a rising star among that current crop of hot young engineers who adapt with ease to new formats and technologies. Best known for the hardhitting, punchy sound he's contributed to mixes for Snoop Dogg, DJ Quik, Tony! Toni! Tone!, and Queen Latifah, Puram is usually found ensconced behind the SSL Axiom-MT digital console at Skip Saylor Recording in Hollywood.

Although he's thoroughly modern in sounds and attitude, Puram came up through the engineering ranks in the traditional way, logging untold hours as an assistant engineer. And, although his biggest credits are in hiphop, he first moved up to the engineering chair doing heavy metal. Puram recognises that his experience in rock gives him an edge in hiphop. "My first freelance gigs were all in metal," he recalls, "with producer Max Norman who worked with Ozzy and Megadeth – stuff like that. I really didn't do much hiphop until I hooked up with DJ Quik. There are similarities; not so much musically, but in the attitude, and in the age group that the music is targeted to."

It was studio owner Skip Saylor who introduced Puram and artist/producer DJ Quik. These days, Puram is a client of HitMixers, Saylor's engineer management company. The two have a long history; Saylor gave Puram his first real studio job, and Puram became Saylor's first management client. The studio owner/engineer manager fit is a natural one for Saylor, and the concept has proved so successful that HitMixers now boasts a roster of six busy engineers.

"I've known Skip since I started in this business," explains Puram. "So I really trust him. It's a great relationship. He's really good at what I'm not – the business end of things. Having a studio owner for a manager offers some unique advantages. And, because Skip is also an engineer, he really understands my job."

Originally from upstate New York, Puram was a self-described "music-loving technical geek" in high school. When his guidance counsellor's wife recognised his potential and got him an entry into Woodstock's legendary Bearsville Studios, the die was cast. Sold on a studio career, he headed to L.A. Fate again intervened, and he hooked up with his future manager while attending a recording class taught by Saylor.

"Skip and an artist named Gary Taylor were teaching a class once a week at Saylor Recording," Puram recalls with a laugh. "They would purposely be hard on the students to show them what a real-life, difficult session could be like. I guess it was a pretty effective class, because nobody else stuck with it. I was the only one who kept coming back, and Skip ended up hiring me."

Puram honed his chops on the Ultimation-fitted, SSL 4080 G-plus in Saylor's Studio A; when Studio B installed the first Axiom MT on the West Coast, he was quick to see its advantages. Projects he's mixed on the digital board include songs for Queen Latifah, Tupac and Japanese superstar producer Tetsuya "TK" Komuro, as well as "Fine," Whitney Houston's current Raphael Saadiq-produced chartclimber.

"What I really like about the Axiom-MT," he comments, "Is that everything is automated, and everything is quick. I find that I'll fix things that on another board I would just live with, or try to fudge around. Things like vocal pops, which either weren't possible to fix, or which would have taken way too much time. In a track with a lot of them, you'd have to set up who knows how many channels, and split them off for before and after what you're trying to fix. Now, it's just a simple fact of rolling off a little bit, with EQ or a filter, when the offending word comes around. You can go off-line and really fine tune moves, or you can do things on the fly, as you think of them. It's easy, because it's digital."

Puram is also a fan of the MT's sound. "It's great. The low end is nice and punchy, and the top end has edge. It's great for what my clients want to hear today. When I first started doing this, people only seemed to care that there was a ton of bottom; now I hear the words 'crisp' and 'punchy' a lot more than just 'fat.' They still want fat, but they definitely want higher fi than a few years ago. A lot of my clients who regularly work on SSL 9000 J boards tell me that they love the sound of the MT."

Multiple remixes are a fact of life in Puram's world and he especially appreciates the MT's recall capabilities. "I just finished a project for 4th Ave Jones," he notes. "And they were very picky, which is good. They lived with the mixes for a bit, then wanted to recall and make some little tweaks. With this board's complete instant reset, it was a breeze to do. When you want to insert something, you just hold down a button on the channel you want to insert it in, and pick your piece of gear from the list. One day we did ten or more recalls and it was easy. That just wouldn't be possible on an analogue board."

Chris Puram

Chris Puram at Skip Saylor Recording

Most of Puram's clients still record to analogue 2-inch, and he mixes to both analogue 1/2-inch and to DAT. "I still mix to both," he says, "But a lot of times I like the digital better these days. Especially because with the MT, it stays digital to DAT. One of the advantages is that you don't need to go through the conversion process.

Although his hiphop and R&B clients keep him busy, Puram tries to take on rock projects whenever he can. "I'm happy to do whatever," he says with an easygoing smile. "I'm not looking to make a big change, but I like to keep my hand in on rock. I always want to be able to do different things."

Rock or hiphop, you can be sure that projects with Puram's name on them sound powerful. "People are usually surprised when they meet me after they've heard my records," he muses, "I guess they expect someone more wild or crazy looking." Puram should have no worries on that account: he's definitely "pretty fly for a white guy!"

First MT music mix goes platinum

Studio Arnold Mühren, the leading Dutch residential recording studio on the outskirts of Amsterdam, was the venue for the recording and mixing of 'Luid en Duidelijk,' the latest album from leading Dutch artist, Marco Borsato.

The album, the first project to be completed on the studio's recently installed Axiom-MT digital multi-track console, went straight to Number 1 on the Dutch charts and has sold more than half a million copies to date.

Owner Arnold Mühren explains, "We were looking for a console that would prove as good a long-term investment as our original 4K and the MT provided the ideal solution. We're a bit tight on space here, and the MT's 48-fader frame gives us 96 channels with more than 200 inputs to mix."

"Operationally, the console's great. Not only does it provide outstanding automation but it sounds amazing – with a level of warmth not normally associated with digital."

Mixer Patrick Mühren pictured at Studio Arnold Mühren

Three Avants for Crawford Audio

Crawford Audio, a division of Crawford Communications Inc.'s premier post-production operations, recently opened its new audio studio complex in Atlanta. The 11,000 square-foot facility is centred around three 96-channel Solid State Logic Avant digital consoles, which are fully integrated through an SSL Hub Router system. This facility-wide, all-digital network forms a cornerstone of Crawford's all new 135,000-square-foot site.

"We are very fortunate to have the opportunity to build a new infrastructure from the ground up to house the Avants," says Steve Davis, director of Crawford Audio. "As the technology and marketplace shifts, a studio usually ends up constantly adapting rooms to those changes resulting in a real hodgepodge of equipment. By having all three main studios centred on the Avants, we achieved a uniform digital nucleus capable of interfacing with existing and future recording/editing systems. As the Avant is a digital system, the console is kept up-to-date primarily through software upgrades. This allows exceedingly busy facilities like Crawford to use the console's hardware interface and resulting studio layout well into the future while remaining on the cutting edge of technology."

SSL Avant digital post/film console

▲ (L-R) Yves Jaget and Thierry Rogen © George Shilling

Mega Studios and Le Voyageur collaborate on MT marathon mix

Earlier this year, a crowd of more than half a million people gathered at The Eiffel Tower in Paris to witness an extraordinary concert from French legend Johnny Halliday. The concert was a one-off spectacular marking 40 years of success with a budget of more than 40m Francs and a display of pyrotechnics to rival those in Paris on the Millennium.

The live broadcast on TF1 was complemented with a stereo simulcast on RTL but this was just the beginning. Following the broadcast, old friends Thierry Rogen and Yves Jaget of Mega Studios and Le Voyageur respectively, had just five days to mix 25 songs for a CD and a VHS soundtrack which were due on the street exactly a week after the show. Luckily, they'd have a little more time for the DVD with 5.1 surround!

One of the many things Rogen and Jaget have in common is their preference for SSL's Axiom-MT digital multi-track console. As Rogen explains, "Mega Studios was the first to have the MT in Europe – the first to take the risk – and I love it. When Le Voyageur wanted to build a big digital truck, I thought it was a fantastic idea to buy an MT because it's definitely the best console."

During the concert, Rogen took the helm in Le Voyageur 1 beneath The Eiffel Tower for recording, while Jaget concentrated on the live broadcast from a second OB vehicle. On the day following the concert, Jaget and Rogen convened at Mega Studio. It was Jaget's plan to overcome a seemingly impossible deadline by starting the project in Mega's MT-equipped Studio B, then copying the setups to the second MT, already drawn up outside in the Le Voyageur mobile. The street was closed and the two consoles linked for the following five days.

As the mix progressed, SSL's automation proved to be a blessing as Rogen confirms, "My feeling is that there is nothing more powerful than this automation system on a digital desk. SSL I think, has a feel for it after so many years, with the 4K and the 9K, and all the work they've done with engineers and producers – they listen to us, even if we are French! For me the MT's sound is not just the best digital console in the world, it's the best console in the world. And I never heard an EQ like this in my life."

Completed on time, the mix was duly delivered for mastering and appeared in the stores just one week after the concert as planned. As expected, the album was hugely successful in France and the subsequent DVD is now awaiting release.

Yamaha get surrounded

Yamaha Epicurus Studio 1

Yamaha Epicurus Studios has ordered a second SL 9000 J Series console for installation in its Tokyo facility. The new console, an SL 9064 J complete with surround-sound option, joins a 64-channel 9000 J Series, already installed in Studio 1.

SSL Japan's Managing Director, Takeo Asano explains, "The demand for 5.1 surround-sound production is now growing steadily in the region. Fitted with SSL's surround-sound monitoring panel, the SL 9064 J's SuperAnalogue™ processing makes it an ideal and proven tool for the re-creation of high fidelity audio for DVD-Audio and Super Audio CD production."

Medienhaus Mainz opens for business

Studio Tonmeister of Mainz, Germany has now completed the construction and equipping of Medienhaus Mainz – an all-new media centre on the outskirts of the city. At the heart of this new all-digital facility will be an Axiom-MT and two Avant consoles.

Studio Tonmeister will occupy 10,000 square feet in Medienhaus Mainz where SSL's Avant digital post-production consoles are installed in control rooms 1 & 2. Both rooms are fitted with Genelec monitoring and fully equipped for 7.1 Dolby surround sound.

Adjoining a 1,500 square foot recording studio, control room 4 houses the MT digital multi-track console and is similarly equipped for Dolby 7.1 surround and Genelec monitoring.

All three consoles are interconnected by an SSL Hub Router which handles up to 2,000 channels of audio, a powerful and important feature for larger facilities needing to access different machine and control rooms. Any available tape machine or other connected devices can be accessed without manual re-patching, and also the amount of wiring required during installation is minimal.

▲ Alfred Huff, owner of Studio Tonmeister

Oasis Studios will set the standard in China

An SL 9080 J Series SuperAnalogue™ console has been purchased by Oasis Studios in Bejing. Dindae Sheena, Chief Operating Officer of YYYD Productions, explains, "We did a market study on the standard that was currently on offer in other private facilities in China – as we wanted to improve on what was available. We decided that the 9K was the way to go and Oasis will be the first private facility in China to own one."

Control room A (The Ocean Room) will house the SSL 9080 J console with monitoring by Genelec 1036As. The main 4,000 square foot studio – with stunning lakeside views – is large enough to house a 60-piece orchestra comfortably. Considerable attention has been paid to acoustics throughout, with design by Sam Toyoshima.

Sheena concludes, "Set to become the first large-scale commercial recording facility in China, Oasis Studios will concentrate mostly on working with artists in the Asia-Pacific region, with most of the focus on artists from Mainland China, Hong Kong, Korea, Japan and Taiwan. But, as a 9K equipped facility, our ambition is to join the global club of premier international studios and we look forward to working with artists from all corners of the world."

◄ Dindae Sheena, Chief Operating Officer YYYD Productions, pictured with SSL's Regional Sales Manager, Tim Harrison

Aysis airs in Manhattan

Thirteen/WNET Public Television uses an Aysis Air console in its new all-digital production facility located in midtown Manhattan. The console is the centrepiece of the main production audio room, servicing live and live-to-tape studio production.

Mac Privette, Director of Engineering at Thirteen/WNET is delighted with the station's choice. "This console is an operator's delight. Any operator can walk right up to the board and make it fly. The Aysis Air is also super reliable – which is critical in achieving the successful day-to-day production of live and live-to-tape programs. Plus, it sounds great!"

techno*file*

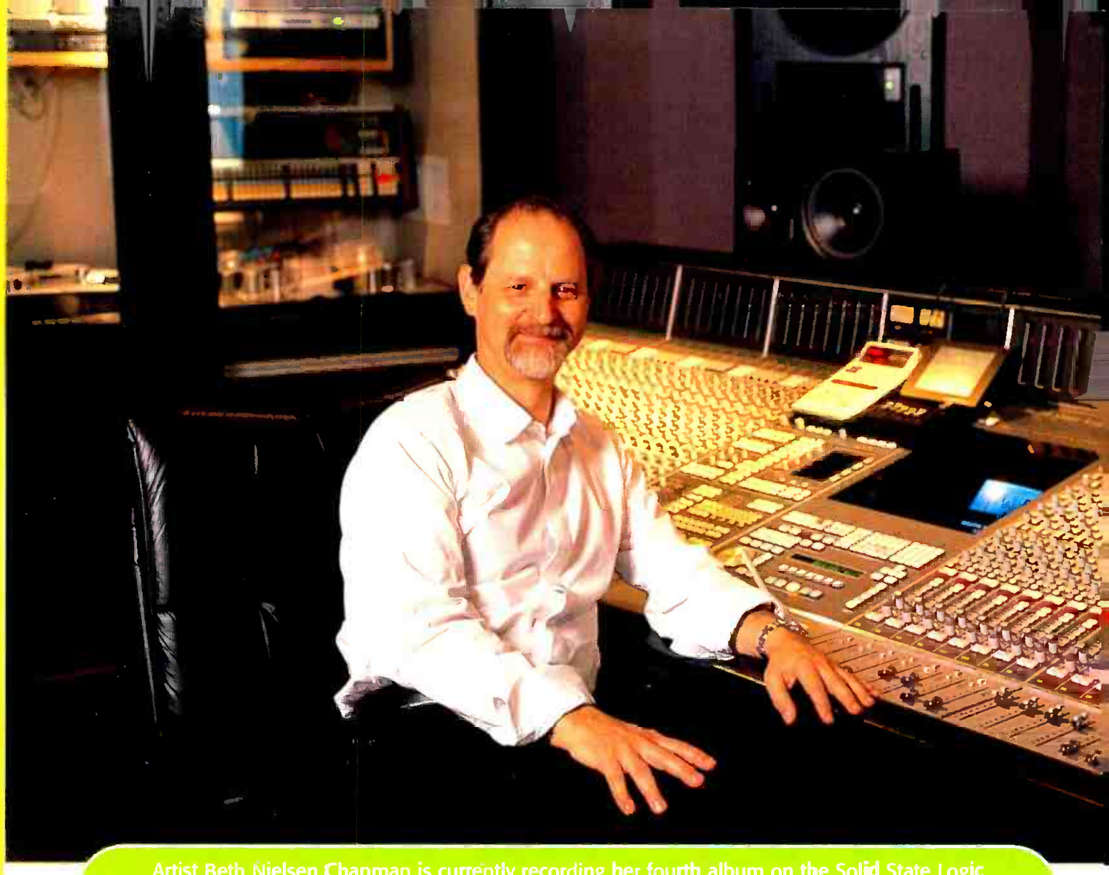
feel the difference

SSL's new digital faders for Axiom-MT were introduced at this year's AES Convention in Los Angeles and add a host of new useful features including:

- **Alpha-numeric message display (for grouping/automation and other useful information).**
- **Virtual detents (allowing the operator to feel null points/level matches).**
- **Individually hot-swappable for quick servicing.**
- **Digitally optimised accuracy and positional tolerance.**
- **Dedicated 'attention' button (for easy channel selection).**

The use of a custom linear motor fader allows electronic control of the fader's friction and reduces the use of moving mechanical components, virtually eliminating maintenance requirements. The fader's display allows the operator to instantly see the grouping or automation status of any individual fader. When using automation, the electro-magnetic clutch feature allows the operator to sense null points and level matches, giving physical as well as the usual visual clues to the automation data under the fingertips.

Niall Feldman,
SSL Director of Product Management



Artist Beth Nielsen Chapman is currently recording her fourth album on the Solid State Logic Axiom-MT console at Backstage Studio at Sound Stage in Nashville. Chapman, who helped write, along with producer Annie Roboff and Robin Lerner, the song 'This Kiss' for artist Faith Hill and will include a remake of the song on her new album, describes herself as "blessed" to be working on the MT. "The sound of the MT is fantastic," says Chapman, and "working with Chuck Ainlay at Backstage has been a great experience." Chapman, who co-produced several of the tracks with producer Tommy Sims, says that the instant reset of the console and being able to recall the mixes 100 percent has made the project very easy. Pictured here are (l-r) Chapman and Sims.

Artist Beth Nielsen Chapman Mixes on SSL's Axiom-MT at Backstage Studio at Sound Stage

CCTV install second Aysis Air

China Central Television, (CCTV) the state broadcaster in the world's most populous country, has taken delivery of a second Aysis Air digital broadcast console.

Last year CCTV installed its first 96-channel Aysis Air in a 9000 square foot studio at the broadcaster's main national headquarters in Beijing. The second Aysis Air, a 64-channel console, will be located in a 5,400 square foot studio within the same building.

The main reasons cited for the second purchase are reliability and the familiar, easy-to-use control surface. According to SSL's Regional Sales Manager, Tim Harrison, "CCTV has had an SL 4000 Series console for some time and has been looking forward to investing in high quality digital products. The broadcaster was able to judge the important reliability issue on their own experiences with SSL consoles – hence the second order."

Leading U.S. engineers love Axiom-MT

Eddie Kramer at Kampo Studios, New York

Claudio Cueni prefers working on the Solid State Logic Axiom-MT digital console at Skip Saylor Recording in Los Angeles above all other digital consoles. Says Cueni, "I've worked with other high-end digital consoles before, but the MT is the first one that produces a killer sound. This board is simply the greatest. I told Skip [Saylor], don't tell anybody how good the MT is so I can keep booking the room."

Leading engineer Derek Bason recently teamed with producer David Malloy and used the 96-channel Axiom-MT at Backstage at Sound Stage Studios in Nashville to mix two songs for artists Jesse's Girls.

"For SSL to build a console that's fully digital and sounds great is such a blessing – I don't want to mix on anything else," says Bason, whose discography includes projects for artists Reba McEntire, Wynonna Judd, Vince Gill, George Strait and Faith Hill. "I'm a big fan of the 9000 J and didn't think anything would ever impress me as much. But the MT is very much like the J in its automation, which is excellent. Many of its features are familiar to 9000 users, and it has the added dimension of being able to automate the EQs, pans and compressors, which is great. The flexibility of the MT will ultimately save a lot of time and money on projects because you no longer have to run a ton of outboard gear."

The SSL Axiom-MT has become the console of choice for many of today's top engineers across the U.S.A.

Legendary producer/engineer Eddie Kramer successfully mixed two hours of music in 5.1 surround on Kampo Studios' Axiom-MT in New York, with 22 songs mixed in both stereo and 5.1 surround being completed in only four days.

"The Isle of Wight concert in the 70's is considered to be the British Woodstock," Kramer explains. "We took Jimi Hendrix's entire two-hour show and combined it with biographical footage to produce a two-hour film performance on DVD. I suggested to Jimi's family that this was an excellent opportunity to mix in 5.1 surround and, since the tracks were already digitally transferred to DA-88 to facilitate the picture editing in the Avid system, the logical choice was to stay digital and that led us to the MT at Kampo."

(L-R) John Davidson, lead mixer, and David Wainwright, post-production mixer at Maryland Public Television

Maryland Public Television puts Avant to the test

"We've really been putting our new Avant through its paces," says John Davidson, three-time Emmy Award winner and lead mixer for Maryland Public Television. "We looked at other equipment and the Avant is the only console that could handle the way we work."

According to Davidson, MPT's engineers like the ability to be in different automation modes simultaneously, even within faders. "Normally, your faders are all in one mode," he says. "But on the Avant, each channel fader can be in a different mode individually. To a post engineer, that's pretty impressive."

MPT, a major supplier of long-format PBS programming, is one of the very few audio post production facilities that sends directly to a broadcast tower with their productions.

"With MPT, the audio in the studio is final," explains Davidson. "The Avant is completely user-friendly because the output matrixing allows us to hear the Dolby Pro Logic mix, the AC-3 mix and our Dolby E decoded mix. We can bit-stream through the console while encoding, allowing us to route that Dolby E encoded signal to any destination. With the Avant we can make sure all our encoding and decoding delays are in sync with picture. This is no small task and it will save us much time in the future."

Mike Post opens private recording studio

Well-known scoring and TV theme composer Mike Post will open a private recording studio built around a Solid State Logic SL 9000 J Series SuperAnalogue™ console with a surround monitoring system for 5.1 mixing. "Mike came to me and said I want to build the best of the best," says Paul Wight, chief engineer for Mike Post Productions. "We looked at many different consoles, vintage and new, and came up with the 9K as the only real choice. The 9K delivers all the desirable sound quality of a vintage board in a well-designed modern package. The 9000 is the best of the best."

newsbytes

Colin Pringle appointed Group Marketing Director

Solid State Logic Group Ltd., the holding company for SSL and all its subsidiaries, announces the appointment of Colin Pringle as Group Marketing Director, responsible for worldwide marketing and market development. Pringle, who was a director of SSL between 1988 and 1995, had most recently been with United Business Media, where he was responsible for Corporate Development and Marketing of the company's International Music, Entertainment Technology and Electronics divisions. He also played a major role in developing a range of 'e' business initiatives as part of United News & Media's $400 million investment in Internet technologies.

"SSL and its subsidiaries are uniquely positioned to leverage their digital and analogue audio expertise," comments Pringle. "The group's expertise ranges from digital audio encoding and communications to large-scale console design and manufacture. We aim to fully exploit this combination of skills to benefit the audio production community."

"This is an exciting phase of growth for SSL," said John Jeffery, Managing Director. "I am delighted that Colin is rejoining the group to help bring our plans to fruition."

John Andrews appointed Broadcast Development Director

John Andrews has been appointed to the new post of Broadcast Development Director at SSL.

Since his appointment as Marketing Director in 1997, Andrews has been responsible for the highly successful introduction of the 'A Class' digital console range, including Avant 'Focused on Film and Perfect for Post', Axiom-MT 'Made for Music' and Aysis-Air 'Born to Broadcast'.

"During this time, there has been increased activity in all key market sectors, but while SSL is naturally very strong in the music console market, with many of our staff having experience of it at the highest levels, the requirements of the expanding broadcast sector are different," said Managing Director John Jeffery.

"John Andrews is uniquely qualified in this area, with his operational and managerial experience at the BBC and his sales and marketing background in the audio broadcast industry," continued Jeffery. "I am therefore extremely pleased that John has agreed to take on the challenge of focusing parts of the company on this vitally important market sector."

Digital/Audio Alliance will serve SSL's German customers

SSL has formed an alliance with Düsseldorf-based Digital/Audio gmbh, the leading German systems integrator.

Explaining the new venture, Digital/Audio's Sales Director Rico Weber comments, "The Alliance will go far beyond the usual manufacturer/distributor relationship. Digital/Audio will help consolidate SSL's position in the German marketplace with its considerable experience in the field of studio integration. The combined skills of the Alliance now enable customers to enjoy a complete turnkey solution including consultation, planning, delivery and studio integration.

For this task Digital/Audio will activate its newly formed Digital Audio Network – a group of 10 independent companies working together on studio projects.

SSL's commitment to customer training will be helped in Germany by The Digital/Audio Academy, already well known for seminars led by Bruce Swedien, Elliot Scheiner and many others. A range of training seminars and workshops based on SSL consoles is currently being planned.

For further information:

Rico Weber, Sales Director
Digital/Audio SSL Alliance
Tel: +49 (0)211 - 737-7888

Game, set and match to MT

The latest MT digital multi-track console to be installed in France made its debut appearance at the French Open Tennis Tournament where Digital Road Runner – Euromedia's new all digital OB vehicle – supplied the video and audio feeds for the major U.S. networks.

With a fleet of six mobiles, Euromedia is also one of the largest studio groups in France with over 720,000 square feet of shooting stages in various locations. The specification for the new truck – which will be used for a mixture of live events, music and talk shows – demanded a complete broadcast solution, eliminating the requirement for a separate, dedicated sound truck.

The 96-channel MT console in Digital Road Runner is designed to cope with large scale sound production, with 60 remote mic amps on fibre. For its first outing, the OB with 10 cameras and full digital video capability was used to cover the country's premier international tennis event at the recently refurbished Roland Garros Stadium at Port de Auteuil, Paris.

Swiss TV station airs with SSL

Television Suisse Romande of Geneva recently completed its three-year search for a suitable large-format digital broadcast console when it selected an Aysis Air from SSL.

Installed in a state-of-the-art news studio, the new Aysis Air console has been specified with 48 mono channels in a compact 24-fader control surface.

According to Sound Supervisor Thierry Bonvin, a key element in their decision to select Aysis air was due to its "analogue-like" control surface. "We produce three live shows every day and we wanted to have full instant access of the parameters at any time during the show. The console also provides us with many functions essential for live operation including source auditioning, grouping and particularly easy to use EQ."

Claire Hall joins SSL U.S.

Claire Hall has joined SSL in the role of National Sales Manager, Broadcast Products, in the U.S. Hall will report directly to Rick Plushner, President of SSL Inc., and will be based at SSL's New York office.

Hall has an extensive background in electronics and broadcast engineering and operations gained in the U.K. with Granada TV and Yorkshire TV, working in all technical areas including audio post. She was most recently U.S. sales manager for Calrec Audio.

Rick Plushner commented: "We are very pleased to welcome Claire Hall to the SSL team. In this demanding age of HDTV and surround sound, her excellent reputation and vast experience in the area of professional digital audio for broadcast will further enhance our ability to respond to the needs of our customers."

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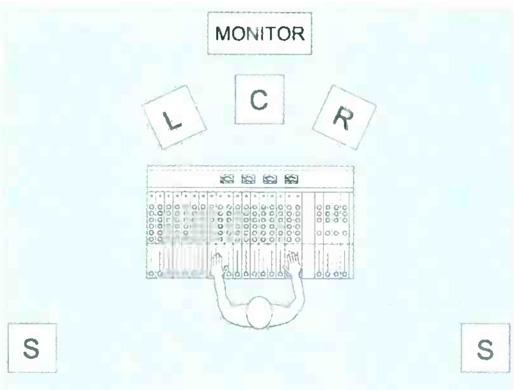
caused by the film weaving in the gate, the surround signal was limited to 7kHz and encoded with a modified Dolby B-Type NR; it was also passed through a 100Hz HPF to avoid the need for full-range surround speakers. For similar reasons (azimuth error on the tape machine) this additional processing stands today. A surround delay is set depending upon the listening environment to ensure that the front channels still arrive at the listening position first. Dolby A was originally used for overall noise, but has now been replaced with Dolby SR.

The first broadcast Dolby Surround system was almost identical to Dolby Stereo in terms of encoding, the main difference being the use of Dolby B, C or no noise reduction in place of Dolby A or Dolby SR. The Pro Logic system, introduced in 1988 is an improvement on the domestic decoder. It's main advantage lies in its use of an 'adaptive matrix' to increase separation to as much as 30dB between adjacent channels, much higher for opposite channels.

So with a reasonable understanding of the above principles, it is possible to mix for Pro Logic with only a domestic decoder and a reasonably flexible mixing console. The effect of the adapted Dolby B in the encoder is very modest and, because you're listening to exactly what the end user will hear (notwithstanding nonlinearities in the transmission chain), you can make a clear judgement on how things sound.

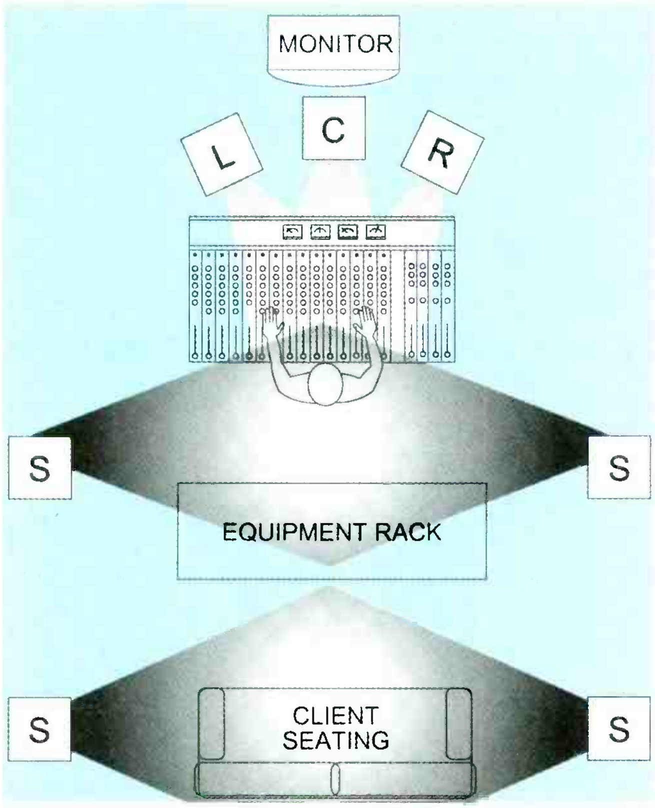
There are different ways in which to configure your desk to act as an encoder. Ideally it will have some kind of surround panning capability, as many of the new digital desks do. This will allow you to take a mono signal and, using some kind of rotary control, move it between group outputs one to four, corresponding to L, R, C and S, normally in that order. You now take groups 1-3 and feed them back into three other inputs on your desk routeing them to your main stereo output, panned left, right and centre, respectively. Then split group output four, feed it back into two more inputs, route one left and the other right, flipping the phase of one by 180°. The stereo output of your console is now, for most practical purposes, equivalent to an Lt, Rt signal. Connect this to the stereo input on your domestic Pro Logic decoder-amplifier (remembering it probably wants to see -10dB), connect up your speakers and, in principal, you're away. The amp should have a noise source with which to align the system ideally using an SPL meter, although a set of trained ears should get you very close.

A simpler way to create a surround element, if you don't have a Surround panning facility on your desk, is to simply split the output of an aux send back into two channels on the desk routed left and right with one side out of phase. This send now controls the amount of each channel in the surrounds. If you're using a digital desk, travelling twice through the system will mean a short delay between a sound sent to the front and to the surround channels, this can actu-

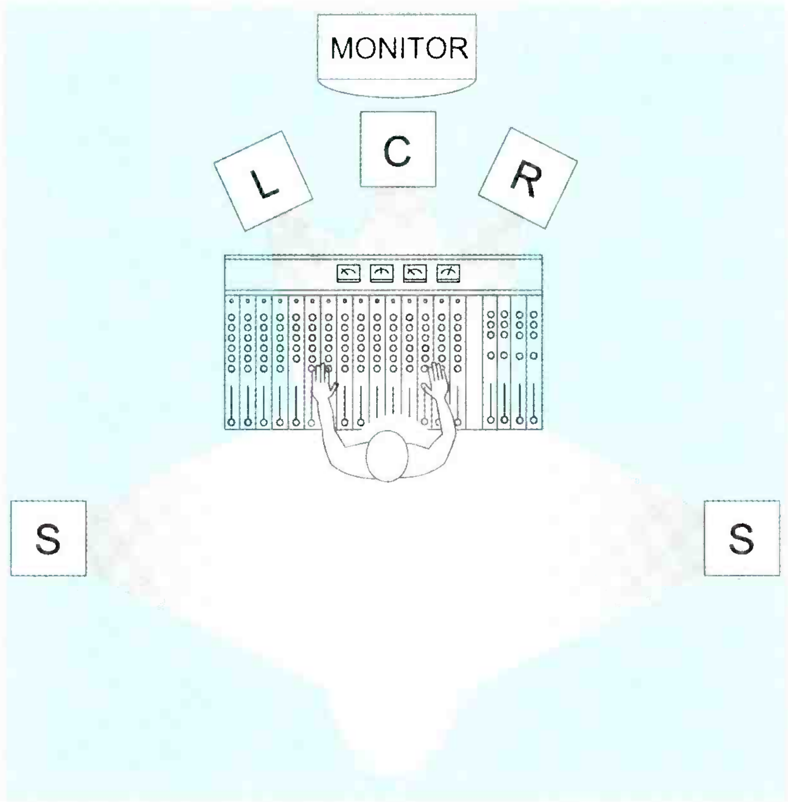


Terminology

Dolby: In cinema terminology this refers to the Dolby analogue multichannel (LCRS) format (see 'Dolby Stereo').
Dolby Stereo: Term originally used to denote the Dolby analogue multichannel (LCRS) format, now (rather confusingly) simply called 'Dolby'.
Dolby SR: A Dolby analogue (LCRS) multichannel mix using the superior Dolby SR noise reduction as opposed to Dolby A. Most films are made SR these days.
SVA (Stereo Variable Area): Also commonly used in cinema to mean the Dolby format, it refers to the two variable area optical tracks (as opposed to variable density) which are part of the Dolby analogue multichannel (LCRS) format specification.
Dolby Surround: The original consumer version of the analogue Dolby multichannel film sound format.
Dolby Surround Pro Logic: Launched in 1987, an improved version of the original Dolby Surround domestic decoder. Note that the encode process is unchanged.
Dolby Surround Pro Logic II: New, improved version of decoder launched June this year. See www.dolby.co.uk for more details.
Dolby Digital: Five discrete, full-bandwidth channels of digital audio (LCR + stereo Surround LS RS) plus a low frequency channel (20Hz-120Hz known as LFE).
AC-3: Dolby's encoding technology upon which Dolby Digital is based.



Four speakers arrangement



Two speakers arrangement

ally be an advantage—see John Bain's suggestion later in the article.

Though a centre speaker is important in the cinema to root the dialogue where you have a big screen and widely varying seating positions, for broadcast it isn't. For an average size control room where the dubbing mixer will be sitting more or less centrally, the phantom centre is just as good and indeed probably more mixers work that way than don't. Also, more home users probably discard their centre speakers than don't. The only thing is to make sure the decoder is switched to Phantom mode.

According to Dolby, the surround speakers should be mounted about two feet behind and at least two feet above the engineer's head. With the limited bandwidth and levels involved they can be fairly modest, although with the 5.1 future in mind, you may want to over specify.

It's important to constantly check stereo and mono compatibility as you mix. The domestic decoder will have a surround on-off switch to check for stereo and, assuming you are using its monitor output, the mono button on your desk will collapse things into mono. Switching to stereo sends what would be the surround channel to the front LR speakers with one side out of phase, resulting in a kind of 'wide' effect. In mono, the surround channel cancels itself out completely and so it's important not to send anything exclusively to the surrounds. John Bain, an outspoken supporter of surround mixing (with and without an encoder) suggests using stereo programme for your surround when possible, sending it to LR with one side phase flipped. This will mean that only the in-phase component will be present in the surround channel in Surround mode but the difference will remain in mono, rather than nothing at all. A compromise well worth trying.

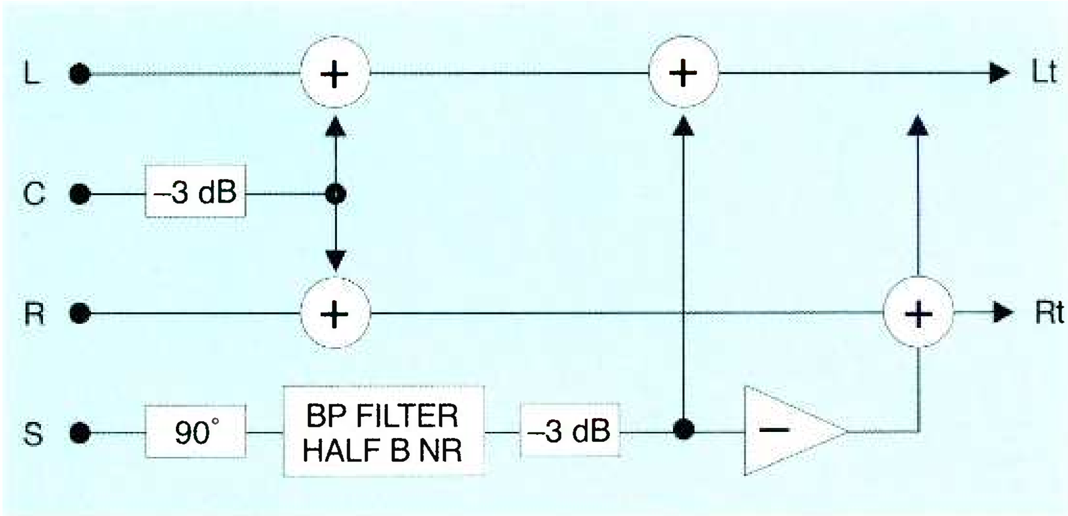
One of the arguments for the proper kit is that the Dolby encoder actually phase shifts one side of the Surround signal +90° and the other by -90°, whereas the method discussed here flips one side by the full 180°. This means that when you position a signal somewhere between front and back, you're actually adding it back with itself with one side out of phase. Normally this would skew the stereo image of that signal in the direction of the positive phase. A very effective way around this, again suggested by John Bain, is to introduce a short delay into the surround channel. You have to be aware of the possibility of comb

filtering here, keeping the delay short limits the problem to HF and below about 1.4ms, to above the 7kHz BPF. In practice more is sometimes needed, but, subjectively, it has surprisingly little adverse effect.

Very useful on the Dolby professional units are the arrays of buttons, LEDs and displays that allow quick cross checks, greater flexibility and reduce the need to know exactly where everything is routed and how. They're also all +4dB balanced or digital. The other consideration is that you can't advertise your programme as Dolby Surround unless you've used the proper equipment. You can, however, say that it is 'in surround'.

To do justice to a discussion on the subjective ins and outs of mixing in surround would demand several more pages and involve many differing opinions; which is exactly why it's important for more engineers to get the chance to experiment with it themselves. It is after all, in some form, the future. Monitoring through a decoder is always a good idea simply to be sure you know what people are hearing in their homes, whether or not you are intending to positively use the format. Once you've gone that far you might as well play around a little. After the vagaries of matrixed surround, the clean accuracy of 5.1 should be a breeze.

For more information about most aspects of this subject, download a pdf version of the Dolby Surround Mixing Manual from www.dolby.co.uk. There's also all sorts of practical discussion and training opportunities offered at John Bains website (http://members.aol.com/JBainSI/). I can also personally recommend the 2-day BBC course held at Wood Norton (Tel: +44 1386 420000) and would like to thank Senior Lecturer Alan Tutton for his input to this article. Happy experimentation. □



INTENDED AS THE CENTRE of a home hi-fi system, the average domestic Pro Logic decoder will be swarming with audio (and possibly video-RF) inputs and outputs, but being either unbalanced -10dB or optical digital, it probably won't be so much use for integrating into a pro environment. One hundred watts per front channel and 50W for the surrounds is common (only a preamp out is offered for LFE for AC-3 applications) and the noise and distortion figures on slightly more expensive models are very respectable. You can expect to find six separate line inputs for the 5.1 channels but less common are six preamp outs for connection to power amps of your choice. Also, look out for proper binding posts for the speaker connections as opposed to the rather shaky spring-loaded alternatives. Some manufacturers offer rackmount kits, some don't. Easy access to a surround on-off button is important, but pretty standard.

I took a look at the new Technics SA-DA10 which has a patented anti-vibration honeycomb base that apparently significantly improves audio performance—it certainly sounded good to me. No rackmount kit is available. I also tried the Marantz SR4000 which is slightly smaller in the rack and has a rackmount option. A third possibility is the recently launched Sony STR-DB840 which offers the bonus of six preamp outs for connection to your own power amps. There's plenty to choose from and the hi-fi press is awash with reviews.

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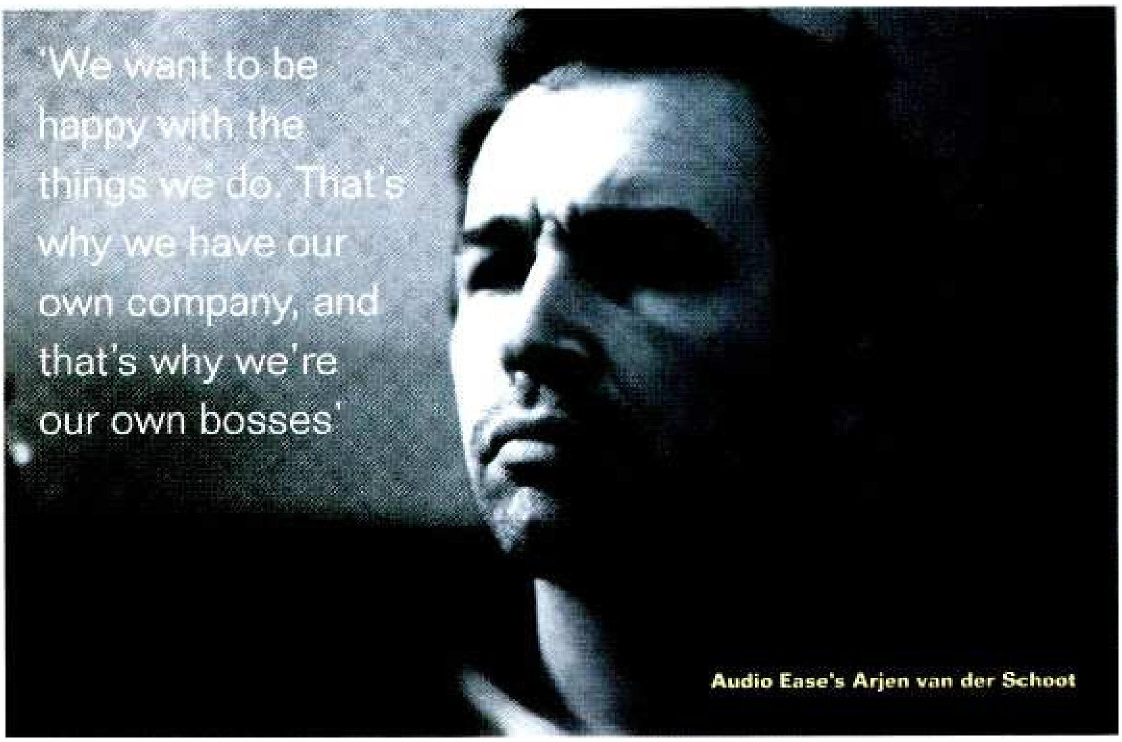
WRAPPED IN AUDIO

Playfulness and creativity mark out Audio Ease as a plug-ins company with a MAS following. **Simon Trask** investigates the highs and lows of the plug-ins world...

UNLIKE THE OTHER COMPANIES covered so far in this series, Audio Ease did not start out producing audio plug-ins. The 5-man company first made its mark with a batch audio convertor program, the oddly entitled Barbabatch aimed at multimedia and website developers, and it's still the core program in terms of earnings. Next came the Rocket Science bundle, three unusual effects plug-ins for Mark of the Unicorn's MAS platform followed by the Nautilus bundle, again for MAS. At the opposite end of the price scale from the $395 Barbabatch is VST Wrapper, a $29.95 program that allows MotU users to run VST plug-ins.

Audio Ease sells its range of software (all Mac only) via its website (http://www.audioease.com /Pages/ Store/StoreFrontUSD.html).

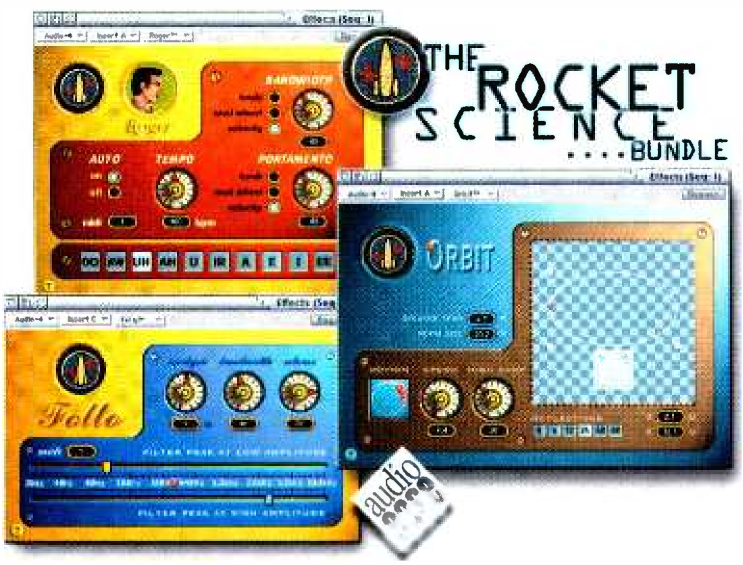
All founder members were previously students in the music technology department of the Utrecht School of Arts, in the Netherlands where the company is based. Here they collaborated on audio software projects and after graduating in 1994, Audio Ease was born. Today, Audio Ease consists of just the same five people who started the company: Peter Bakker, Jankoen de Haan, Renier Linssen, Arjen van der Schoot and Danny Weijermans. Currently, two of the five—de Haan and Weijermans—run a commercial studio, Sound Palette, that does commercials, TV and film. The studio is located next door to the Audio Ease office—useful for testing.



Audio Ease's Arjen van der Schoot

The success of Barbabatch established the company. Recalls van der Schoot: 'We were stimulated by the fact that we seemed to have struck gold with that. Which doesn't mean it made us very rich but we were visible for the whole world, when we went on the Internet and it looked like we had made something professional, which was a test in itself. When we found out that very professional people were using our software, it was a big reason why we decided to continue developing and building Barbabatch into the program that it is now.'

It was only a matter of time before working on file conversion software was no longer enough so, a couple of years back they began work on the plug-ins that would become the Rocket Science bundle: Roger the MIDI-controllable vowel filter bank, Follo the level-dependent resonant band-pass filter, and Orbit the dynamic sound localiser (which enables sounds to be placed and animated in 3-D space).



'Everything we did until then was adding features on to Barbabatch, or adding little applications that worked with Barbabatch,' recalls van der Schoot. 'We got fed up with file conversions and we were interested in doing something more creative. Also we decided we needed more products. More of us wanted to depend on Audio Ease for our income, and we didn't want to gamble on one horse. So we had the image of selling this very high-quality, high-end product and then we released something that was a lot more fun, and more like a toy, which was the Rocket Science bundle.'

While VST might have seemed the logical choice as a plug-ins platform, as it has the largest user base, Audio Ease opted to develop for Mark of the Unicorn's MAS (MotU Audio System) instead. Van der Schoot explains that this was partly because they themselves were MotU Performer users, but also because it meant they could get into a market that offered little or no competition. In addition, they were able to attract the distribution and marketing support of Mark of the Unicorn itself.

'We didn't really want to compete with the likes of Waves, which had set a real standard with the L1 and the C1, so we decided to do some more adventurous stuff. And when we had to decide on a platform, while MAS wasn't the only thing we used ourselves, it was a platform for which no third-party plug-ins existed at the time. Of course now MotU has a lot of plug-ins available, but not nearly as many as there are available for VST. So, there is still something to be gained from being MAS only. You get a lot of credit from the users for being that, and you can really focus on the features that MAS offers for plug-ins. Right now we don't need to worry about going cross-platform and therefore maybe not supporting some of the cool features that you get in Digital Performer, which have mainly to do with sync-to-beat stuff and automation.'

Still, wouldn't porting to VST open up a larger market for the company?

'We want to be happy with the things we do,' responds van der Schoot. 'That's why we have our own company, and that's why we're our own bosses. Releasing a plug-in is a very creative thing for us, while porting something isn't that creative. A port to VST is quite a lot of work, and quite a lot of dull work, and of course we're a company with extremely limited resources when it comes to programming time. We want to do fun things right now, for a while, and it's a lot more fun to create new plug-ins than to port old ones. So we might do some porting later on, but right now it's not an issue.

'From a business point of view, it might seem wise to port to VST. But maybe we wouldn't get the backup from Mark of the Unicorn. We're really much better at programming cool plug-ins than we are at marketing ourselves, so we're very happy to have their marketing muscle behind us.'

At the same time, being on the Internet has made a major difference to Audio Ease, many of whose customers live in California.

'We operate very heavily on the Internet,' says van der Schoot. 'It's made a very big difference in the appearance of Audio Ease to the outside world. It really doesn't matter where you are and how big you are, if you have a nice website you're good enough. We have our own online store, and also we get a lot of user feedback via the Internet. And the Internet was the only reason that we could be an audio software selling company based in Utrecht in the Netherlands.'

The impact of the Internet on Audio Ease's business really becomes apparent when van der Schoot discusses the sales transactions and downloads made through the company's web-based store:

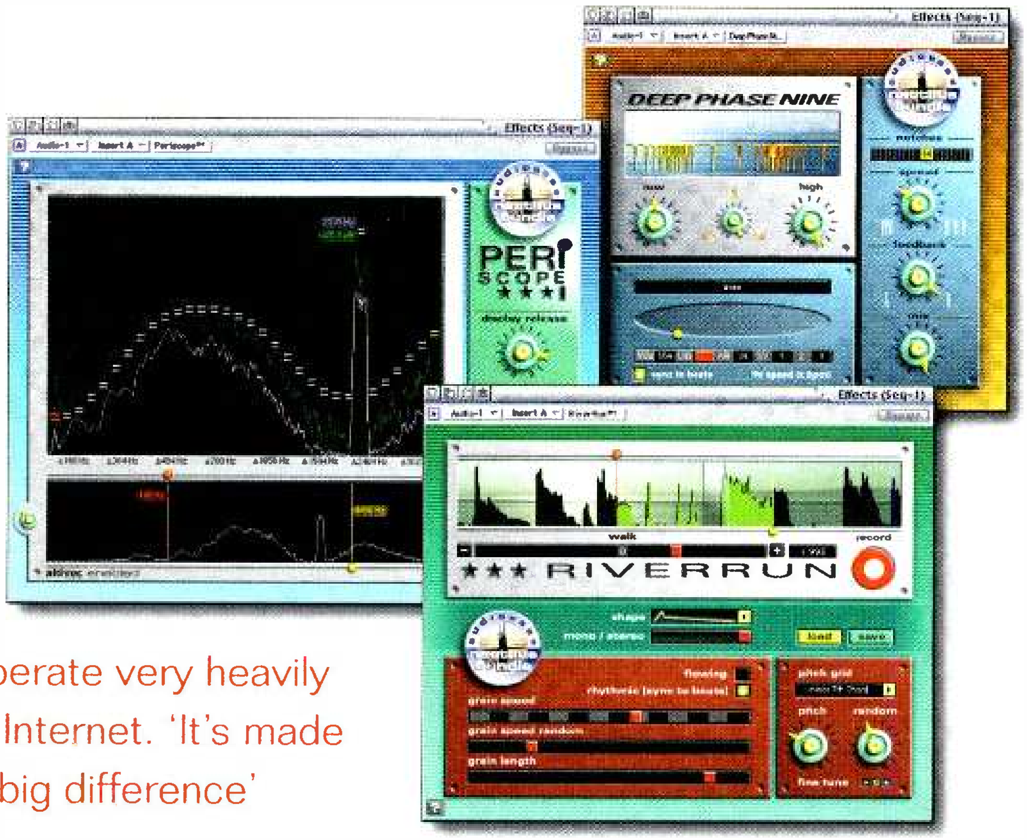
'For the Rocket Science bundle, our online store is by far the most important sell-through channel. I think 75% of our sales of the bundle are electronic only. People have the installer on the Mark of the Unicorn CD, and they only need the registration number to get rid of the beeps that make up the crippling of the demo. For Barbabatch I think it's about 50-50 right now between online and the local retail channel.'

Another one of the company's programs is only viable as an online sale and download: 'We only sell the VST Wrapper software online. Selling it any other way wouldn't be worthwhile for us, it would be just too expensive to do any sort of manual billing and packaging and sending out. And people seem to be happy with buying it online and getting a registration number right away, or downloading and trying out a demo with their plug-ins first to see if it works. It's very hard to keep every plug-in going in the VST Wrapper, and that's something we don't manage right now. The big VST synthesisers right now are not compatible with the current release of VST Wrapper, and we're working on a version that is compatible with VST 2 Instruments.'

For a company that had developed such a valuable relationship with Mark of the Unicorn, was releasing VST Wrapper such a good idea?

'It was a bit of a gamble,' admits van der Schoot, 'because we didn't know what Mark of the Unicorn would think of it, as they were pushing MAS and then there goes Audio Ease unleashing 200 VST plug-ins onto their platform! And for only 30 bucks! But we were pretty friendly with them when we came up with it, and we had some discussions with them about it and finally they thought "We'll give it a try." From their perspective it's nice to have someone else have the responsibility for running VST plug-ins on their platform. Also if someone is choosing between Cubase and Digital Performer and they're wise enough to ask how many plug-ins are available, at one time the choice would have been fairly simple, but now with VST Wrapper it doesn't matter any more. And finally, with TC Works' Spark doing something similar, Mark of the Unicorn saw that they couldn't stop people from doing it and probably they decided they'd rather have us doing it than not, because we know the platform so well.'

Perhaps not surprisingly, VST Wrapper has proved to be a popular piece of software. As van der Schoot reveals: 'If you look at our download statistics, VST Wrapper is by far the most downloaded program, followed by our two freeware programs, Make A Test Tone and thOnk_0+2 [a file-based granular synthesis program]. In terms of numbers, VST Wrapper is our biggest-selling program, but in terms of income Barbabatch is still the biggest thing, with the Rocket Science bundle coming second. But Barbabatch can be a lifesaver, whereas it's hard to know who's going



'We operate very heavily on the Internet. 'It's made a very big difference'

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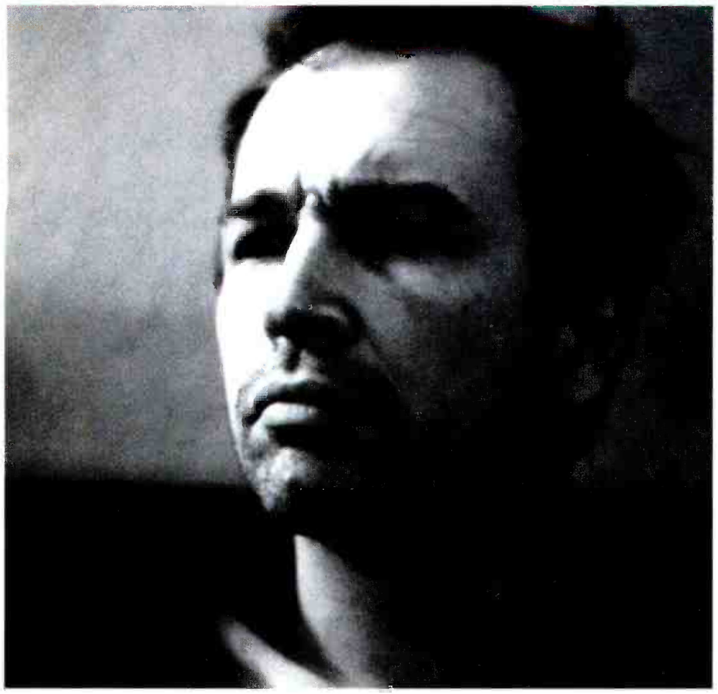

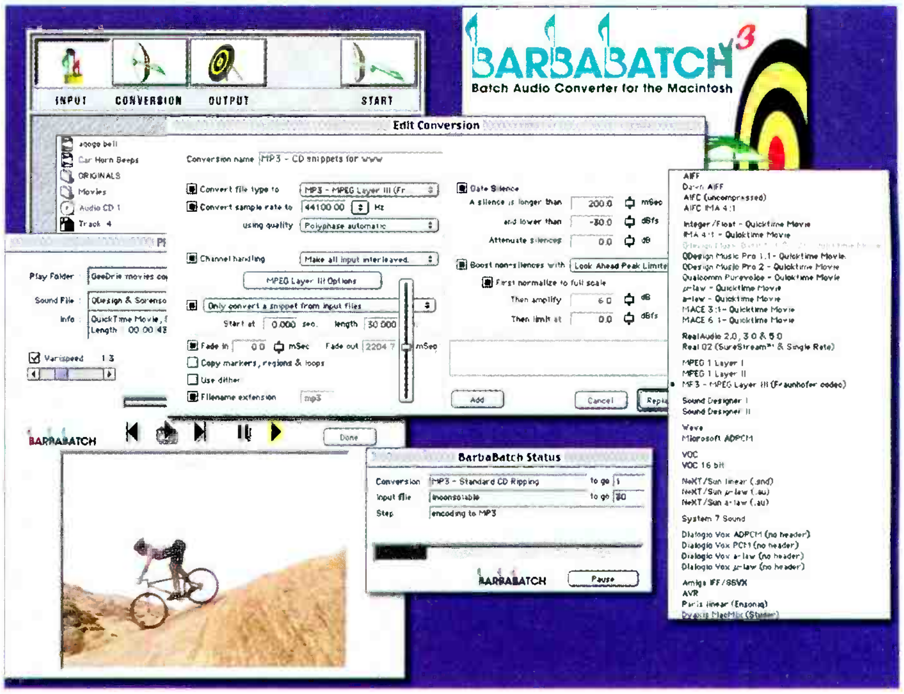

to need the effects in the Rocket Science bundle. They're great little sound-processing things, stuff that's nice to have, but lifesaving... Well, Orbit maybe, because it's a dynamic sound placement effect that can be used in postproduction quite heavily, to have someone wander about the room, for instance.'

After having made a splash with its first bundle, Audio Ease is now set to move deeper into plug-in waters with the upcoming release of its Nautilus bundle. Like the Rocket Science bundle, Nautilus consists of three MAS plug-ins—in this case going by the names Riverrun, Periscope, and Deep Phase Nine. Once again offbeat in name and off-beat in nature, Riverrun is a real-time granular synthesis plug-in, while Periscope is an equaliser plug-in with a difference, and Deep Phase Nine is what van der Schoot terms 'a true phaser rather than a flanger sold as a phaser', but also with a 'very flexible graphic LFO.'

'The ideas for Riverrun almost all originated from thOnk, but now we offer full control in a real-time environment, and also it's more flexible, you can sync to beats and it's fully automatable,' enthuses van der Schoot. 'You can feed it audio and make the sound freeze, then slowly walk back with it—use it as sort of a sound microscope if you like. You can really look closely at the structure of a sound vibrato, or the pluck of a string, and get incredible sounds.'

Meanwhile, Periscope shows that even with a 'regular' effect Audio Ease has to push it into new realms of creativity.

'When we got the equaliser engine going, we decided to do an equaliser after all. The sliders go down to -144dB, and pulling away a band silences it. Also it's very sharp, so if you want to take out say a 10Hz-wide band at 3kHz you can do that.'

Periscope is also one of the first if not the first effects plug-in to make use of the Altivec processor in the Power Mac G4. 'It makes a huge difference,' says van der Schoot. 'Periscope does something that's very elaborate, and normally it would eat up almost a third of your G4 processor, but with Altivec too it's well under 5%.'

While it may not be one of the biggest names in audio effects plug-ins, Audio Ease clearly has something unique to offer, and consequently the company has managed to establish a niche for itself.

'The ideas for Riverrun almost all originated from thOnk. You can use it as sort of a sound microscope if you like. You can really look closely at the structure of a sound vibrato, or the pluck of a string, and get incredible sounds.'

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New is not always better

A rare opportunity to tap into the technical history of CBS uncovers a battleground of politics, egos and genius writes **Barry Fox**

AT THE END OF SEPTEMBER, while the AES debated watermarking in Los Angeles, Panasonic-Technics was launching DVD-Audio in Europe at a London hi-fi show—without any DVD-A discs to play. I chose the hi-fi show and who should be there? Teo Macero, the man behind those classic CBS recordings of Miles Davis, Duke Ellington, Charles Mingus and Ella Fitzgerald. He was in town to receive a Lifetime Achievement award from UK magazine *Hi-Fi News*.

'It's the only award I've ever got,' Macero told a small gathering of hi-fi buffs, some of whom clearly had no idea who he was. 'I never got a Grammy. They just give Grammys to the people who reissue my recordings. They weren't there. They don't know how it sounded'.

Through the encounter, Macero made no bones about his feelings for CBS. 'Clive Davis tried to get me fired,' he laments. 'Goddard Lieberson said, "leave him alone", and when Lieberson left they took away his ID card. He couldn't even get into the Goddam building. Then they came after me. Called me into the office and said, you're through. I left with a lot of hatred but a lot of money. Lieberson had started this great bonus system. It cost $1500–$2000 to make a recording, and after break-even we got as percentage. We could pick up $80,000 if a record sold a million. And we got unlimited expenses. I had a yacht, and aeroplane—until I got divorced.

'I went to the new people at the top, like Bruce Lundvall, and said we should be recording Duke Ellington and others while they were still alive. They said no. They wouldn't even let me record Duke's Sacred Concert at Westminster Abbey. When Miles did *Sketches of Spain* they said, "Why are you doing this?"; I said, "Why don't you just shut up?".

'Miles was crazy. Mingus was absurd. But that's why they never stopped growing. Dave Brubeck just got comfortable. Mingus once walked out of a session. I told him I couldn't care less. He got out the door and came right back.

'In the sixties we were all set to record Miles live at

Carnegie Hall. He called up just as we were leaving CBS with the equipment and told me it was off. I was so angry I borrowed a mono Wollensack and a four-pot mixer, hooked it to the stage mics and made a 7.5ips tape without his knowing. After the concert I went backstgage and threw it at Miles. This is what you missed, you idiot, I screamed. The next day Miles called and said he wanted to release the recording. You can't I said. It's an illegal copy, mono and lousy quality. An engineer at CBS, not the greatest in the world, tricked it up with some fake stereo. But there was a lot of distortion. It was a massive seller. [The disc notes simply refer to 'original mono tapes'.]

For years now, Macero has been trying to get the original tape back from CBS-Sony. 'You left it in our vaults so it's ours, they keep telling me. I've never received a penny in royalties.'

Macero wants to see if his original labelled box is in the vaults. 'But they won't let me in the building'.

'Later on Miles got lazy. He demanded $350,000 for each recording and he'd leave it to the band to do all the work and then just come in and put on his tracks. He'd pay a limo to wait outside his house for three weeks and use it once.'

Macero doesn't like CD. He prefers vinyl. 'There's something wrong; no warmth, no breadth, no EQ.' He doesn't like SA-CD either. He told Sony he preferred the CD layer of the hybrid disc. He thinks that re-issues with extra material are 'terrible'. 'It's like re-printing a book with errors the editor had previously taken out. Reissues like the *Plugged Nickel* box set just put the mistakes back in. The record was finished and now they are unfinishing it. And they don't have the rights to the extra material. I tell the musicians to ask for more money.

'I worked with Miles for 28 years. He gave me a free hand to do what I wanted. We looped vamps and used a tape echo with four heads, all independently adjustable. But he always had the final say on what was released.'

The most magical session? 'The one we did with Basie and Ellington (Battle Royal) in a church on 30th Street. It was like Heaven opening up. I put the bands left and right, with both the rhythm sections in the middle. Basie wanted the bands mixed together. I had to talk him out of that. Then Basie refused to play his solo and Billy Strayhorn had to mimic him. We used a lot of mics but mixed down to three-track tape. We didn't dare give the musicians a break because they'd have gone off to get straight and we would never have seen them again.

'I miss the spontaneity in today's recordings. At CBS we were doing three record dates a day. We'd record a Broadway cast show and release it within a week. Now it takes a year. One reason why those things sounded better is that we would check the frequency response of every mic before every session. But the tapes CBS used were poor. We called it Hiss Records.'

Macero is now releasing around 40 CDs of recordings he made in small studios after leaving CBS; some jazz, some banjo music. 'I'm a banjo Freak,' he says. 'I don't have the time to go back to the original multi-tracks. I work from DATs of the stereo mix.'

How? 'I found out by chance that if I took the left and right channels, mixed them together and added a little back to the stereo mix I can bring out the solos.'

Organised Crime

While reckoning to represent the interests of thier members, many organisations muddy waters or even occasion change writes **Dan Daley**

THE OTHER NIGHT I was sitting in the garden of a very nice restaurant in Milan, having dinner with Ron Goodwin, a British national who has lived and worked in Italy in sales for Tapematic for the past 20 years but who started out his audio career at the venerable Trident Studios in London. Though he never got past the post of tape op there, he performed that service on many a classic recording, including mixing for the Beatles' *White Album* and recording parts of *Let It Be*.

Anyway, after about the fifth or sixth grappa, and in the midst of a conversation about sound effects then and now, Ron reminisced about how engineers used to con studio musicians into doubling a part by feigning that the last pass was a bit dicey, turning off the monitor return, and recording it a second time to another track. This was done, of course, to dodge the union rules that would have incurred additional charges for playing more parts. I let on that as a producer for United Artists Music in the seventies, I was introduced to the same practice, but with background singers. And as a session vocalist myself, I had it pulled on me. After a few years in the studio—or a few months, if you were sharper than I at the time—most caught on. And usually it got to the point where there would be knowing nods and winks through the control room glass; few performers chose to risk not getting a callback by sticking too hard to the rules.

For those in the business over the age of 40, you're probably nodding your heads in reveries of your own experiences in this regard. For those younger, you're probably saying to yourselves, 'Union? What's a union?'

The fact of the matter is, members of this industry—engineers, producers and musicians alike—have become increasingly unorganised over the last four decades. (Please note I said 'unorganised,' not 'disorganised,' a literal distinction from which only drummers are exempted.) For instance, Local 802 of the American Federation of Musicians, in New York City, never counted much membership among rock musicians starting in the 1960s, due to the fact that the Local's elderly leadership, made up mostly of Broadway theatre-pit tuba players, considered rock music a passing phase and that they'd all soon be back to Rogers & Hammerstein and *My Fair Lady*. (And since Disney took over Times Square a few years ago, they were proved right, but that's another story.) A half-hearted attempt, in the mid-1970s, led by a slightly younger contingent in the Local (that is those who still had prostates), sputtered, underscoring what would become the pattern for the rest of the century: trying to organise music types in an age of accelerating technology was as pointless as herding cats.

Fast forward to the present. You thought geriatric tuba players were bad, try multinational corporations and venal trade organisations. The major record labels distanced themselves from producers and engineers when they closed their recording operations. And more recently they have used their international clout to curtail rights even further; for instance, it is now standard practice on label contracts to demand that newly signed

bands and artists (and some producers) assign the ownership—in perpetuity—of their websites to the labels.

Much of this is fearful knee-jerk reactionism to the threat of downloadable music that the MBAs who run those labels missed out on while snorting corporate cocaine and who are now trying to use any means available to recoup. Unfortunately, the thousands of independent labels which were spawned from that same digital revolution haven't proved much better about such matters.

Then there was the scam almost perpetrated by the Recording Industry Association of America (RIAA) earlier this year, which, after quietly, but intensely lobbying Congress, tacked on an amendment to upcoming revisions to the US copyright law which would have gutted the provision allowing the rights to master recordings to revert to the artists after 35 years. A few sharp music business lawyers spotted the move for what it was, and the resulting hue and cry propelled the creation of the Recording Artists Coalition, whose main spokespersons, Cheryl Crow and Don Henley, forced the RIAA and their Congressional cronies to back down.

The reality is, even as digital technology empowers and connects members of the audio community as never before, it also isolates them as never before. Like some high-tech Pharaohs, the garage band has entombed itself in the garage, able to make, mix, master and market their own records without ever opening the door. (Which would be done with a remote control, of course.)

There have been moments when organisation has moved to the fore, mostly notably in the States with the formation of the Music Producers Guild of America (MPGA) several years ago. And mastering maven Denny Purcell still has not given up on his attempt to organise mastering engineers. And there are numerous ad hoc local organisations, such as Los Angeles Women in Music, and the New York [Engineers] Pool. And the UK and Europe have been even more active, creating the APRS Record Producers Guild and the European Producers Guild.

But at least in the States, meaningful organisation has remained elusive. Even the MPGA has now affiliated with NARAS, a questionable move considering how snugly in bed the Grammys are with the major labels. Perhaps true organisation is impossible, considering the ethereal nature of digital technology and the independently minded sorts who make up the creative side of the music business. I don't know. What I do wonder, though, is: does Lech Walesa play an instrument?

Standard and deliver

Digital television is inviting and inciting its players to promise more and greater services writes **Kevin Hilton**

IT COULD BE that I am looking for some spiritual support and guidance in this decentralised, dehumanised technological society—or it could just be that the Bible is still a good source of apposite quotes—but during IBC 2000 I was reminded of words from the Gospel of St Matthew.

'But many that are first shall be last; and the last shall be first' kept repeating in my head during Digital Terrestrial Television Service Models, a 'mini-conference' organised by the Digital Television Action Group (DigiTAG). The UK, narrowly followed by Sweden, was the first country to implement DTT services and the experience of its broadcasters, producers and multiplex providers have doubtless been invaluable in assisting those in other countries. Each national situation is different but there are commonalties, which is possibly why the presentations of both British speakers were so general.

OnDigital has already implemented email for its viewers, with the ability to switch between TV and web content now being promoted. Peter Marshall, technical director of the Digital Television Group (DTG), spent much of his time discussing digital-analogue interference and the money the UK government stands to make from auctioning off freed-up spectrum. Like many who followed him, Marshall made the point that DTT must be made appealing to challenge digital satellite and cable, concentrating on the blend of programming between simulcast familiar channels and new services, both TV and text based.

MHP (Multimedia Home Platform) is the new specification from the DVB (Digital Video Broadcasting) Project, designed to introduce, among other things, electronic programme guides, associated information services and interactive advertising and programmes (the ability to put *Who Wants to be a Millionaire* contestants on the wrong track). DVB-MHP will begin with enhanced broadcasting, moving up to interactivity—complete with the much-discussed return channel—and then Internet access. With fears over mismatched formats and consumers shelling out for new equipment that becomes obsolete before they open the box, modularity has been built into MHP to make upgrades easier.

An interloper from consumer electronics was Adrian Northover-Smith, digital TV project manager at Sony Europe, who said that manufacturers and broadcasters need to emphasise that DTT is not just about pay TV. Maybe it was a shameless plug or it could have been realistic, but he stated that the Integrated Digital TV set (IDTV) would be a key factor in this, with demand driven by the replacement of existing units rather than the desire to subscribe to pay services. The UK, he said, had created the myth of DTT as largely being about subscription services. 'This is not true. We have to make a common platform, with a single step up from analogue. An IDTV will have access to either double the number of channels analogue offers, or those plus pay TV.'

It is clear that broadcasters should not forget that TV is TV, but the influence of computing and the web, driven by telecoms companies, is changing attitudes and bringing new influences. Finland has a particularly dynamic telco sector that is aggressively promoting the creation of so-called information society services. Finnish national broadcaster YLE is supporting this; it is perhaps significant that the country's government has delayed the launch of its three multiplexes until it has an 'open platform solution', which it hopes will be by August 2001.

Ismo Silvo, director of digital programmes at YLE, observed, 'Then we believe that all the main elements of an open platform will be at hand and receivers will have reached the consumer markets. Then we hope that DVB-MHB services will flourish.' DVB-MHB will doubtless be central to what Silvo called the Internet profile set-top box, which he said will allow full Internet content to be brought easily to TV screens.

Interactive brings thoughts of countless screens and menus; indeed Silvo made the point that Finland considers itself a 'reading nation', so the concept of text services is not completely alien. National differences and tastes will tailor each DTT profile, but if they involve the implementation of a particular technology for a specific need, there is no reason why other countries should not consider them.

Peter Branagan, director of digital planning at RTE, challenged the view that Ireland was the only country interested in a wireless return channel for interactive digital television, web browsing, secure e-commerce transactions and, for the future, intermittent file transfer. The draft technical specifications for this, DVB-RCT, were due to be submitted to the technical module of DVB a week after IBC, with the final approval of the steering board expected during the start of 2001. Commercial product could be on the market in the third quarter of that year; Branigan said as this has to be an always-on technology, it eliminates PSTN and GSM and has to be low cost, which makes UHF a likely candidate.

Perhaps I am doing the UK speakers a disservice here and they were asked to limit themselves to nonspecific overviews. Local broadcasters must be considering the services and technologies highlighted by the other participants, but, at a cursory glance, Britain's DTT services could be set to follow some of its sports teams and prove old St Matt right yet again.

THE AMPEX MM1200

Regarded by many as the definitive vintage multitrack, the MM1200 was made in a variety of versions and with a number of flaws. **Tony Arnold** gives his definitive tips and tricks on fettling the beast

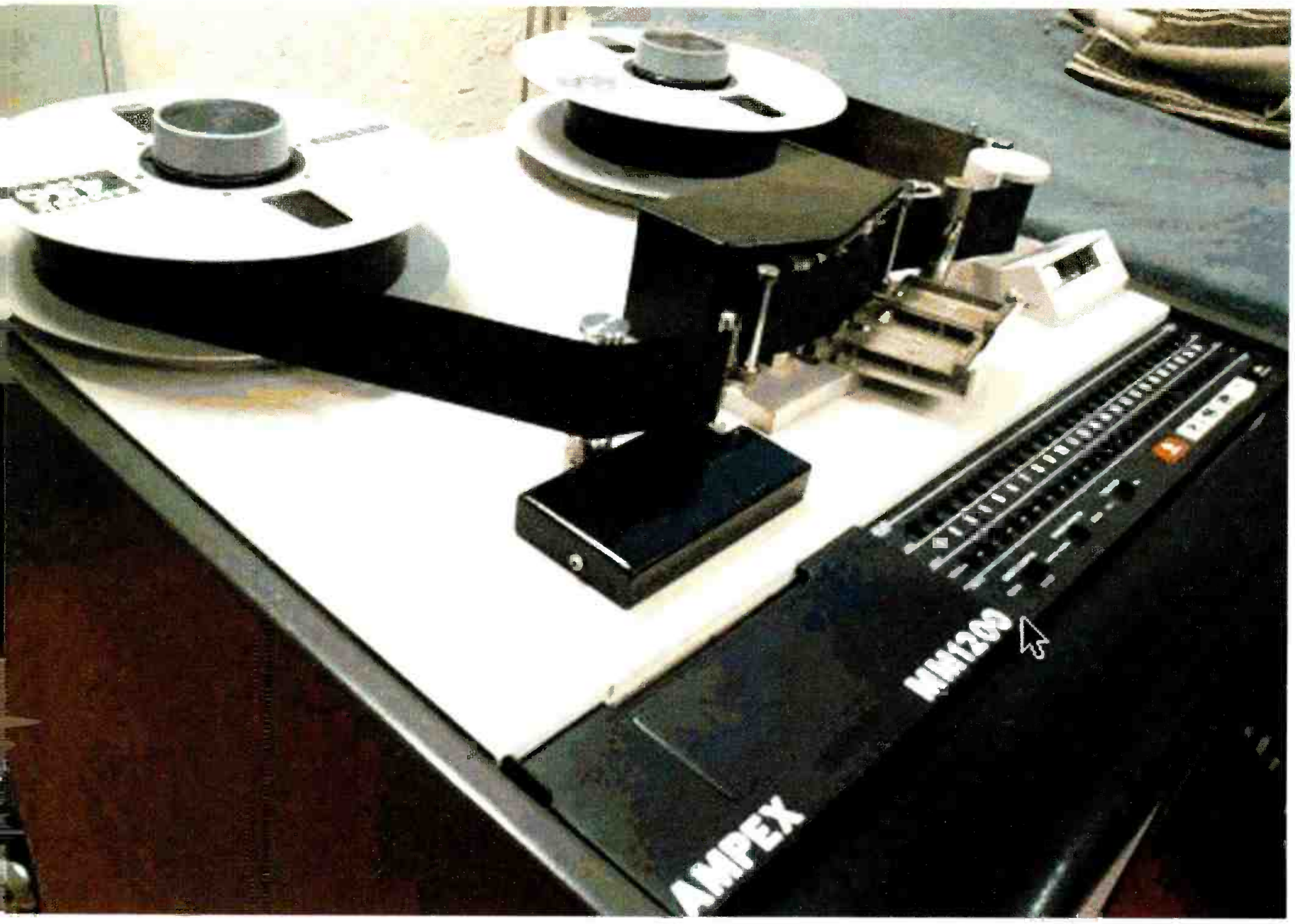
IF YOU HAVE A POST 1980 VERSION of the Ampex MM1200 then you own a real workhorse, but it's a worksurface that is so easy to maintain that any local TV shop could fix it. This and its use of discrete electronics make it possibly the top vintage multitrack to own.

There were basically three models of the MM1200, the first (V1) introduced in 1976 still had the original MM1100 transport logic card with only 10 single-turn presets. The brakes remained open in Stop mode with the stationary hold being held by the reel motors, the tape would drift slightly to a different position and the engineer could place the recorder into record with the tape in a different position as to when he stopped the tape. The take up boost in replay had no adjustment and the meters were of the MM1100 type, which made bulb changing a task.

Later with the V2 Ampex added 4-turn presets, with three extra to control take-up boost and 1-inch take-up. The brakes were initialised in stop mode and silent braking was fitted. Easy bulb changing Modutec meters were fitted, plus two extra rear cooling fans.

Finally, with the V3, Ampex fitted a constant tape tension control system and added an extra TO3 transistor to each reel motor driver and improved the cooling of the MDA.

If the rear wheels on your MM1200 revolve and are not fixed in one direction then it is most likely you have a very late model. If you have any Ampex including the MM1200 with green or natural colour PWA cards instead of blue then you have a post-1981 machine and it should prove extremely reliable. Other items to look for are PURC bias cards which enable gap-



less drop-in and pull-out of record. There were three types of these, the first was a disaster (part number 4500291-01 to 05) and if you have this type I suggest you keep your original standard bias cards.

Next came the 4500291-06. These were difficult to set up, but if calibrated correctly at regular intervals they did the job. These and the previous cards can be easily recognised as they consist of two PWAs, one above the other.

The final cards, which consisted of a single PWA Part No 4500383-01 to 02, were fantastic, totally reliable, quiet, and fast.

The original MM1200 MDA (motor drive amplifier) used two DC coupled output devices per reeling motor. The reliability was poor. The MDA ran hot and often failed during mode switching. In an attempt to correct the problem, Ampex reversed the direction of the cooling fan and added fan shrouds to improve the airflow to the output transistors. At the same time 1.5kV 200A transient suppressers were added to the MDA circuit card to limit switching transients. This improved the reliability of the MDA, but did not totally correct the problem.

The MM1200 3-transistor per motor reel MDA upgrade kit corrects the problem when installed properly. By adding a third output device the MDA runs cooler and can swing higher current to the reeling motors.

Ampex changed to this improved MDA on the last few machines that were manufactured so be sure to check before proceeding with the mod in the unlikely event that your machine is of this vintage.

Many engineers think the photovolaic diode selenium maybe faulty, but they should first check the condition of the flat phosphor bronze spring (Service Manual No 4890321-05, page 6-38, Item No 6 Part No 4270131 spring sensor). This item can often end up with a crack in it after many years which could indicate a problem with the tension arm sensors.

The engineer should also check that the bulb is working in the unit itself and make sure that when blocking the light from the bulb he gets a change of voltage at the Motor Drive Amplifier itself. If he does then most likely the photovolaic selenium diodes are fine.

Another good test is that they should give a DC resistance of 1.1M red probe to red cable and black to black cable, and 3M DC resistance red to the black cable and black to red. This test should be carried out with the photovolaic selenium diodes covered over and completely in the dark so no light is entering.

Also check that they are functioning correctly by using a torch and varying the distance from the Photovolaic Selenium Diode while monitoring the DC resistance.

If your MM1200 is running very hot then I suggest that you check at the rear of the machine to see that you have three fans fitted behind the audio bays —some early machines had only one. At least once a week clean the fan filters.

Adjusting the azimuth or phase of the record and playback heads is not possible in some instances. Ampex made the mistake of using an engineer's type block to fix the heads to and these were spot on for azimuth when assembled. However, this did not allow adjustment for tapes recorded on other machines that were out of phase

Warning!

ALWAYS ALWAYS MAKE sure that your MM1200 has tape lifter sleeves fitted over the tape lifter arms. otherwise a master can be severely damaged due to wear on the original arms.

Machines are often changed from 1-inch to 2-inch, but engineers fail to change or turn the tape lifter sleeves and sometimes they do not fit them at all and end up wondering why they have a severe scratch down the middle of a 2-inch tape or why the edges are curled.

Stopping monitor break in monitors
On the audio switching card remove C4 = 10 uF.
Replace Zenner VRI with a 15k resistor.
Change C6 = I uF to 0.47 uF x Tant.
Change C3 =10 uF to 47pF.

How to speed up punch-in/out time on earlier MM1200s.
This should only be incorporated if the customer complains of a slow function in this respect, otherwise a pop on tape may result because of the sharp bias ramp generated by the modification.
On the MM-1200 PWA 4050774.
Change R2 from 1k to a 6.2k X 1/4W X 5% resistor Ampex Part No. 041-538.
Change R5 from 10k to 5.1k X 1/4W X 5% resistor, Ampex Part No. 041-561.
Verify that no pops or clicks are present after the modification.

and with misaligned azimuth. Flux Magnetics have not only improved the quality of these heads, but have now allowed for the new heads to be phase-azimuth adjustable.

Another complaint about the MM1200 is the speed variation between the beginning and the end of tape. This is not really a problem but a take-up tension controller kit (Part No 4051099) does improve the drift to modern standards. It is usually due to head wear and how the transport has been set up.

After adjusting and setting up the transport, I always put a 440Hz tone for one minute on track 1 at the beginning of tape. I then turn the tape over so it is playing back on track 8, 16 or 24 (depending on how many tracks your MM1200 has) and feed the 440Hz tone in to a guitar tuner. It should be spot in-tune, and a sine wave can also be used with a digital counter, or a 3kHz tone with a drift meter.

If after this test you experience high frequency drop out then your heads are badly worn.

The MM1200 take-up tension controller (TTC) provides for automatically regulated tape tension between the take-up reel and capstan when in Play mode. The need for a tape tension sensor is precluded because the information necessary to regulate tension is determined as a function of tape speed and reel rotation (take-up tension controller kit, Part No 4051099).

Electronics for the TTC are contained on four PWAs, three of which are mounted on brackets and one in a piggyback configuration. All PWAs are installed internally on the MM1200 and interconnection of the various PWAs is made possible via multipin connectors which already exist in the MM1200, or are supplied in the kit.

PHOTOGRAPH: WWW.REGGAEPHOTOS.COM

One of the great things about the MM1200 is its beautiful low-end response and this is mainly due to the head design. Unlike a lot of heads they are hyperbolic and often when they are relapped they end up round, which really upsets the low end response. Care should be taken in the relap process to maintain the original profile. A lot of reggae producers favour the original black heads for their particularly smooth low-end response and as a consequence they are very desirable

To discover if one is already fitted, check underneath the shelf where the control box or transport control box sits and you should see a green PWA . Also to the right of your transport-capstan-search to Cue PWA box, where you switch on the MM1200, there should be a small LED added. Finally check on your Capstan PWA where there should be a small piggy-back PWA.

Relays

On each audio switching card there are four relays marked K1 to K4. Relays K2, K3 and K4 are momentary make before break relays, to stop any audio clicks during changeover, and these relays are now almost impossible to obtain. We have checked K3 and K4 and have discovered that these do not have to be make before break, so they can be changed for a standard relay which is still available from most suppliers. But what do we do about K2? We now have stock of this type but we do prefer to offer a reconditioned exchange unit which carries a guarantee.

The only other problem regarding parts for the MM1200 is the DTL ICs incorporated in the capstan and transport logic PWAs. Should you come across a stock of these then I suggest you purchase. They are very expensive and very rare and because they run on 15V rails we have been experimenting with CMOS ICs as we are slowly running out of replacement DTLs.

Should you have any questions or any information that you may wish to pass on to the Ampex Owners Club, then please email me and I will do my best to solve any problems.

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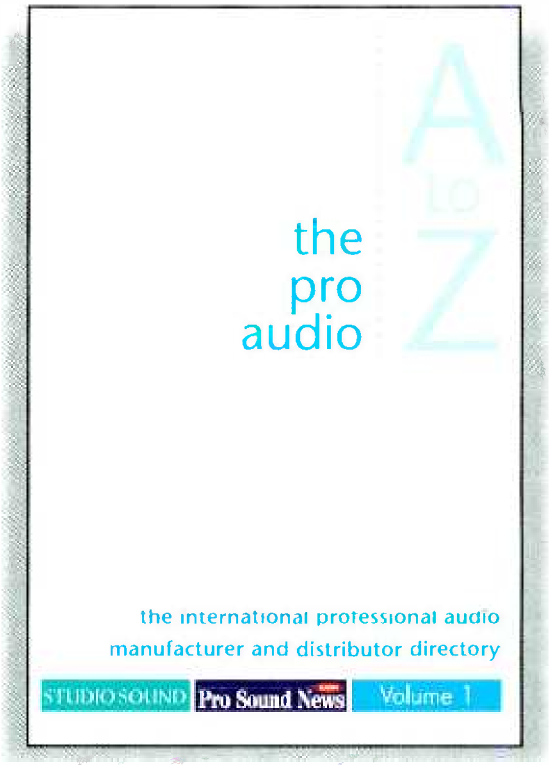

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PERCEPTION

Understanding the human visual system is a prerequisite for working with any sort of pictures. **John Watkinson** enters the wild and illusory world of sight

VIDEO AND AUDIO have in common that they are both useful only to excite the human senses. The parallels continue, because both hearing and sight attempt to create a model of one's surroundings with some help from the balance organs. In an inaccurate audio system, the distortion of the model can cause listening fatigue. In an audio-visual system, disparities between the information from the picture and the sound can have the same result. Only by considering sound and pictures as a continuum can the best results be obtained.

Here I propose to look at the human visual system, but not through the traditional biology lesson approach. I don't believe that being able to identify a gudgeon pin helps with road safety, and in the same way naming the parts of the eye is much less important than explaining what the eye does and why.

The human visual system (HVS) evolved as a survival tool. A species that could use vision to sense an impending threat, to locate food and a mate would have an obvious advantage. Watching television, from an evolutionary standpoint, is very recent. The eyes are coupled to a series of less obvious, but extremely sophisticated processes that take place in the brain. The result of these processes is what we call sight, a phenomenon that is difficult to describe.

At an average reading distance of 350mm, the print in this magazine subtends an angle to the eye of about a third of a degree. The lines from which the letters are formed are about one tenth of a millimetre across and subtend an angle of about one minute (one sixtieth of a degree). The combined field of view of the eyes is nearly a hemisphere. A short calculation will reveal how many pixels would be needed to convey that degree of resolution over such a wide field of view. The result is simply enormous—so large that it is utterly inconceivable that the nerves from the eye to the brain could carry so much data, or that the brain could handle it.

Instead the visual system helps create a model in the mind of the reality around it. Reality can be defined as an illusion brought on by a shortage of alcohol.

Fig.1 shows the idea. The model can be considered like a kind of 3-dimensional frame store in which objects are stored as the HVS identifies them. Inanimate objects can be modelled once and left in the model until there is evidence to suggest that there has been a change whereas moving objects need more attention.

The eyes can swivel to scan the environment and their owner can move within it. This scanning process allows the model to be built using eyes with a relatively narrow field of view. Within this narrow field of view, the provision of high resolution and colour vision does not require absurd bandwidth, though it does require good lighting. And, although the pixels are close together, the total number is fairly small.

Such narrow vision alone is not useful because events outside the field of vision do not alert the user to the need for an update of the model. Thus in addition there is a wider field of view that has relatively poor resolution and is colour blind, but which works at low light levels and responds primarily to small changes or movements.

Sitting at a laptop computer writing these words, I can only see a small part of the screen in detail. The rest of the room is known only from the model. On my right is a clock. In peripheral vision it appears as a grey lump, but in my mind it has not changed colour. The ticking of the clock is coming from the same place in the model as the remembered object, reinforcing the illusion.

If I were replaced with a camera and a stereo microphone, and the two then turned to the right towards the clock, the visual image and the sound image would both move left, suggesting that the whole world is moving. However, if I myself turn right this doesn't happen. The signals from the balance organs in the ear suggest that I am turning and the rotation of the sound image model and the visual model the opposite way produce data consistent with the fact that it was I that moved and not the model. By becoming another object in the model, my limbs are included in the model so that I can see an object and pick it up.

This interaction between the senses is very strong and disparities between the senses are a powerful clue that one is being shown an illusion. In advanced systems for use in, for example, flight simulators, it is vital to maintain accurate tracking between the visual image, the sound image and the sense of balance.

When seeing via a model, we often see what we expect to see rather than what is really there. Optical illusions demonstrate this. The technique of camouflage destroys familiar shapes and confuses the modelling process. Animals and birds may freeze when predators approach because their lack of motion prevents them from triggering the predators' peripheral vision.

Inside the eye, the retina is responsible for light sensing and contains a number of layers. The surface of the retina is covered with arteries, veins and nerve fibres and light has to penetrate these in order to reach the sensitive layer. This contains two types of discrete receptors known as rods and cones which have different distributions and characteristics. Rods dominate the periphery of the retina whereas cones dominate a

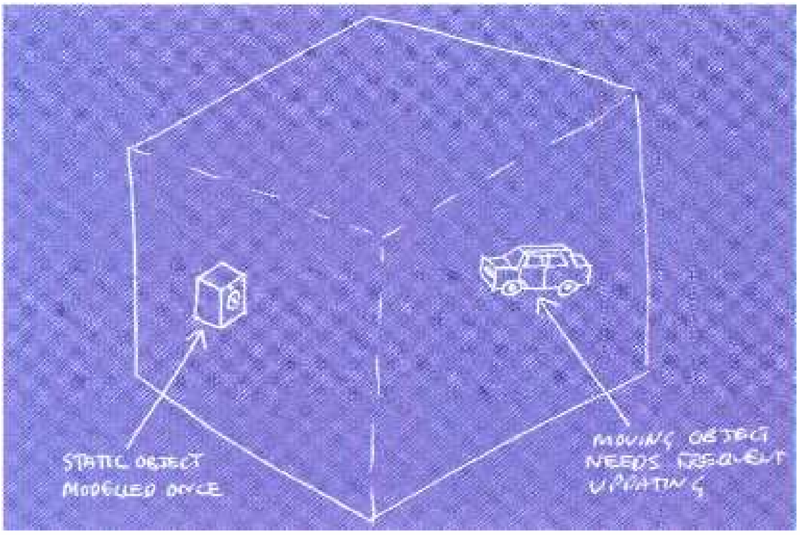

Fig.1: The senses create a spatial frame store which holds objects. More than one sense can contribute, for example sound and vision can be combined to locate an object, or sound can be used to detect an object which the eyes then turn to identify

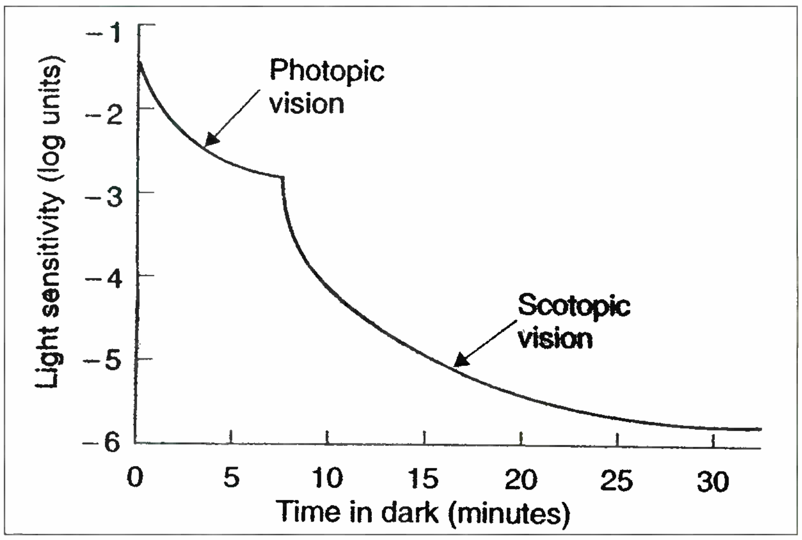

Fig.2: Retinal sensitivity changes after sudden darkness. The initial curve is due to adaptation of cones. At very low light levels cones are blind and monochrome rod vision takes over

central area known as the fovea outside which their density drops off. Vision using the rods is monochromatic and has poor resolution, but remains effective at very low light levels, whereas the cones provide high resolution and colour vision but require more light.

Fig.2 shows how the sensitivity of the retina slowly increases in response to entering darkness. The first part of the curve is the adaptation of cone or photopic vision. This is followed by the greater adaptation of the rods in scotopic vision. At such low light levels the fovea is essentially blind and small objects which can be seen in the peripheral rod vision disappear when stared at.

A significant area of the retina, where the optic nerve connects, is completely blind. However we are not aware of a hole in our vision because we don't see the image on the retina literally. Instead we see the visual model which inserts information in the hole. The hole is in a different place in each eye which assists in this concealment process.

The cones in the fovea are densely packed and directly connected to the nervous system allowing the highest resolution. Resolution then falls off away from the fovea. As a result the eye must move to scan large areas of detail. The image perceived is not just a function of the retinal response but is affected by processing of the nerve signals, one function of which allows the visual system to ignore the fixed pattern of shadow on the retina due to the nerves and arteries.

The resolution of the eye is primarily a spatio-temporal compromise. The eye is a spatial sampling device like a CCD camera. However the measured acuity of the eye exceeds the value calculated from the relevant sampling parameters. This is possible because a form of oversampling is used. Fig.3 shows that the eye is in a continuous state of unconscious vibration called saccadic motion. This causes the sampling sites to exist in more than one location, effectively increasing the spatial sampling rate provided there is a temporal filter which is able to integrate the information from the various different positions of the retina.

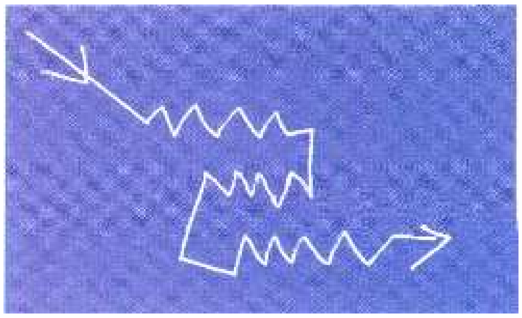

Fig.3: The eye constantly changes its angle of view. This increases visual acuity when successive images are integrated

The HVS does not respond instantly to light, but requires between 0.15s and 0.3s before the brain perceives an image. At low light levels the time taken to perceive an image increases. The position of moving objects is perceived with a lag that causes the geometry of stereoscopic depth perception to be in error, a phenomenon known as the Pulfrich effect. This is why cricket matches are stopped in poor light.

Scotopic vision experiences a greater delay than photopic vision as more processes are required. Images are retained for about 0.1s. Flashing lights are perceived to flicker until the critical flicker frequency is reached, when persistence of vision makes the light appear continuous for higher frequencies. Fig.4 shows how the CFF changes with brightness. The CFF in peripheral vision is higher than in foveal vision. This is consistent with using peripheral vision to detect movement.

The critical flicker frequency is also a function of the state of alertness of the individual. When people are drunk, tired or under stress, the rate at which the brain updates the model appears to slow down so that the CFF falls. Interestingly the critical bandwidth of the hearing system also widens under these conditions, so for high quality audio-visual work you will have to manage with just sex and rock and roll.

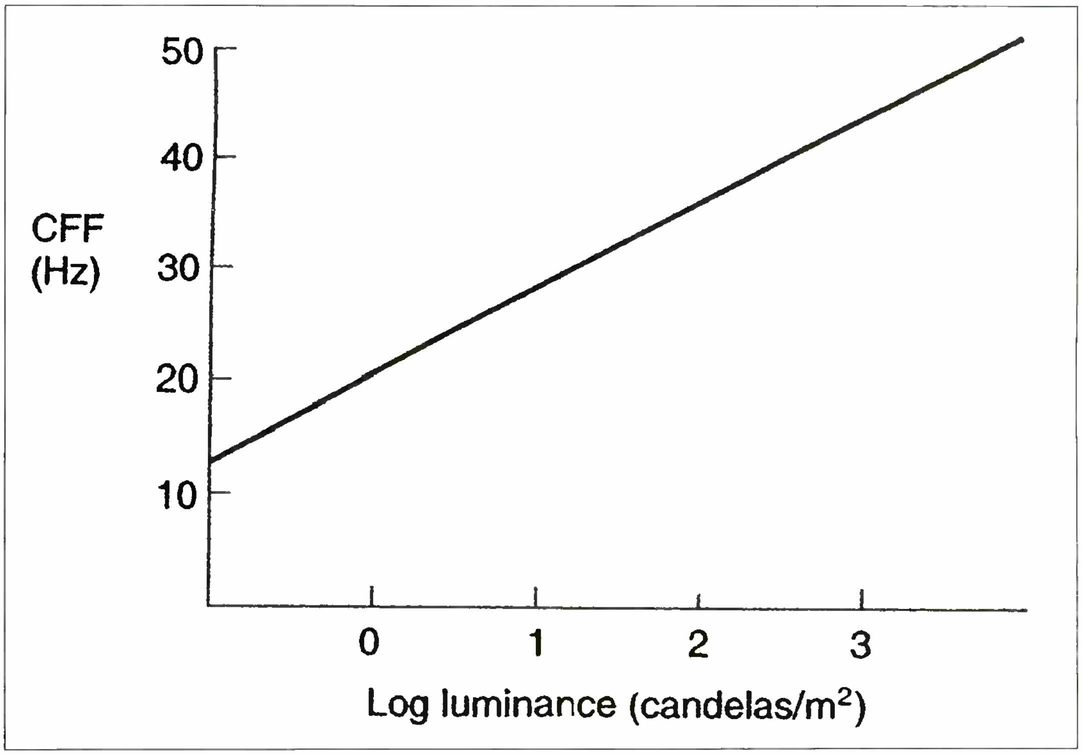

Fig.4: Critical flicker frequency varies with brightness. Rates used in television and film are marginal for bright displays

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Fax: +44 1582 472946.
Email: irene.chamberlain@uands.com
Net: www.uandsol.co.uk

7–10
Satis
Paris, France.

8–9
Sound Broadcasting Equipment Show
Hall 7, NEC Birmingham, UK.
Contact: Point promotions.
Tel: +44 1398 323 700.
Fax: +44 1398 323 780.
Email: info@pointproms.co.uk.
Net: www.sbes.com.

8–10
Replitech
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14–15
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Fax: +44 (0)20 7611 7530.
Email: gsherman@ibc.org
Net: www.ibc.org

15–17
36th Inter BEE 2000
Nippon Convention Centre,
Makuhari Messe.
Contact: Japan Electronics Show Association.
Fax: +81 3 5402 7605.
Email: bee@jesa.or.jp
Net: http://bee.jesa.or.jp/

17–19
Music Live 2000
NEC Birmingham UK
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Net: www.ioa.org.uk

24–27
21st Tonmeistertagung VDT International Audio Convention
Hannover Congress Centre,
Hannover, Germany.
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Fax: +49 (0) 2204 21584.
Email: vdt@tonmeister.de
Net: www.tonmeister.de

December

5–7
JTSE 2000
Brussels Royal Opera, Brussels, Belgium.
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Fax: +33 (0)2 40 48 64 32.
Email. rak.productions@wanadoo.fr

2001

February

3–6
Middle East Broadcast 2001
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Contact: Overseas Exhibition Services.
Tel: +44 (0)20 7862 2046.
Fax: +44 (0)20 7862 2049.
Email: mebroadcast@montnet.com
Net: www.aeminfo.com.bh

13–16
Memex 2001
Dubai World Trade Centre, United Arab Emirates.
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International Conference and Exhibitions.
Tel: +44 1442 878 222.
Fax: +44 1442 879998.
Email: andy@ice-ltd.demon.co.uk

22–24
Broadcast Thailand 2001
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Bangkok, Thailand.
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Overseas Exhibition Services.
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Fax: +44 (0)20 7862 2068.
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Net: www.besmontnet.com

May

10–13
Showtech 2001
Messe Berlin, Messedamn 22
D-14055 Berlin, Germany.
Contact: Overseas Trade Show Agencies.
Tel: +44 20 7886 3106.
Fax: +44 20 7886 3101.
Email: james.eliot@montex.co.uk
Net: www.showtech.de

12–15
110th AES
RAI Conference and Exhibition Centre,
Amsterdam, The Netherlands.
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Letters

Character flaws

IN HIS REVIEW of the AEA R44C microphone (*Studio Sound*, August 1999), Dave Foister reiterates something that has become one of the myths of ribbon mics, that they do not have any intrinsic noise. They may not have any active electronics but the mere fact of the ribbon's electrical resistance means that they do indeed have noise, just like any other resistor. That's quite apart from any noise due to mechanical resistance (damping). In fact most ribbons, even with a perfectly quiet mic amp following, will have a noise floor considerably higher than that of most modern condensers: the character is different which makes exact comparison difficult but something in the region of 20dbA to 25dBA is typical. It's probably their biggest drawback—I say that as a regular user of the 4038.

On a different tack, the article on Ambisonic recording reminded me of an idea I had a while ago. Wouldn't it be nice if recordings could be made and distributed in native Ambisonic format, with all three dimensions appropriately encoded, and then decoded to match the speaker setup in use at the replay end? My dream would be to have a 'calibrate' mode built into the surround decoder which, with the addition of a microphone (wouldn't have to be terribly fancy), could actually measure the parameters of the listening room including speaker placements, and set up the decoding in software accordingly. The DSP to do this would be no more complex than that already built into many sub-£500 domestic surround processors.

Richard Black
100115.3701@compuserve.com

Summer madness

FURTHER TO MARY HARRISON'S letter on fairies (*Studio Sound*, August 2000), I am writing to you to test the hypothesis that you will publish anything you are sent during the lean summer months.

Yours faithfully,

Simon Croft

I am disappointed that you regard Mary Harrison's quest in this light. Perhaps you have not properly considered the difficulties facing a researcher working in such a peripheral field. If you have, you will surely appreciate the boldness of her decision to approach a professional audio magazine with such a request—I wonder if you would be so brave.

I for one, have provided Mary with comprehensive details of my own experiences of 'elementals' and I encourage you to do the same.

Tim Goodyer, editor

Wintelligence gathering

WITH REFERENCE to Martin Polon's piece, 'Mother, Jobs and Speed' (*Studio Sound*, August 2000) on the evolution of the PC, I am responsible for choosing a number of PC's for a primary public school with a new arts-based curriculum, and have my feet firmly planted in both of the main computer camps. The *PC Magazine* article mentioned reported that the only areas where the G4 outperforms PIII's is in the specific processes tested by Mr Polon's source article from the *San Francisco Chronicle*, and in rendering a QuickTime movie.

I am not trying to enter a debate about this but felt an obligation to convey some additional test data. My dilemma with the choice for the school is deciding what will be the best platform for teaching computer techniques, as well as well-established applications in both business and the arts. I expect I'll need to buy a blend of Wintels and Macs.

Keep up the great work on *Studio Sound*. It is, to me, the *National Geographic* of the audio industry. I consider it to be the only consistently reliable source of information.

Gary Hedden, GHL Audio, Franklin, TN

First, understand that the benchmarks used for my report were based on what is considered to be the best Win-To-Mac comparison by most, if not all, in the Mac community—Adobe Photoshop. Any other kind of comparison is considered by many to be compromised by the different techniques used to write the so-called same software for the two platforms.

Second, all of these comparisons are now invalid since Apple has upgraded all of it's full-standing G4 machines to dual-processor multiprocessing at this time (except for one model at the bottom end of the line). Although Mac OS 9.04 does not necessarily take full advantage of this change, as Mac OS X will at the beginning of next year (Apple tells us), the future potential for such dual processing is enormous.

Third, whether using a PC or a Mac, the processor speed already in place and the processor speed yet to come, can overwhelm current software—especially that software not specifically rewritten for the increased speed. It is kind of like buying a 'suburban assault vehicle' as opposed to a gas-electric car during a gas shortage. Both vehicles will get you there, but the larger vehicle with it's 12mpg consumption will require a second mortgage to pay your gas bills while the gas-electric will yield 70mpg. Bottom line is to make your computer-age application specific!

Martin Polon

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< Continued from page 98

MIDI Interface:
MOTU Micro Express [4-in, 6-out]
'I need a MIDI interface and this is one of the cheapest.'

Software:
SampleSearch
'This is a really good database with all of the functions. If you're using samples at all, or just files that have names, then it's just great to be able to file them. I tend to stash all of my samples on 230Mb M-O drive—it stands up pretty well and is fairly cheap.'

Backup:
DDS-3 DAT tape drive/Retrospect
Yamaha 8424 CD writer

Microphones:
Neumann M147; Shure Beta 58
Audio Technica 3525
'I only need a few microphones, given the limited range of things I'll be recording. Still, I think it's important to have one really good-quality, general-purpose, record-everything mic, and that's the M147 which is great for lots and lots of things. Then I also need a good dynamic mic, and the Beta 58 fits that bill, while the Audio Technica is dead cheap and sounds really good on certain things like acoustic guitars.'

Outboard:
Focusrite ISA215; Focusrite Compounder; TL Audio Fatman valve compressor; Alesis 3630 compressor; Lexicon MPX-1; Quadraverb GT (secondhand); Lexicon ALEX (secondhand); DeltaLab DL-4 (secondhand); Hughes & Kettner Red Box Pro; Line6 POD
'The dual-channel ISA215 is expensive, but it is really good for pretty much everything, so I thought it was worth putting that in.

'This is a list of dead cheap stuff that I think is really good, and the Alesis 3630 fits right into that category. It's a very comprehensive, basic, flexible compressor, and just something to put across a mix if you're doing rough mixes very quickly. The MPX-1, on the other hand, is a nice quality multi-effects unit, with flanging, reverb, and so on. I mean, I like virtually anything that Lexicon does—the reverbs are just richer and more solid-sounding than a lot of other stuff, particularly Japanese stuff.

FX pedals:
Lovetone Meatball; Lovetone Ring Stinger; Vox Valvetone distortion; Boss AW-2 Auto-wah; Sansamp Sherman Filterbank
'These are all modulators that I use as part of the analogue end of recording something.'

Monitoring:
Yamaha NS10s; Auratones
Quad 405 power amp
'I'm quite happy with NS10s. I use them all the time like most people do. I've tried lots of different small monitors of that size—like the Alesis and Genelecs—but the NS10s and Auratones do it for me. Of course, you can't buy the horrible-looking, brown little cube Auratones anymore, so if you know somebody who's a got a pair of those that they want to sell...'

Headphones:
Beyer DT150 (2)
Behringer Powerplay Pro amp

Acoustic design:
White Mark (David Bell)
'For my own room at home I've got some RPG panels—a series of panels that you hang on the wall for defracting, absorbing, whatever—and in the process of experimenting I've come into contact with David Bell of White Mark who's been quite happy to give advice at this sort of level with regard to a project studio environment. I think that, whatever room you've got, it's worth investing some money to improve things even just a little bit.'

Extras:
Custom Consoles Isomac; Speaker stands; Cables/adaptors; Speaker switch box; Racking
'Custom Consoles are based in Nashville and I've just ordered an Isomac because it seems like a really good piece of kit. Speak to any suppliers of Pro Tools about noise problems and the only answer you get is to bung it all in a rack and put in the machine room. The Isomac is box which has a hinged glass front door, acoustic foam inside and two big fans that are completely silent. With just one room you've got to have something like that, otherwise there's just too much ambient noise... although, of course, I'm ignoring the fact that I might be dealing with a domestic bedroom on a main street.'

THE BALANCE SHEET

Total expenditure: £28,952

And that's without a probable 10% to 12% discount off the retail price, meaning that Mick Glossop has a few thousand tucked into his back pocket to spend on whatever his heart desires...

ROCK 'N' ROLL ANIMAS

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Voodoo Chilli

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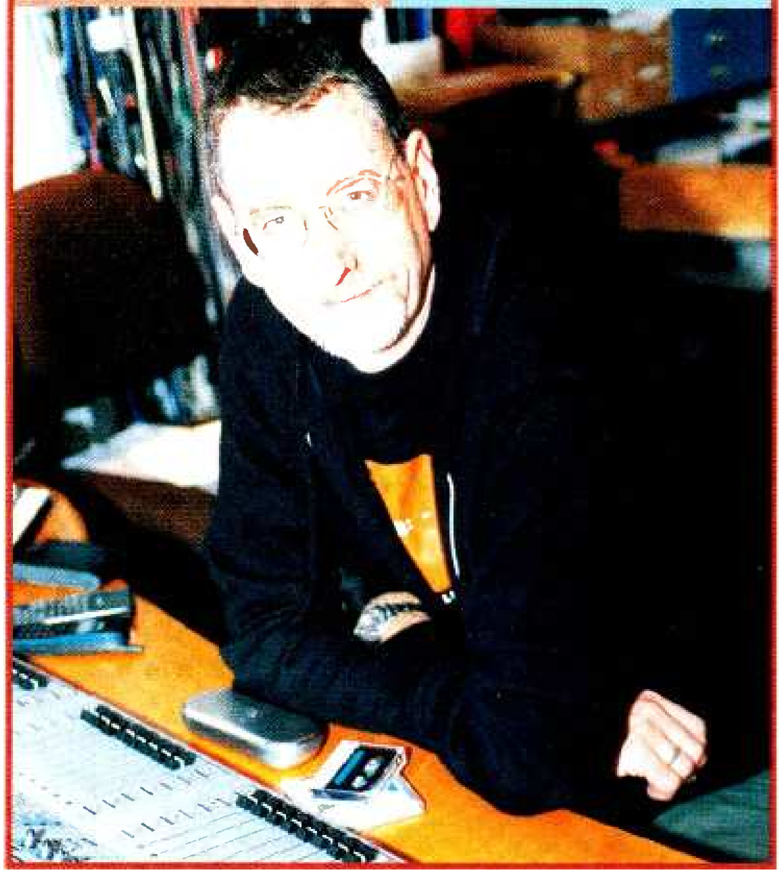
Ain't Nececelery So

MICK GLOSSOP'S PROJECT STUDIO

Discovering what goes on behind closed doors, **Richard Buskin** invites producer Mick Glossop to equip a private project room to compete with commercial studio values

THE TASK
Equip a professional personal project studio

THE BUDGET
£30,000



AS THE PRODUCER-ENGINEER of artists ranging from Frank Zappa, Van Morrison, Tangerine Dream and Flesh For Lulu, to Queen, Mott The Hoople, Mike Oldfield, Sinead O'Connor, The Skids, The Waterboys, The Wonder Stuff and The Men They Couldn't Hang, Mick Glossop knows a thing or two about cutting-edge technology. He played an integral role in the redesign of The Manor in Oxfordshire and The Townhouse in London when serving as chief engineer there during the seventies, and he also gained plenty of good on-staff experience at London facilities such as Wessex and Nova, as well as at Studio Son Quebec in Montreal. Freelance since 1980, Mick Glossop has spent the past two decades keeping abreast of technological advancement, be it MIDI programming, digital sampling or computer-based mixing.

Now, however, with just £30,000 to spend on 'a project studio that doesn't compromise in terms of sound quality, but, more than just demos, is capable of handling top-quality professional work,' he faces the ultimate challenge.

Workstation:
Pro Tools Mix System; SampleCell (32Mb RAM) + TDM board; 882/20 interface; Apogee PSX-100 ADA convertor; Apple Mac G4/400, 256Mb RAM, SCSI card; 17-inch monitor; Glyph Cheetah 9Gb hard drive (2)

'With £30,000 to spend on a professional-standard room—unless I'm going to get something like a secondhand A80, which would probably cost around £5,000—I have to go digital. What's more, you get so much with Pro Tools in terms of production values, it would be silly not to use it. I mean, I've been using my Pro Tools setup for about five years, and it's proved itself to me as a recording system, aside from anything else. So, that's what I'm opting for, because there's no point opting for an ADAT; there's not much you can do with it.

'Having said that, in trying to keep the cost down the best thing is to get the Mix rather than the Mix Plus because you save about £2,000, and while you get less DSP, with a G4 you've only got three PCI slots. One of them is going to be taken up with a SCSI adaptor card, so that means you've only got one slot left for SampleCell, and I think a sampler is important. I have used the SampleCell and I find it really convenient, because everything is there on the Mac, and with TDM you're going digitally into the system, which is great. Therefore, given that, if you want SampleCell and a Mix Plus you'll need an expansion chassis, and that's complicated and expensive.

'In terms of interfaces, given that, subjectively speaking, the Apogee convertors are better—and the Prism convertors are better still, but more expensive, I wouldn't feel happy using an 888/24. That's partly because of the A–D conversion, but even more importantly, if you're basing a project studio around Pro Tools then Pro Tools is everything, and that means the D–A conversion is an essential part of your monitoring system. So the quality of the two channels of D–A is almost more important than the A–D convertor, because you're making all of your decisions based on the way you're listening to that stereo. That's why I've gone for the Apogee PSX-100, and I can interface it with the dead cheap 882/20 which carries 24 bits. It's a slightly funny hybrid way of getting top quality in and out in terms of analogue, while a big assumption that is quite valid is that you only need two top-quality D–As and two top-quality A–Ds, because you're in one room, you're going to be recording everything in there, and you'll be very unlikely to need more than two recording channels.

'The G4 seems to be what you buy these days, and I couldn't believe how cheap a 17-inch monitor now is. As for the hard drive, I've put two 9Gb drives in there instead of one 18Gb drive, because, although that's more expensive, two drives give you a higher track count with Pro Tools. Technically you get 64 instead of 32, although realistically it's not as much as that, depending on how much editing there is.'

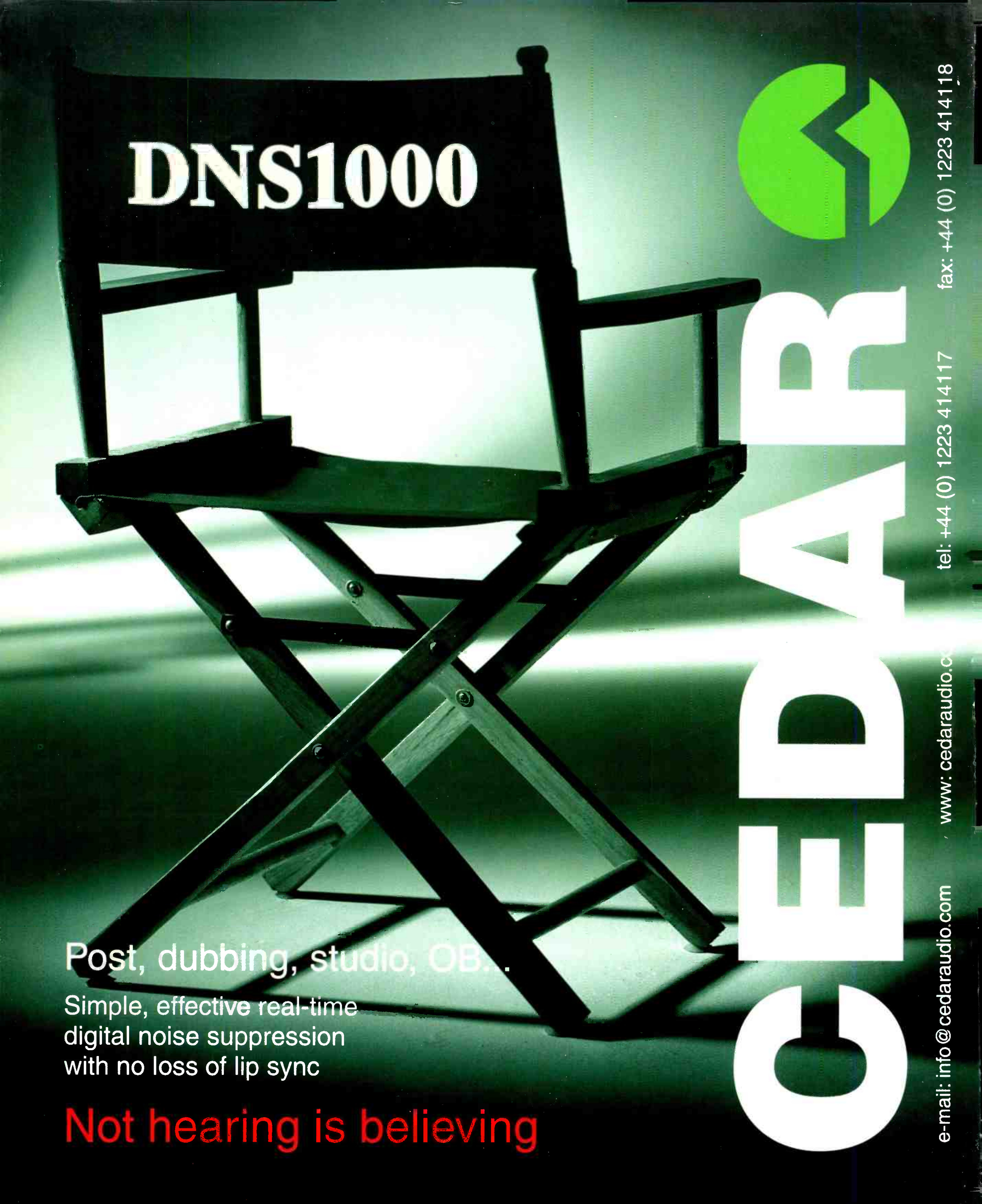
Plug-ins:
Serato Pitch 'n' Time; Auto-tune Waves Pro-FX; Amp Farm; MC DSP Compressor Bank; MC DSP Filter Bank; Sound Replacer

'Quality-wise, I think the Serato is the best pitch-shifting and time-stretching device around, while the Auto Tune is a necessity and the Waves Pro-FX has a nice collection of flangers and things like that. The Amp Farm is also nice to have. The Compressor Bank and Filter Bank have a definite character about the way they compress and filter, and I always prefer equipment that has a signature to it. In fact, the Compressor Bank must have about 50 factory presets that you can dial up, and the range of sounds is fantastic. You see, with Focusrite the plug-ins sound pretty much the same no matter how you use it, whereas the variations you get with the Compressor Bank are amazing, so it's really good value in that sense. So is the Filter Bank, which gives you a lot of control, and the Sound Replacer is pretty essential; it's a really, really good plug-in for replacing kick and snare sounds and all kinds of things.'

Controller:
CM Automation Motor Mix

'I've actually got four of these, and for the price they're tremendous value. In this case I've opted for one just to have some kind of hardware fader control over what you're doing. It would be nicer to have 32 faders.'

Continued on page 96 >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