EXCLUSIVES

Marantz PMD300-series
Lindos Electronics LSC24
Universal Audio UA1176
Microboards StartREC
Sennheiser MKH800
Manley monobloc
Lucid SRC9624
DAS Monitor 6

MX-2424

TASCAM MX-2424
Does this change everything?

The
STEVE
POWER
Interview

BROADCAST AUTOMATION: HANDS-OFF CONTROL
FUTURE COMPUTER: THE EVOLUTION OF THE PC
SURROUND SOUND: MIKING FOR SURROUND
HEADPHONES: CANS IN CLOSE-UP

Forgive and regret

OUR LOOK AT THE TASCAM MX-2424 serves to remind how delicate the relationship is between intermediate and milestone technologies. As someone who can remember DASH and ProDigi being dismissed in the 1980s as intermediate technology I could be forgiven for waiting for this machine for what has been an eternity.

Of course 'intermediate' is a word that is frequently confused and misused by those of shallow pocket who then proceed to power out of their personal handbrake turns as soon as they can afford to. Digital tape was, and is, an intermediate, but it hasn't stopped DASH representing the high-end interchange format in music recording or the DA-88 serving as an established post mix medium for long enough to pay off the financing deals for shrewd operators many times over.

What wide-scale technology adoption patterns all have in common is that they either facilitate efficiency and convenience or they are simply so cheap and simply make too much sense to ignore. The acid test of business plan justification and qualification generally throws up a lowish risk, no-brainer decision that slots in seamlessly alongside existing systems and work methods that don't immediately have to be consigned to scrap.

The MX-2424 is not the first to offer affordable high track numbers on hard disk, but it remains significant in doing it quite so well for quite so little. We live in an increasingly disposable society and a major attribute of many recent 'milestone' developments has been a lack of emotional involvement and dependence for the user. It's why we can recount and indulge ourselves in the finer points of analogue master tape machine ownership in this same issue despite the fact that we are unlikely to ever enjoy a similar attachment to the early incarnations of DTRS or, for that matter, early attempts at DAWs.

That change continues is not the surprise and not the issue, that is discards it predecessors, themselves milestones in their own day, quite so quickly takes an adaptation of attitude and a hardening of outlook. Increasingly we are being stripped of the technological crutches that have previously tied us to the world. We are left instead with our own transferable crafts and skills as the truly salable commodity. The most important thing you take into the room is what you have in your hand.

Zenon Schoepe, executive editor

Covering fire

COMPARING MUSICAL NOTES with Philip Newell in Spain recently threw up an interesting difference of opinion. Where I had always regarded the term 'cover version' to apply to any reworking of another artist's song, no matter how greatly it differed from the original, Philip considered it applicable only to a fairly faithful copy. To illustrate his point he quoted Cilla Black's 'Anyone Who Had a Heart' as being a legitimate (read 'near identical') cover of Dionne Warwick's original, while Joe Cocker's interpretation of the Beatles' 'With a Little Help from My Friends' was another proposition entirely. I had to agree that he had a point.

Returning home, I chanced across a copy of Ann Harrison's book *Music: The Business* (Virgin Publishing, ISBN 0 7535 0433 2), whose glossary contains the following: 'Cover version: a song written and recorded by someone else, that an artist records a new version of.' As the book is subtitled The Essential Guide to the Law and the Deals, I thought Philip should know...

My email prompted a fax reply that closed 'Joe Cocker took a weak insipid ditty... and rearranged it into a monstrous rock classic which wiped the floor with the original. That, to me, is not a cover version but a huge development of the song—a new original interpretation'. Philip was sticking to his guns.

So whose definition of 'cover version' were the scriptwriters of *The Patriot* and *U-571* working to when they decided to rewrite history? For those not in the know, in telling the story of the American war of Independence, *The Patriot* (directed by Roland Emmerich and starring Mel Gibson) inaccurately casts the British as a murderous bunch, massacring their enemies in the manner of the Nazis. *U-571* (directed by Jonathan Mostow and starring Matthew McConaughey, Bill Paxton, Harvey Keitel and Jon Bon Jovi), meanwhile, transfers the credit for the capture of the first German Enigma machine and code books from the British to the Americans during WWII.

In this case I defer unconditionally to Philip's view; the films might be cast as the development of a theme, but a good, old-fashioned cover version would have been infinitely preferable. **Tim Goodyer, editor**

www.americanradiohistory.com

A new generation of
has just

Midge Ure

VM7200/7100
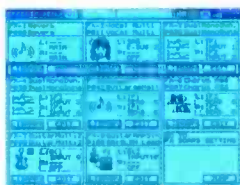
Up to 16 stereo or 32 mono effects, plus master effects, derived from the award winning V-studio range.
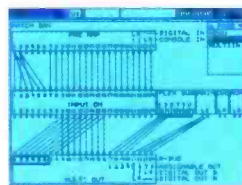
The virtual patchbay offers unrivalled flexibility. Route any of the inputs and outputs internally, with not a patch lead in sight.

Real-time spectrum analyser with noise generator and oscillator. Eq the room and eliminate feedback instantly.

4-band parametric eq. Full parametric control on hi, mid and low-mid. Also includes high pass filter, and full per-channel dynamics.

■ Tokyo-based Yamaha Epicurus Studios has ordered a 64-channel SSL SL9064L analogue console. The second j-series for the facility comes as a response to growth in demand for 5.1 mixing Japanese interest in Euphonix' CS3000 digitally controlled analogue consoles for music and post applications has been high with the V Vision postproduction house installing a 32-fader CS3000 in a new studio. producer Yota Tsuruoka's Rakuonsha music facility ordering a 48-fader CS3000D-3-48, and Osaka's Studio Aqua project and multimedia studio ordering a 72-fader CS3000-4-72 console for games work. Rakuonsha specialises in Animé (Japanese animation) and has recently produced soundtracks for Worner Gundam and sound effects for the film and TV versions of the popular Pokemon series.

Yamaha Epircus Studios, Japan.
Tel: +81 3 3719 910.
SSL, Japan. Tel: +81 3 5474 1144.
Euphonix, US. Tel: +1 650 855 0400.
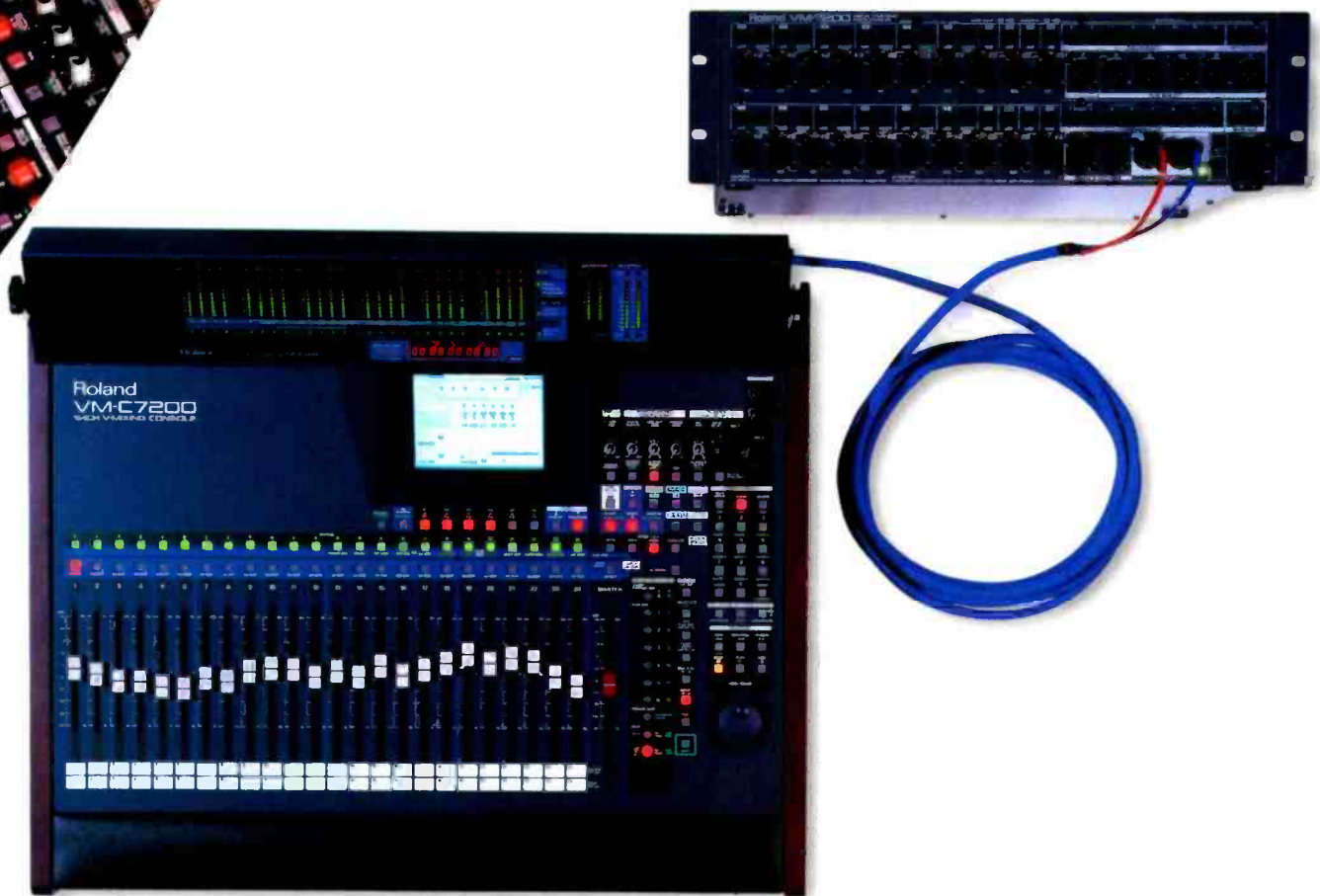■ New York's broadcasters have been quick to take up the new AMS Neve Libra Live Series II digital console, with Time Telepictures taking two for live-to-tape audio production, MTV Networks ordering one for Total Request Live and studio music performances, and HBO Studio Productions readying a new digital control room for theirs.

AMS Neve, UK.
Tel: +44 1282 467011.
■ London post house Molinaire is to take two Soundtracs DPC-II digital production consoles as part of the refurbishment of Suites A and B where they will work in conjunction with AudioFile editors on DVD, film, TV drama and documentaries. UK-based postproduction and duplication company 10th Planet has taken delivery of a SADiE 24:96 digital audio editor for its recently-completed West End studio facility. The system has already seen use mixing and editing the music for the Dancestar 2000 Woldwide Dance Awards composed by in-house composer Jon Voda.

Molinaire, UK.
Tel: +44 20 7478 7000.
10th Planet, UK.
Tel: +44 20 7637 9500.
Soundtracs, UK.
Tel: +44 13 7284 5600.
SADiE, UK. Tel: +44 13 5364 8888.
■ Australian Tiger Recording, located in the fashionable McMahon's Point district of Sydney has installed a 72-fader Euphonix CS3000 console in its main Studio A control room. The facility produces original music for the city's thriving television advertising industry alongside 'a hybrid of advertising work and major-label album projects'.

Euphonix, US. Tel: +1 650 855 0400.
■ Italian sound engineer Daniele Giordana has upgraded his One Voice studio in Chivasso, Turin with Digidesign Pro Tools Mix Plus v5 running on an Apple G4/350. The system includes Digidesign 888 and 882 audio interfaces and a number of plug-ins. Giordana's old Pro Tools III system will be used for writing, editing and preproduction.

One Voice Studios, Italy.
Tel: +39 011 910 6211.
Digidesign, US.
Net: www.digidesign.com
■ Los Angeles-based Warner Brothers Postproduction Services has chosen a 48-fader AMS Neve Libra Post console for its Studio C stage. The digital desk will be used for film and trailer mixing.

Warner Brothers, US.
Tel: +1 818 954 3723.

AMS Neve, UK.
Tel: +44 1282 457011.
■ Scottish audiophile label Linn Records is the first customer for the new DPA Type 4041 large-diaphragm mic. In the three months since their arrival, Philip Hobbs has used the 4041s to complete more than 20 classical recordings beginning with a little-known piece by Buxtehude for choir and orchestra. With the 4041s as the main pair, the anticipated 90-minutes balance time was reduced to 15 minutes.

DPA, Denmark.
Net: www.dpamicrophones.com
■ German postproduction facility Voodoo Lounge Medienproduktions has installed two Martinsound Multi-MAX monitor controllers at its Grünwal premises. The controllers are being used to manage a Quested surround system and Yamaha NS-10 close-fields in Studio A, and a Genelec 1032A with 1092A subwoofer surround system in Studio B. Both studios use Digidesign Pro Tools 24 workstations and Pro-Control control surfaces. Voodoo Lounge specialises in the creation of on-air advertising and promotional spots in surround sound. Two MultiMAX units have also been installed in identical film mixing studios at Soundtrack in Linz, Austria. Both rooms are centred on Digidesign 24 Pro Tools systems and use Genelec 1037B surround sound monitor systems with 1094A subwoofers.

Martinsound, US.
Net: www.martinsound.com
■ Slovenian state broadcaster RTV Slovenia has taken a fourth SSL Aysis Air console for installation in a new TV OB vehicle. The purchase follows the earlier installation of an Aysis Air into a radio OB vehicle and Aysis Airs in RTV Slovenia's television headquarters and the radio control room at the Cankarjev Dom international concert hall.

SSL, UK. Tel: +44 1865 842300.
■ Baltimore-based facility Tonal Vision Studios has purchased a Fairlight MFX3plus workstation for use in the creation of original sound for picture. One of the Baltimore area's most respected and busiest sound-for-picture facilities, Tonal Vision specialises in work for television shows.

Fairlight, Australia.
Net: www.fairlightesp.com
■ British-based Richmond Film Services, Dreamhire, Tiger Hire, Yorkshire Audio have been adding to their hire stock. While RFS has taken a number of Audio Technica's new AT895 mic for sport broadcast use, Dreamhire has added a second Sony PCM-3348HR digital machine to its stock, bringing the company's total of professional multitrack recorders to 14 and Tiger Hire has invested in a 44:24 Soundcraft Five monitor console in readiness for Radiohead's imminent tour. Meanwhile, the Yorkshire Audio has taken delivery of eight Crown MA-3600 VZ power amplifiers to drive its EAW KF-850 rig.
■ UK's Pearson Television, the production company responsible for ITV's police drama, The Bill, has installed two pairs of DynaudioAcoustics M3 passive main monitors and five pairs of active BM6A close-fields as part of a refit of existing studios and building of new rooms. Pearson's priority lies with dialogue and identifying locations sound problems, for which the speakers were regarded the best choice.

Pearson, UK. Tel: +44 20 7691 6000.

hDVD

US: The forthcoming hDVD release of HDPlanet will offer a tour of the Grand Canyon at dawn, a trip down Route 66 and journey to the moon via footage captured during a recent voyage by the STS-93 Space Shuttle Columbia courtesy of Sonic Solutions. HDPlanet, a collaboration between Chicago radio legend Steve Dahl (WCKG-FM), award-winning television journalist Bill Kurtis and record producer Joe Thomas, combines HD video of nature and travel with music performances by musicians including Joe Walsh, Christopher Cross and Jimmy Webb on a new format that supplements DVD. An extension to DVD, hDVD adds one or more of the high-definition MPEG2 video formats specified by the Advanced Television Standards Committee offering more than 4x the resolution of standard DVD-V titles.

'It's only fitting that this ground-breaking, High-Definition video and musical performance be published using the ground-breaking new hDVD format,' said Mark Ely, director of product marketing for Sonic. 'hDVD technology will be used to distribute the same 1080i video content from the live performance, combined with a high-quality uncompressed multichannel soundtrack. The resulting hDVD disc will be playable on a 650MHz or faster Pentium III PC with a DVD-ROM drive using a software-based hDVD video player.'

To take advantage of the higher-resolution and higher bit-rate video material, Sonic Solutions has extended the DVD multiplexing and formatting engine used in Sonic DVD Creator, DVD Fusion and DVDit! to accept and author High-Definition MPEG-2 video content. Sonic plans to ship hDVD-ready authoring systems in conjunction with hDVD software players in autumn 2000.

▲ **UK: Construction is under way on a new London recording complex, Sphere Studios, being described as 'the most ambitious music recording complex to be built in the capital for many years'. The new studio hopes to rival London's finest and will offer three main control rooms dedicated to tracklaying, stereo mixing and 5.1 surround mixing. There will also be six white rooms available for hire to artists and independent producers, plus extensive ISDN facilities, plenty of parking and a licensed bar. Behind Sphere are musician and producer Francesco Cameli and Malcolm Atkin, formerly general manager of Sir George Martin's Air Lyndhurst facility. The design of the main studios and white rooms is being handled by Munro Associates while Cameli and Atkin, with the assistance of Sphere's technical manager Tom Schlum, are currently finalising the technical specification for the three studios. Atkin says: 'In order to develop the music industry on a global basis there is a genuine requirement for recording facilities that can offer the most up-to-date technical solutions. With Sphere we are in a new-build situation and therefore we have the opportunity to develop a facility that is truly designed for the future. The construction of the complex and the equipment we install will reflect our intention to offer facilities that far exceed those currently on the market in the UK.' By September 2000, all six white rooms will be up and running, with the main studios opening in Spring of 2001. Pictured are (L-R) Francesco Cameli, Tom Schlum, Malcolm Atkin. Sphere Studios, UK. Tel: +44 20 7924 4550**

▲ US-UK: A new western starring Jackie Chan is set to become a modern classic. The film mix for *Shanghai Noon* was undertaken by Andy D'Dario at Buena Vista Sound on an AMS Neve DFC console who took particular pride in the finale fight scene wanting 'to make the sound of every weapon distinctive'. Randy Edeleman's score was engineered at London's Lansdowne studios by Steven Lewis also on an AMS Neve desk, this time the analogue VXS72

Webcasting

France: A collaboration between IDT, leading French manufacturer of broadcast sound-processing technology. Internet broadcaster TV-Radio.com subsidiary of Télédiffusion de France served the French International Radio Show with a 'broadcast quality' webcast. Using a prototype of its forthcoming Digital Virtual Processor @ Net unit, IDT delivered around 50 radio channels over the Internet to the visitors at Paris' Palais des Congres. When released, the processor will feature functions specially adapted to high-quality webcasting and is claimed to be the first Internet processor to abandon multiband working in favour of FFT.

Net: www.idt-fr.com
www.tv-radio.com

Custom 'fex

UK: Following design projects for the BBC World Service, Channel 4 TV, Danish Radio, Northern Telecom and the Finnish Parliament, Sonifex is offering bespoke design and manufacture services alongside its current portfolio of proprietary products. Julian Speed, export sales manager for Sonifex commented 'Project work of this nature has generally been undertaken as a result of clients or distributors approaching the company direct. Some customers, therefore, are either not aware of the design and manufacturing services that we can offer, or have looked elsewhere and found the process to be too time consuming and expensive. As we have already been successfully, but informally, offering this service for sometime, the logical next step is to formally market it alongside our current product portfolio.'

Net: www.sonifex.co.uk

Super DVD

Germany: Syrinx Music & Media, Audionet-Idektron (www.audionet.de) and Frequenzwerkstatt have jointly announced the release of the Super DVD recording of Retold-Hamamura Quintet (Audionet Standards no.1) at Frankfurt's High End 2000 show. This is claimed to be the first DVD to meet both the high-resolution DVD-A standard as well as the DVD-V standard. The disc can be read in an ordinary DVD-V player, with the consumer able to choose between 96kHz/24-bit stereo, 48kHz/16-bit stereo, or 5.0 Dolby Digital multichannel. The same disc in a DVD-A or universal player also offers 192kHz/24-bit stereo or 48kHz/24-bit 5.0 multichannel.

The recordings were made in the Milchkettenmusik Studio in Hamburg, with Syrinx in charge of recording, audio postproduction, screen design. DVD-premastering and providing the central elements of the production system. Overall production was in the hands of Frequenzwerkstatt. Since the new format is compatible with the 'old' DVD-V standard, the audiophile market-segment can be served without additional replication costs.

Net: www.syrinx.de

Logical development

US: Dolby Laboratories has unveiled Pro-Logic II, narrowing the gap between the original Pro-Logic matrix developed during the mid eighties and digital 5.1-channel systems. 'Pro Logic II lets consumers enjoy [Pro-Logic] programmes with a convincing 5.1-like presentation,' claims Dolby, with its eye on the Pro-Logic catalogue. Pro Logic II is intended to convert conventional stereo recordings into 'a natural, believable surround experience'.

This system was invented by Jim Fosgate, who has been involved in surround decoding technologies since the late sixties. Fosgate said, 'I have spent the past 25 years figuring out how to expose the hidden information in 2-channel stereo recordings, both new and old. This breakthrough in matrix decoding technology allows users to enjoy all their existing 2-channel programmes, whether Dolby Surround encoded or not, with an enhanced level of spatiality and directionality.' Pro Logic II decoding can be implemented in analogue and digital circuitry, intended for use in home cinema and new music surround products.

▲ Czech Republic: The last CEDAR Series 2 unit has been purchased by Sound Studio (pictured) at the Music Faculty at the Academy of Performing Arts in Prague to work alongside its DC-1 De-Clicker. The popular Series 2 line is being discontinued after six years in production as some of the hardware components are now obsolete, leaving CEDAR to regard it as no longer practical to build more systems but to retain remaining component stock to ensure support for existing units. Series 2 users include The US Library of Congress, the British Library National Sound Archive, Abbey Road Studios, Skywalker Sound, Warner Hollywood, Disney, and TV and radio stations in Russia, Indonesia, Egypt, and Botswana. CEDAR Audio, UK. Net: www.cedar-audio.com

August

23–26
BIRTV 2000
China International Exhibition Centre, Beijing, China.
Contact: P&O Events.
Tel: +44 (0)20 7370 8231.
Email: shanghai@eco.co.uk

September

4–7
IECEP 2000
Philippine International Convention Centre, CCP Complex, Roxas Boulevard, Manila, Philippines.
Contact: Overseas Exhibition Services Ltd.
Tel: +44 (0)20 7862 2090.
Email: philippines@montnet.com

8–12
IBC
Amsterdam, The Netherlands.
Contact: International Broadcasting Convention.
Tel: +44 (0)20 7611 7500.
Fax: +44 (0)20 7611 7530.
Email: show@ibc.org
Net: www.ibc.org

10–13
Plasa
London. UK.

16–20
Cinec 2000
MOC Events Centre. Munich-Freimann, Germany.
Contact: Messe München.
Tel: +49 89 94901.
Email: info@messe-muenchen.de
Net: www.messe-muenchen.de

22–25
109th AES
Los Angeles Convention Centre, Los Angeles, California, US.
Contact: Chris Plunkett.
Tel: +1 212 661 8528.
Email: 109th_exhibits@aes.org
Net: www.aes.org

October

17–18
Broadcast India 2000
Centrum Centre 1. World Trade Centre, Mumbai, India.
Contact: Kavita Meer. Saicom Trade Fairs and Exhibitions.
Tel: +91 22 215 1396
Fax: +91 22 215 1269.

Email: saicom@bom2.vsnl.net.in
Net: www.saicom.com/broadcastindia.

November

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
Hong Kong, China.

17–19
Reproduced Sound 16
Stratford Victoria Hotel, Stratford upon Avon, UK.
Contact: Institute Of Acoustics.
Tel: +44 1727 848195
Fax: +44 1727 850553.
Email: ioa@ioa.org.uk
Net: www.ioa.org.uk

24–27
21st Tonmeistertagung VDT International Audio Convention
Hannover Congress Centre, Hannover, Germany.
Contact: Gisela Jungen, VDT.
Tel: +49 (0) 2204 23595.
Fax: +49 (0) 2204 21584.
Email: vdt@tonmeister.de
Net: www.tonmeister.de

2001

February

3–6
Middle East Broadcast 2001
Bahrain International Exhibition Centre, Bahrain.
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.
Contact: Andy Drew.
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
Queen Sirikit National Convention Centre, Bangkok, Thailand.
Contact: Adam Ridgway, Overseas Exhibition Services.
Tel: +44 (0)20 7862 2069.
Fax: +44 (0)20 7862 2068.
Email: thailand@montnet.com
Net: www.besmontnet.com

May

12–15
110th AES
RAI Conference and Exhibition Centre, Amsterdam, The Netherlands.
Contact: AES.
Tel: +1 212 661 8528.
Email: 110th_exhibits@aes.org
Net: www.aes.org

June

19–22
Broadcast Asia and CommunicAsia 2001
Singapore Expo, Singapore.
Contact: Overseas Exhibition Services
Tel: +44 (0)20 7862 2080.
Fax: +44 (0)20 7682 2088.
Email: broadcastasia@montnet.com
Net: wwwbroadcast-asia.com

20–23
Musikmesse Prolight & Sound
Lenexpo Exposition Grounds, St Petersburg, Russia.
Contact: Hubert Demmier.
Messe Frankfurt Trade Fairs
Tel: +370 2 250 726.
Fax: +370 2 233 800.
Email: hubert.demmier@lithua-nia.messefrankfurt.com
Net: www.musikmesse.de

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Tel: 0043 800 201 651 Fax: 0043 800 201 651

Belgium S: EML
Tel: 00 32 11 23 23 55 Fax: 00 32 11 23 21 72

Brazil P: Proware Audio Innovations
Tel: 00 55 11 55052866 Fax: 00 55 11 5594 6586

Canada S: digidesign
Tel: 00 1 800 333 2137
www.digidesign.com

Chile P: Clio Produtores Musical
Tel: 0056 2 274 9621 Fax: 0056 2 298 9575

China/Hong Kong S: Digital Media Technology (DMT)
Tel: 00 852 2721 0343 Fax: 00 852 2366 6883

Croatia P: Slovenia, Bosnia, Macedonia & Serbia: Music Export (Germany)
Tel: 0049 89 746 12390 Fax: 0049 89 746 12392

Denmark S: New Musik AG
Tel: 00 45 86 190899 Fax: 00 45 86 193199

Finland P: Studiotec KY
Tel: 00 358 9 5123 5300 Fax: 00 358 9 5123 5395

France S: DM2I Audio Solution
Tel: 00 33 1 48 63 04 43 Fax: 00 33 1 48 63 78 09

Germany P: TC Electronic Deutschland GmbH
Tel: 00 49 40 531 08 390 Fax: 00 49 40 531 08 388

Greece S: Bon Studio
Tel: 00 30 1380 9605 Fax: 00 30 1382 7868

Holland P: The Netherlands: Total Audio
Tel: 0031 20 467 6467 Fax: 0031 20 467 6464

Iceland S: Exton
Tel: 00354 551 2555 Fax: 00354 562 6400

India P: R S I Electronics
Tel: 0091 22 636 9147 Fax: 0091 22 636 9681

Ireland S: CTI - Control Techniques
Tel: 00 3531 454 5400 Fax: 00 3531 454 5726

Israel P: Sontronics
Tel: 00 972 3 570 5223 Fax: 00 972 3 619 5297

Italy S: Grisby Music Professional
Tel: 00 39 0 71 710 8471 Fax: 00 39 0 71 710 8477

Japan P: All Access
Tel: 00 81 52 443 5537 Fax: 00 81 52 443 7738

Korea S: Best Logic Sound Co
Tel: 00 82 2 515 7385 Fax: 00 82 2 516 7385

Mexico P: Vari International S.A. de C.V.
Tel: 00 525 604 6946 Fax: 00 525 605 9656

New Zealand S: Protel
Tel: 00 64 4 801 9494 Fax: 00 64 4 384 2112

Norway P: Lydrommet
Tel: 00 47 22 80 94 50 Fax: 00 47 2 80 94 60

Poland S: Hexagon Warszawa
Tel: 0048 22 44 66 99 Fax: 0048 22 44 83 35

Portugal P: Caius Tecnologias
Tel: 0035 12 318 4456 Fax: 0035 12 208 6009

Russia S: AT Trade
(Russia) Tel: 7 095 956 1105 Fax: 7 095 956 6882

Singapore P: Team 108
Technical Services Private Ltd
Tel: 00 65 748 9333 Fax: 00 65 747 7273

Southern Africa S: Powerhouse Electronics
Tel: 00 27 11 728 3102 Fax: 00 27 11 728 6789

Spain P: Media Sys.
Tel: 0034 93 424 6500 Fax: 0034 93 424 7337

Sri Lanka S: HiFi Centre Ltd
Tel: 00 94 1 580442 Fax: 00 94 1 503176

Sweden P: Polysonic
Tel: 00 46 31 7000000 Fax: 00 46 31 706 9110

Switzerland S: Bleuel Electronic AG
Tel: 00 41 1 751 7550 Fax: 00 41 1 750 1153

Taiwan P: Digital Media Technology (DMT)
Tel: 00 866 2 2516 4318 Fax: 00 866 2 2515 9881

Thailand S: KEC
Tel: 00 662 222 8613/4 Fax: 00 662 225 3173

Turkey P: Imaj Music Yapim VE Organizasyon Ticaret A.s.
Tel: 00 90 212 216 7180 Fax: 00 90 212 216 7185

UK S: Focusrite Audio Engineering Ltd.
Tel: +44 (0)1494 462246 Fax: +44 (0)1494 459920

USA P: digidesign
Tel: 00 1 800 333 2137
www.digidesign.com

Venezuela S: C&M Audio
Tel: 00582 265 8780 Fax: 00582 267 4319

Vietnam P: Victor
Tel: 00 84 4 8 824 3058 Fax: 00 84 8 8 825 0099

www.focusrite.com

www.americanradiohistory.com

Old gold

I ESPECIALLY ENJOYED the June edition of *Studio Sound* because of the tribute to Dick Swettenham and the excellent review of the Helios EQ1 by George Shilling.

In that article reference is made to Tony Arnold's design being based on the early modules of Eric Clapton's console. Interestingly, that particular console is alive and well and working in a busy 'vintage' studio in South Norfolk. We restored the console and it has been worked hard for four years now without a single fault.

I owned two Helios consoles with identical mic-EQ modules the other being the old Ronnie Lane mobile. So the modules I sent to Tony Arnold could have belonged to either console—sorry to be an anorak. How about a follow-up article showing the old desk as it is now, working well and sounding fab?
Kevin van Green

Masterworks

IN REPLY to the article 'Masterclass—Tape Heads' (*Studio Sound*, March 2000) and the remarks of Mr Summers in 'Letters' (*Studio Sound*, May 2000), we would like to point out that Studer still makes and delivers original heads for all its 2-inch, 24-track machines. Studer even offers original heads or replacement heads for most of the older machines and for all standard formats (from ¼-inch to 2-inch) like the Studer B62.

Unfortunately, the article and the chart in the article contained some misinformation that found its way into it due to a misinterpretation at Studer. We would like to clarify this situation and state the correct details: 2-inch, 24-track heads are available for all machines and their versions, from the Studer A80 Mk.II to Mk.IV, A800 Mk.I to Mk.III, A820 to the Studer A827, including the Gold Edition.

2-inch, 24-track heads:

Machine Type	Record Head	Reproduce Head
A80 Mk.II, Mk.III, Mk.IV	1.317.184.00	1.317.185.00
A800 Mk.I, Mk.II, Mk.III	1.317.380.00	1.317.385.00
A820		1.318.780.00	1.317.785.00
A827	1.318.780.00	1.318.785.00

Note: The Studer A80 Mk.I was never built in a 24-track version.

The article states that the exact replacement for 1.316... heads may not be available anymore. In fact, most of them have been replaced by the 1.317... series of heads offering much better performance and less wear. It therefore is possible to improve the performance of an older machine by mounting the new 1.317... heads. The cleaning fluid which should be used for the heads caused another question. Owners of original Studer heads are allowed to use both, methylated sprit as well as isopropyl alcohol (IPA), for cleaning heads. There are no restrictions like with some of the non-original head designs.
Any further queries are welcome at:
Studer Professional Audio AG,
Email: nicolas.boehmer@studer.ch
Tel: +41 1 870 75 11.

Fairy stories?

I AM THE AUTHOR of many books, several dealing with the unexplained such as *The Children That Time Forgot* and *Mystic Forces*. I am now researching my new book that looks at ELEMENTALS, creatures from a different sphere of existence, commonly referred to as fairies, elves, and so on. I have collected some fascinating stories, for example David, a perfectly sober normal man, was walking along a riverbank one sunny afternoon, leading two horses back to his stables, when he heard a distinct voice coming from the stream. He looked down and to his amazement he saw a tiny man (about a foot high) standing on a stone in the water, dressed in trousers, boots and shirt, remarking in an ordinary manly voice that there had been no fishing that morning. He continued about the rain and how he hoped it would be better next day and how were they expected to survive without good fishing. He then added that he shouldn't complain as he wasn't bad off as some of the others! David watched him turn round as if looking out for something but the moment he made eye contact with the entity it vanished. The horses were disturbed and remained agitated until they were safely back in their stables. I would love to hear from you if you have had any similar experience.
Mary Harrison, 12 Thirlestane Road, Northhampton NH4 8HD.

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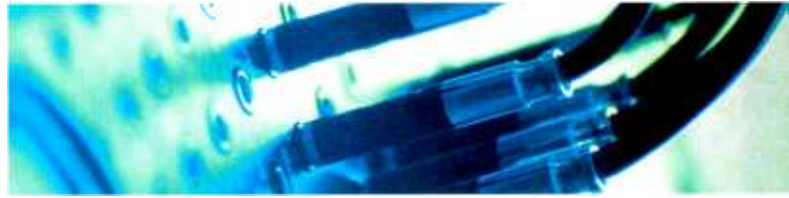
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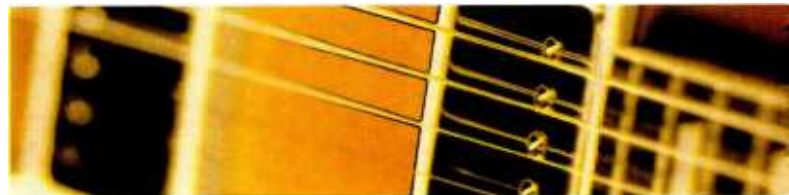
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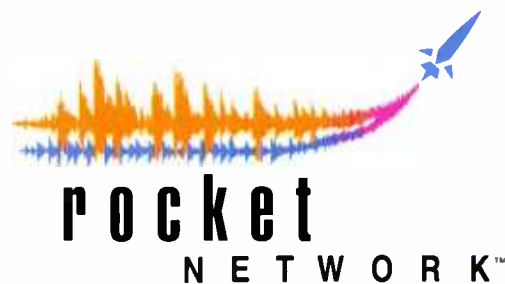
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Tascam MX-2424

Offering possibly the first genuine step beyond the digital tape machine Tascam's new hard-disk recorder-editor attracts the attention of **Rob James**

THE PRO-AUDIO LANDSCAPE usually changes slowly and steadily. Incremental improvements mark progress that often only really becomes apparent over a period of years. Occasionally, however, something comes along that really changes the world, almost overnight, eclipsing other aspirants to the 'revolutionary product' crown. Tascam has turned these out before: the DA-88 and DTRS format ushered in the age of affordable professional digital recording. Alesis' ADAT had the same effect in some areas, but it is DTRS that became common currency for programme interchange in film and sound-for-picture. It is just possible Tascam can repeat the trick with the MX-2424. The machine alone is impressive enough to warrant close attention, but the implications inherent in the control and networking capabilities make it potentially astounding.

The MX-2424 is a 24-bit, 24-track hard-disk recorder-editor when run at 44.1kHz or 48kHz; 24-bit, 96kHz operation halves the number of tracks. The machine comes with an internal 9Gb SCSI hard drive, 2-channel digital I-O, sensible audio interface options and a range of sync connections. Unlike the earlier MMR-8 Dubber, it uses a dedicated chip-set rather than a PC motherboard and is ready to use less than 15s after switching on.

Two distinct modes of operation are supported. In TL-Tape mode, the MX-2424 functions like a tape recorder. Recordings are destructive and there is no undo function. When a project is created in TL-Tape mode, an empty audio file is created for each track for the entire project length. Recordings and punch-ins to pre-existing recordings edit the relevant audio file(s). If tape-style working suits the job in hand, there are several advan-

tages: the disk space is fixed so that if there is enough space to create the required length of project, you know you won't run out. There is also virtually no waiting for housekeeping after intensive punching in and out on 24 tracks. Non-destructive mode allows for changes of mind via the UNDO and REDO keys. All recordings are kept even where they are entirely 'overwritten' and effectively 'orphaned'. Repeatedly rerecording the same section over 24 tracks rapidly uses up disk space and multiple consecutive punch-in and outs over 24 tracks will result in a subjectively lengthy wait for housekeeping once the transport is stopped.

There can be up to 999 tracks per project. Any 24 of these can be played back. Tracks can be freely loaded and unloaded even between projects. At present the files recorded by the machine are SDII (Digidesign Sound Designer II) on Macintosh formatted disks. The alternative of Broadcast WAV files onto PC FAT32 formatted disks should be available soon. OMF support for exporting projects to Pro Tools is also being investigated.

The EDL format used by the MX-2424 has been dubbed 'Open TL' or Open Track List. Tascam is making this available to any manufacturer who wishes to use it. Projects can be backed up to a variety of supported media including DVD-RAM disks and Travan tape streamers.

Although the MX-2424 will function perfectly happily as a stand-alone machine, the ViewNet Audio MX software package offers easier setup, provides a moving track display, enhanced editing capabilities and can control up to 65,000 machines across a network. Up to 32 machines can be connected together via the TL bus for sample accurate

sync and a possible total of 768 tracks. There is also a remote control, the RC-2424 that duplicates all the front-panel keys and adds dedicated editing keys and a macro function.

There are ten banks of menus to navigate, although only nine are currently in use. Setup accesses the setup and configuration menus and banks may be accessed directly by pressing a number key. Individual menus are found by using the Up and Down keys or by scrolling with the jog wheel. To change a parameter you press TRIM, adjust the parameter using the Up and Down keys or jog wheel. The shuttle ring moves the cursor where appropriate. Changes are confirmed by pressing STORE —it sounds more complicated than it is. I found it reasonably intuitive, but much preferred using the pull-down menus in the ViewNet software.

Although MX-2424 is clearly intended to be used with a mixer there are useful input

routeing options. The 2-channel input can be routed to record tracks in blocks of eight. So for example, Tracks 17–24 might be fed in odd-even pairs from this input while Tracks 1–8 and 9–16 are fed from one of the multi-channel option cards. Each block of eight tracks can either take the same numbered analogue or digital inputs—Tracks 1–8 take analogue or digital inputs 1–8, so on—or each block can be fed from analogue or digital Inputs 1–8, so Tracks 1–8, 9–16 and 17–24 all take analogue or digital Inputs 1–8. This will be particularly useful when working with the common 8-bus consoles. Outputs are always in a one to one correspondence with tracks.

Tascam has covered most of the bases. Internal sample rates are currently 44.1kHz and 48kHz plus the pull-ups and pull-downs for drop frame working. Time code is auto-matically switched to the correct rate between 24, 25 and 30fps when chasing SMPTE. However, the drop frame variations on 30fps must be manually selected. The rate must also be set manually for MTC chase. The MX-2424 will not chase one frame rate and generate another. This requires an external time-code gearbox. Varispeed of up to ±12.5% is possible, although the output sample rate will vary accordingly.

TL bus enables up to 31 machines to chase a single master MX-2424 without any external synchroniser. Both wordclock and timing reference are taken from the designated master. The simplest remote option is to use an Alesis LRC which connects to the footswitch jack. For more comprehensive control Tascam offers the optional RC-2424 remote. Apart from these the machine responds to MMC commands. If you need Sony 9-pin control of one or many machines you'll have to wait for the forthcoming TL-Sync synchroniser.

The Scrub function works well, across all 24 tracks at once if you want. It sounds much like a tape machine with audio changing in pitch as the speed of scrub varies. Shuttle is similar, but the maximum speed is limited to 1½x play. I would have liked at least double and preferably 4x play speed. Rewind and fast forward do not give any audio output, but they both ramp up and down in speed like a reel-to-reel machine. I was pleased to find there is a reverse play function, less pleased to discover there is no dedicated key. You have to press PLAY and REWIND together. >

Front Panel

For a 24-track machine, the front panel is not particularly busy, although many keys have shifted functions. The big meter display is eye catching with 24 corresponding triangular track keys below it. Sixteen-segment bar graphs follow the usual Tascam pattern with only the 0dBFS segment coloured red. If three or more consecutive samples hit zero, the red latches until either PLAY or RECORD is pressed. Above this, a green segment indicates input monitoring. At the bottom another green segment shows if the track has been selected for editing and a red blinks to indicate record ready, going solid in record.

The triangular keys are not simply eye catching, they aid the ergonomics. With many keys in a small area the shape and spacing makes it easier to hit the ones you want.

All the keys have white legends printed on them indicating the primary function. Where there is also a Shifted function its white legend is beside the key. No keys are internally lit, but there are adjacent LED indicators where appropriate.

Left of the meters, below the power rocker switch, lie four Track Function keys. These determine which function will be applied to the track or tracks selected with the track keys. Thus, ALL SAFE toggles AUTO INPUT when shifted, RECORD SELECT, INPUT and EDIT switch all tracks when shifted.

Below the meters the MOUNT key attempts to mount all connected drives if none are currently mounted. With shift it unmounts.

Four blocks of five LEDs indicate various parameters:
Sample rate: 44100, 48000, 2X, pull down, NON STandarD.
Time code: 30, 29.97, drop, 25, 24
Record mode: 24-BIT, tape mode, DIGital INput, 2 CHannel I-O, auto input
Sync: Sample rate, Chase Lock, TL-Bus Master, TL-Bus Slave, TC Chase
A further block of four LEDs to the right of the meters indicates:
Status of: Error, Busy, MIDI and Disk
The round Edit keys are (Shifted functions in brackets):
CUT (LOCAL CUT), COPY (SPLIT), CLEAR (DISCARD), PASTE (SYNC PASTE), INsert (SYNC INSERT), OPEN (IN-NOW), UNDO and REDO.
Two rows of transport related keys sit right of the drive slot, top row:
ONLINE, LOOP (LAST), TO (PREVious), FROM (NEXT), IN (OFFSET) and OUT(TC-READER)
Larger, tape recorder style transport keys are below.
The concentric scrub-shuttle wheel has a semicircular halo of shaped keys.
Clockwise these switch:
JOG-SHUTTLE, NUDGE (CAPTURE EVENT), TRIM, INCREMENT, DECREMENT, SETUP (TEMPO), PROJect (NEW) and VIEW (UNLOAD)
Above scrub-shuttle the locator section has a numeric pad plus CLEAR-CANCEL, CAPTURE-SHIFT, and the all-important STORE-YES and ReCaLL-NO keys.
The 2-line, 40-character LCD is used for operator and system messages. When menus are not being accessed it shows the playhead position in the top line and entry register time values at the bottom.
The final item of interest is a 'Smart Media' slot. This is used to update the operating system.

< Apart from the normal tape recorder controls there are a number of autolocator functions. Capture places the current location of the play head in a register. From here it can be used to load the In, Out or Offset registers or be stored in one of the 100 locator memories. Auto Punch uses the In and Out points to set punch-in and out points for rehearsal and recording. When Auto Unload is set to on each Loop-Auto Record cycle creates a virtual track automatically.

Front panel edit functions are all based around the concept of setting In and Out points, selecting the track(s), and then performing edit functions. Clear removes the selection to the clipboard without slipping subsequent audio. Cut does the same, but slips all subsequent audio on the track or tracks up. Local Cut is similar, but only audio from the same event is slipped up. Copy and Paste work as expected, with Paste overwriting any existing audio after the In point for the duration of the pasted audio. Sync Paste uses the position of the play head in the source to define a sync point. In this way the peak of a jet by or similar can be hit without calculation. IN TO NOW moves the section between the In and Out points to the current play head position, overwriting any existing audio.

Insert slips all subsequent audio. Sync Insert works in the same way as Sync Paste, but slips subsequent audio. Open inserts silence, slipping subsequent audio. Reverse is an offline DSP function that reverses the selected audio. Render creates a single audio file from whatever multiple events are between the IN and OUT points. All edits, punch-in and outs use are real-time crossfades. The length can be set from a menu anywhere between 0–90ms. Using Viewnet more sophisticated editing is possible including level and fades in real time.

There is a great deal in this machine and its accessories to discover and digest. It is also continually necessary to remind oneself of the asking price. Next month I shall be looking at some of the possible applications for the MX-2424 and how it performs in practice. ■

Connectivity

THE PLAIN VANILLA MACHINE comes with a single stereo digital I-O in both AES-EBU and SPDIF flavours with input sample rate conversion. Most, if not all, purchasers will fit one or more of the optional multichannel interfaces. Two audio option slots allow simultaneous fitting of one analogue and one digital card. At present there Is only one analogue option. Twenty-four channels of 24-bit 48kHz (12 channels of 24-bit 96kHz) A–D and D–A convertors is on one board. In fact calling it a board is almost an insult. It is housed in a metal screening box with only the top open (this is covered by the top of the machine casing) For the digital slot, as you might expect, one option is 24 channels of TDIF. ADAT Lightpipe and AES-EBU are also available. The AES-EBU card features Input sample rate conversion on all 24 channels. For 96kHz operation both 'bit split' high resolution two wire and single wire 'wide' are supported. Digital audio sync possibilities comprise wordclock In, Out and Thru on BNCs with the welcome alternative of video sync IN and THRU BNCs. Audio syncs may also be derived from the digital audio I-O or time-code.

Transport sync connections are; SMPTE time-code In, Out and Thru plus MIDI In, Out and Thru and two TL-bus proprietary 9-pin D-sub connections.

Additional SCSI storage is taken care of by a half-height 5.25-inch drive slot on the front panel and a 68-pin High Density Fast-Wide connector for external storage devices. The slot is intended to be fitted with either a 'hot swappable' hard drive frame. (Kingston Rhino Junior and Data Express frames are recommended. The Data Express has the advantage of being quieter in environments where this is an issue.) or supported DVD-RAM and tape drives. A list is supplied with updates on the Tascam website.

A punch in footswitch connects to a ¼-inch jack. Polarity is automatically detected. It will also accept an Alesis LRC remote for basic transport remote control. More comprehensive control is available from the RC-2424 control surface which connects via a 9-pin sub-D, not to be confused with the TL-bus connectors. Apart from IEC mains input there is one further connector, an RJ-45 socket. This insignificant looking port is a 100 base T Ethernet network connector and is, arguably, the gateway to the machine's likely success.

Marantz PMD330, PMD331, PMD340

The one thing that goes hand-in-hand with the wide adoption of CD-R as a medium is the continued requirement for sophisticated CD players to accompany them. **Zenon Schoepe** reports

DESIGNED FOR POST, broadcast and DJ applications the Marantz 300 series of CD players comprises three units. The entry level PMD330 offers a good portfolio of standard professional features which are enhanced in the PMD331 and PMD340 with the inclusion of a 10s antishock buffer, instant start and advanced pitch control and remote features. The top-of-the-range PMD340 distinguishes itself additionally with the incorporation of an industrial grade transport mechanism and optical pickup unit with an integrated preamp.

A bullet point hitlist of the range's features includes CD-RW capability as well as CD Text—which for those that are unaware of its arrival employs album and track naming in much the same way as MD does. There's also frame search (75fps), index search, programmable cue-point memory, digital pitch control ±12%, end monitor, a 10-key pad, fader start, backlit LCD and adjustment free mechanisms with digital servo.

There can be no doubt whatsoever about the professional nature of these machines. They are built to a very high standard with a decidedly pro priority layout and dependable and programmable logic that can be customised by the user. Large transport controls are restricted only to PLAY-PAUSE, CUE and STOP all of which are lit with smaller buttons provided for the less immediate tasks.

Because the range adopts a derivative approach by adding features to a core unit as you go up the range I'll major on the PMD330 and the PMD340 as representatives of the extremes of the series and point out that functionally the PMD331 sports all the features of the PMD340, but runs a PMD330 transport rather than the heavier duty one on the PMD340. Differences between these transports were not evident in the course of my use of these boxes. However, I think the inclusion of a beefier transport in the flagship product is a significant one, not because it suggests that

the ordinary transport is in anyway lacking, but because Marantz clearly thinks it will appeal to anyone looking to invest in such a machine on a truly long-term and extremely high mileage usage basis.
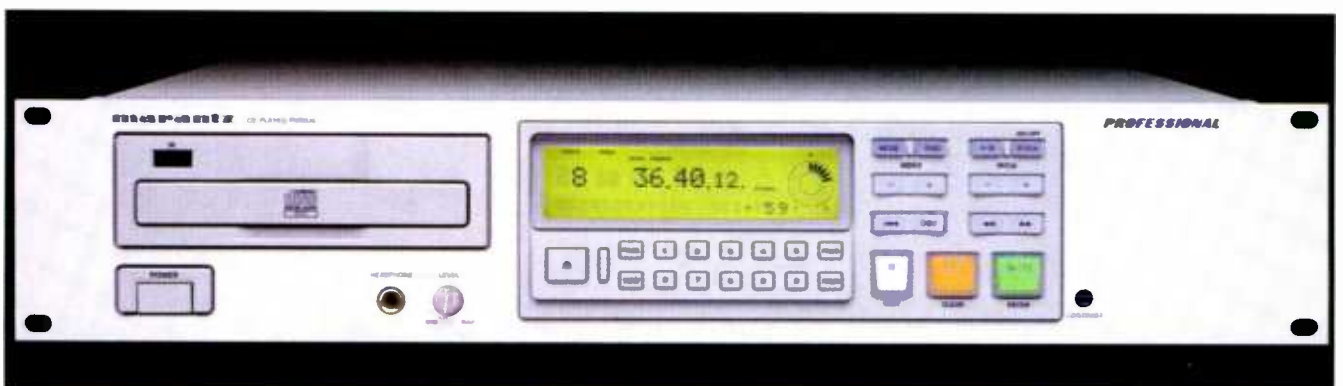
With the exception of the inclusion of a dial on the PMD331 and PMD340 (without which they would all look identical) all three machines look extremely similar with no outwardly visual clue to the existence of a different transport in the PMD340.

Rear panel arrangements also follow this theme with the entry-level machine being offered with phono analogue outputs and SPDIF on RCA plus a fader start jack and RC5 compatible remote socket and output with accompanying selector switch. An infrared remote was not supplied with my review models. The PMD331 and PMD340 add balanced XLR analogue outputs with individual grub-screw trim pots, a balanced XLR SPDIF output and an optical digital port. Remote control possibilities are

afforded by a 25-pin D-Sub that accommodates among most other functions the lack of dedicated fader start port as found on the PMD330.

As already alluded to, operation is identical across the range with the exception of the use of the dial. Time cycles between track and disc total, remaining and elapsed times while Text switches the display between time and text indication modes. Numeric buttons permit direct access to tracks and are used when programming playback which employ the PLAY-PAUSE, CUE and STOP transport keys which double as Enter, Clear and Exit buttons, respectively.

There are track and index increment buttons, and forward and reverse scan. The CUE button permits the location and creation of a cue point that can be entered pretty much on the fly or can be fine tuned by using the traditional looped audio segment approach that is nudged up or >
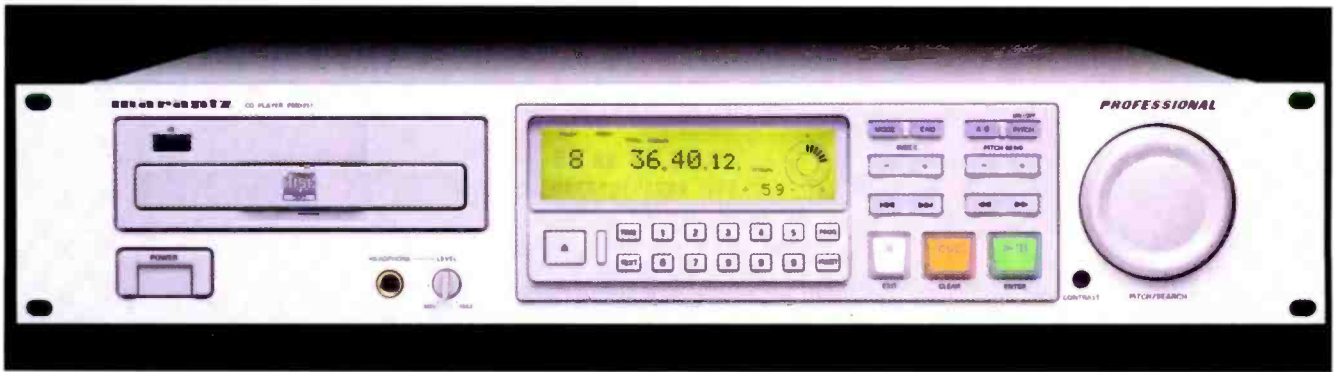
PMD330

PMD331

< down the programme by front panel buttons or the dial on those models equipped with it.

LCD contrast is adjusted on a front panel grub screw and the machine can be set to play single track, programmed, random and repeat modes as well as 'power-on auto repeat playback'.

Pressing END while paused auditions a preset selectable end chunk of the currently selected track while A-B predictably selects start and end points for repeated playback. Pitch adjustment employs a pitch switch with up-down keys on the PMD330 to do what the dial does on the PMD331 and PMD340 while the more expensive machines also get the ability to bend pitch up or down by 8% instantaneously at the press of a key and which drops back to the original speed just as quickly.

There is a surprising amount of detail and functionality hidden in the 300 series for what are after all just CD players. There are enough options and custom configurations available to make a machine completely baffling for an outside user and, consequently, enough to configure it in almost every aspect to the specific requirements of, say, a human driven radio station.

On the button

The PRESET button accesses the machine's customisable parameters while in Stop mode. You step through the parameters on repeated presses of the PRESET button or move forward and back through the menu on the index ± switches with parameters altered on the track increment keys. Parameters available for adjustment include auto cue activation, auto cue level, end monitor playing time, end warning flash time, and fade in and out times (up to 8s in 0.5s increments). Other functions cover pause mode to playback delay, default time display, key lock, and a useful fader start lock mode which locks out all controls aside from the Text and Time buttons when playback has been triggered by fader start with full control returned in Pause.

There are also settings for locking the tray from opening while in play, for setting the auto close time of the tray, the default mode entered after loading a disc (stop, pause or play) and time for the machine to go in to auto standby mode. Control connectors can also be disabled along with a CD sync function and display alert to signify that a 'skip' has occurred. The PMD331 and PMD340 additionally get the facility to defeat the digital outputs, if selected instant start, shockproofing and pitch bend are themselves defeated, and to fire an output on the remote I-O pin 11 when selecting index 2 or 3. Clever stuff.

ACTIVATE YO
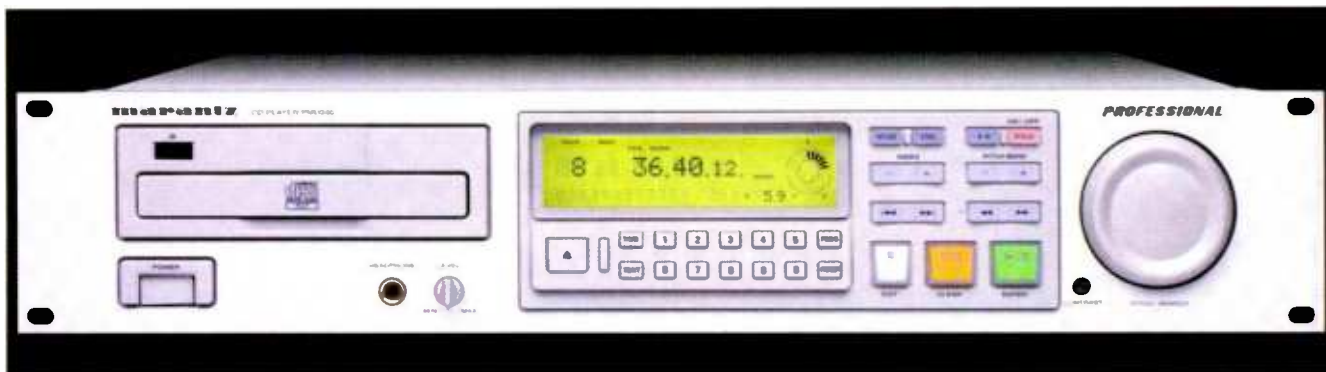
PMD340

Things like how long you want the tray door to stay open before auto closing itself is just one example of the thought gone in to these machines and how they have been made accessible. The remoting possibilities mean, in the case of the PMD331 and PMD340, that they can also be really well integrated into the work environment while the PMD331 does at least qualify with the essential entry requirement of fader start.

All models benefit from a rather good display that combines the usual sorts of things with a less usual rotary pitch display and numerical read-out plus an extremely useful 10-segment bar that moves from left to right to tell you where in the current track you are currently at —this is far more immediately meaningful than elapsed and remaining times, for example. The shockproofing available on the PMD331 and PMD340 seemed to do the job despite my best attempts to upset it, but when the sturdy cases are locked in to a rack you'd probably have to go off road while on-air to catch it out.

I'm particularly impressed by the way Marantz has split the 300 series into its distinct price points. Rather than kick it off as many do with a substandard bottom of the range machine that progresses to a happier medium product and is followed by an over-the-top flagship complete with cup-holder, cigar clipper and floss dispenser, it's weighed in with a base model that is extremely well speced bettered by models that add what are a few fairly lux features.

All in all these are great packages all with Marantz high audio quality outputs that extend to excellent headphones circuits. The issue then becomes which one to choose and the decision here will depend to a large extent on your circumstances. The PMD330 is probably adequate for most requirements for a dependable and trustworthy CD player with the PMD331 cutting in for those who need to remote a machine more comprehensively, up its connectivity possibilities and like the comfort of the shockproofing. Those who in addition to this work ordinary machines to death on too frequent a basis and-or are perhaps located in the more far flung and more isolated reaches of our readership will gravitate most naturally to the PMD340. It's a nice decision to be able to make and you won't be disappointed. ■

Contact

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Net: www.marantz.com

Drawmer 1969 Mercenary Edition

Adapting the successful 1960 dual vacuum tube compressor to suit all-American tastes has delivered the Mercenary Edition. **Dave Foister** signs up

THE NAME OF DRAWMER remains synonymous with dynamic processing. Fashions and techniques shift and develop, but Drawmer is always in there somewhere, adding digital control, digital outputs, valves, and anything else the industry wants and needs, and always with a little something to make it its own. Drawmer was in there with the retro movement, producing the 196X range of valve processors that included the big 1960 compressor, heavy on the British valve compression character. The 1960 was taken up enthusiastically around the world, yet Mercenary Audio's Fletcher still found himself reaching for the real vintage gear for particular sounds, and felt it would be useful to have a similar beast yet with a brighter, more American flavour. The result of his ideas and his input to Drawmer is the 1969 Mercenary Edition compressor.

At a casual glance the 1969 looks very similar to the 1960, but this is not a simple tweak of the original with a new badge: there are several differences in detail and operation. The basic package is essentially identical: two channels of valve-based compression, each with a fully specified microphone preamplifier as well as its line input, and a single instrument input with its own unusual features, that can be routed to either compressor channel or even both. The compressor controls are simple and direct, the two channels can be linked for stereo with Channel 1's knobs controlling the compression, and a pair of traditional vu meters shows either output level or gain reduction.

Like the 1960, the compressor has a soft knee characteristic that makes a ratio control superfluous. Thus the only knobs on the compressor section itself are THRESHOLD, ATTACK and RELEASE, and two of these are rotary switches rather than pots. Here we see the first clear difference from the other unit: the 1969 has six Attack settings where the 1960 only had three, for slow, medium and fast. These in turn do not have actual

time values specified for them, partly perhaps to avoid mixing by numbers, but also because all the times are varied according to what release time is chosen. Here too there is a change, in that of the six switched settings three are programme-dependent, operating over different ranges of times: position 4 gives 200ms to 2s, while 6 gives 1s to 10s.

Obviously with this many settings, this much programme dependence, and an interaction between the two adjustments, there is far more flexibility available than it might initially seem, and in the best traditions a bit of experimenting is usually rewarded. There is no automatic gain make-up, but a pair of output level controls allow it to be adjusted manually; unfortunately these are not ganged for stereo operation. Stereo linking has a new feature—a third position on the toggle switch marked BIG. This contours the compressor side chain to make it less sensitive to low frequencies, avoiding the risk of heavy bass content pumping the rest of the signal up and down. I found this a big benefit, saving the need for patching an EQ into the side-chain inserts on the back.

The microphone preamps are valve-based, and now have switched gain controls and the addition of a phase reverse switch on each channel, which also operates on the line input along with the 2-position high-pass filter. The auxiliary instrument input takes the increasingly familiar idea of the front-panel DI jack to extremes, making it almost a complete guitar signal path needing only a speaker simulator on the end. It's got bass and treble boost EQ, a BRIGHT switch to emulate the typical presence peak of a guitar amp, and its own gain control that allows its dedicated valve preamp section to be overdriven. This makes it a much more likely candidate than most to be the front end in a guitarist's rack, and adds useful facilities for bass and keyboards too.
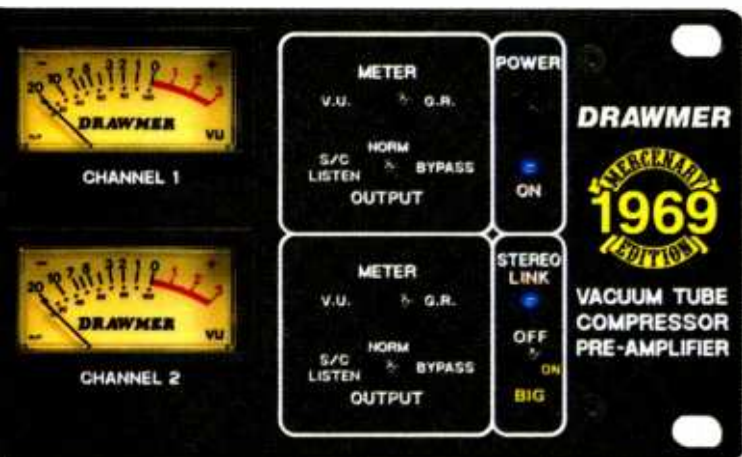
There's something confidence-inspiring about a front panel like this, and it's very

> There's something confidence-inspiring about a front panel like this, and it's very easy to get drawn in to what the 1969 can do and mess with the settings. There is a distinct character to its sound, which is wide open and clean with a hint of brightness that counteracts the tendency of some compressors to sound slightly dull

effects are, of course, often brought about by injudicious adjustment, and the deceptive simplicity of the 1969's controls makes this easy to avoid. It has all the control you need to handle all the expected compression tasks, and I found it particularly effective on an overall stereo mix, where it could be needle-bendingly extreme or unobtrusive as required, and always with its pristine clarity intact.

If there are some types of devices that should have no character, a compressor is definitely not one of them. The background to the 1969 shows this well —there's nothing wrong with the 1960, it just has a certain character that suits some applications better than others. The 1969 fills that gap, and could reduce still further the need to put up with the crackles and hums of forty-year-old equipment just to get that certain special something. ■

Manley 120W Monoblock

Studio Sound's bench test amplifier reviews continue with the Monoblock. **Paul Miller** reports

HAVING TESTED a variety of fairly conventional, solid-state amplifiers over the last few months, we thought it timely, as the saying goes, to look at something completely different. Different, in this instance, means valve-based amplification courtesy of Manley Labs. Based near LA in the States, Manley has footholds in the professional and domestic hi-fi markets with no less than 50 valve-orientated products. These range from microphone preamps, equalisers and limiters to standalone D–A convertors and preamplifiers accommodating both line-level and phono sources.

It is Manley's wide range of valve power amplifiers, however, that really straddle the divide between pro and domestic scenes, and here it is possible to choose anywhere between the 18W single-ended triode 300B 'Retro' to models as powerful as the 500W Monoblock. This test revolves around Manley's 120W Monoblock which weighs in at a not insubstantial £2,895 per pair but that employs the 'technology', including user-selectable feedback, associated with its bigger pre and power amps.

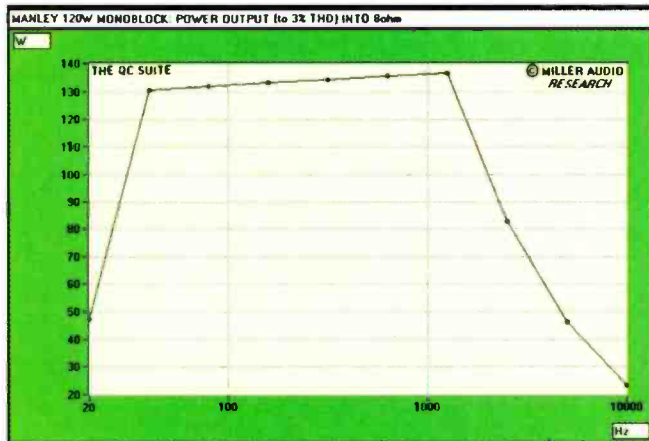This sort of amplifier is not unknown in the control room, but is probably better suited to less
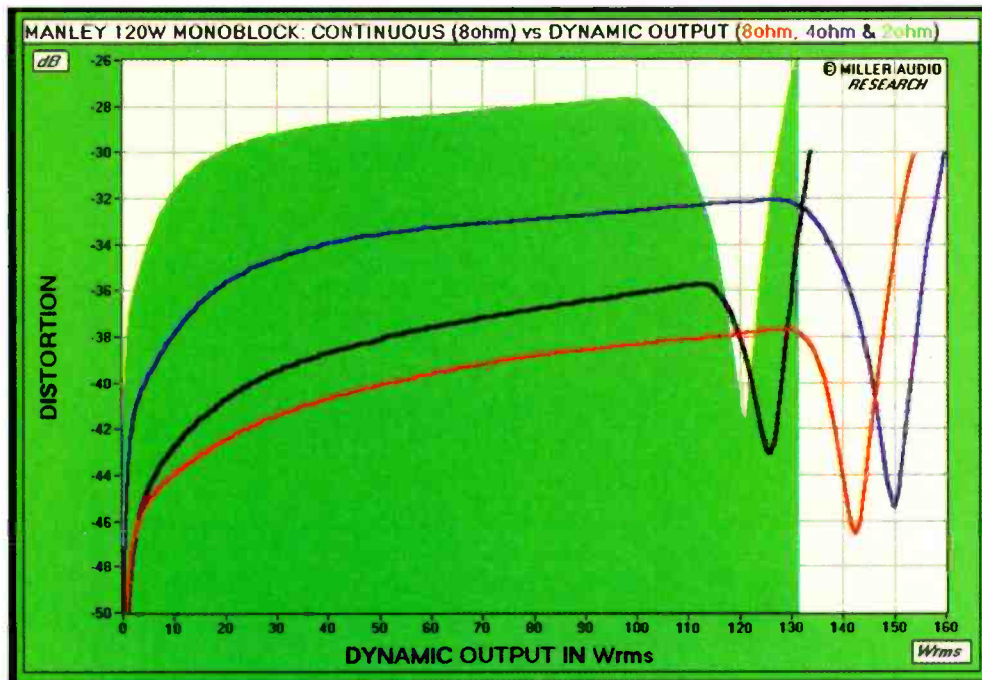

Fig.1

Fig.2

Power Amplifier: Manley 120W Monoblock (Rated Spec. in brackets where given):

	20Hz	1kHz	20kHz
Max Continuous Power Output.			
3% THD into 8Ω (one channel)	47W	135W (120W)	15W
5% THD into 4Ω (two channels)	-	135W	-
Frequency Response @ 0dBW	-0.06dB	0.0dB	-0.8dB
Dynamic Headroom (IHF)		+0.6dB (154W)	
Maximum Current (10msec. 1% THD)		8.1A	
Output Impedance (9dB NF)	1.442Ω (0.5-1Ω)		
Damping Factor	5.547 (~10)		

RCA Input

Total Harmonic Distortion (9dB NF)	
(0dBW, 1kHz)	-49dB
(2/3 power, 1kHz)	-37dB
Noise (A wtd, re. 0dBW)	-79.2dB
(re. 2/3 power)	-97.0dB (-91dB)
Residual noise (unwtd)	-58.4dBV
Input Sensitivity (for 0dBW)	188mV
(for full output)	2186mV (1000mV)
Input loading	100kΩ
DC offset, left/right	0mV
Serial Number	M12021
Retail Price	£2895

intensive work in the editing-and-or mastering suite where bombproof, 24-hour operation is rarely a pre-requisite. The engine-room is provided by a pair of KT90 pentodes, though 6550's, KT88's and even EL34's (with some small modification) may be loaded as alternatives. The single-ended input is serviced by a 5751 double-triode that, in turn, feeds a 7044 double-triode in a phase-inverted configuration.

The amplifier may be operated in Ultralinear mode, where the feedback signal is derived from one of the transformer taps, or triode mode, where the feedback is taken from the point at which the plates are connected to transformer primary. The lower-output triode mode ensures the tube is ostensibly more richly biased and therefore more linear, but maximum sound levels and, potentially, valve life will be forfeit. The output transformers themselves are wound in-house, promising a more consistent part that's also 'tweaked' very specifically for the product at hand. An output tap rating of some 5Ω provides a 'halfway house' between separate 8Ω and 4Ω taps commonly encountered on other valve power amplifiers.

Tested in the preferred Ultralinear mode, this '120W Monoblock' succeeds in delivering some 135W into 8Ω at 1kHz with a THD limit of 3%. This represents a gain of +23.5dB. Unlike most solid-state designs, their valve cousins rarely clip so abruptly, but overload in a progressive and arguably more graceful fashion. Nevertheless, with transformer core saturation to cope with at very low frequencies and slew-limiting at high

Fig.3:

Fig.4:

Fig.5:

frequencies, this same 3% limit is quickly reached at just 47W (20Hz) and 15W (20kHz), respectively (Fig.1).

It is possible to squeeze slightly more juice from the Manley's under dynamic conditions but, as depicted by Fig.2, the increase in THD with output follows an almost logarithmic path. Plotted here on a *linear* power scale, it's possible to see the anomalous *decrease* in distortion just before the notional clip point at 154W (red), 160W (blue trace) and 131W (green trace) into 8Ω, 4Ω and 2Ω loads, respectively. The amplifier's continuous output profile is traced in black. Clearly, the Manley lacks both the load tolerance (dynamic current = 8.1A) and headroom (just +0.6dB) of its solid-state competitors, so moderate-to-high sensitivity speakers with a minimum 8Ω impedance loads are recommended.

The nature of the speaker load takes on even greater significance in the light of this amplifier's high output impedance trend. Three levels of negative feedback (low 6dB, medium 9dB and high 12dB) are available and which inevitably effect this trend (see Fig. 3), but none can quell the inductive rise at HF which reaches nearly 5Ω at 20kHz (minimum feedback). Even with maximum feedback (red trace), the overall amp-speaker response will depend heavily on the swings in impedance of the speaker load, especially at HF.

The linear losses are reflected on Fig.4 which shows the response into nonreactive 8Ω (green trace) and 4Ω (red trace) loads with minimum feedback. The responses may be compared with those obtained into no load (blue trace) and 8Ω with medium feedback (black trace). Note that all responses are normalised at 1kHz, concealing

the uniform loss between 8 and 4Ω loading, for example. Either way, with the HF impedance trend of many speakers dipping below 4Ω, the impact on the overall system response will certainly colour the final sound of the amp-speaker combination.

Tonal colour, of course, is also modified by the high and extended harmonic complement of the amplifier's distortion spectrum, traced out on Fig.5 with minimum (blue), medium (black) and maximum (red) levels of applied feedback. The lower spectrum clearly highlights the high 2nd, 3rd and 4th harmonics which typically persist at around 0.7%–1.0% through the midrange at modest power levels and medium (9dB) feedback. Higher-order harmonics are also clearly visible which do not enjoy the same degree of signal masking and, therefore, contribute to a greater subjective impact. It's also possible to see some supply modulation, visible as the cluster around the base of the ~1kHz tone, which, typically, influences bass resolution and the crispness of stereo imagery.

All of which is reflected in the very rich, warm and undeniably 'colourful' sound produced by the 120W Monoblocks. A sound that audiophiles would, and with some justification, describe as 'musical' and enthralling if not especially neutral. So, as a transparent window on the musical event, this amp's slightly rose-tinted perspective is, perhaps, not ideal as a tool for mixing and production purposes. And yet, when the desks are powered down and the studio quiet of an evening, I can imagine the Manley's being brought out if only to enhance the simple pleasure of listening to the fruits of a day's labours. ■
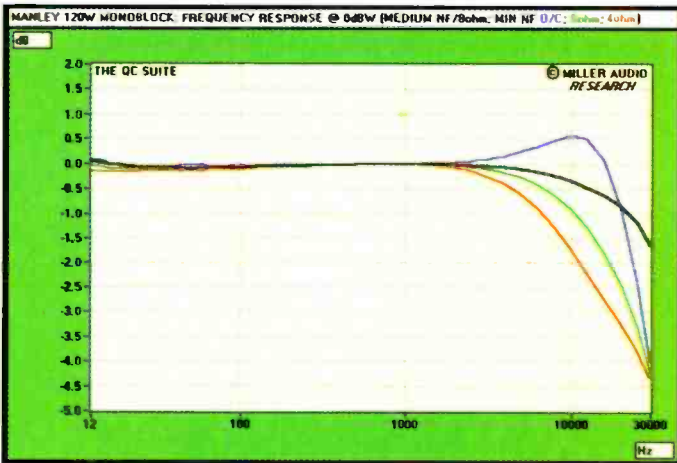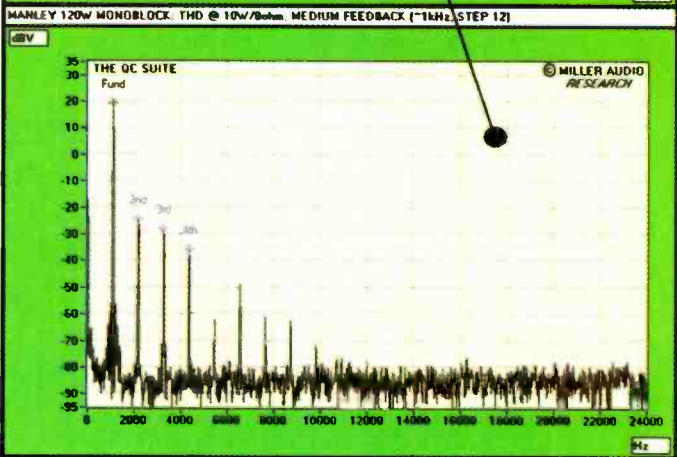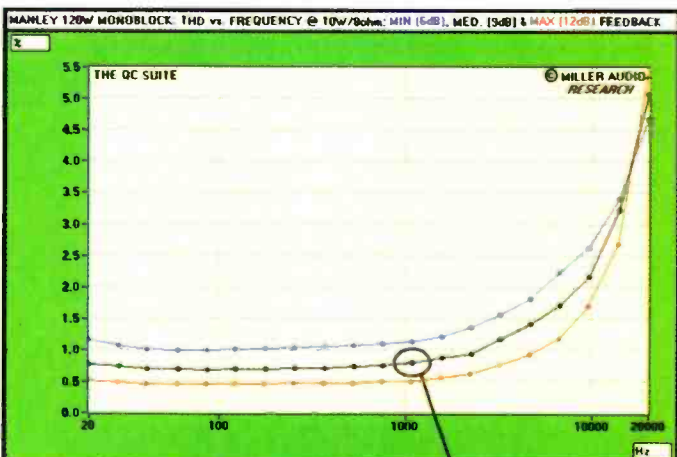
DAS Monitor 6

Studio Sound's 'bench test' loudspeaker reviews continue with the Monitor 6. **Keith Holland** reports

THE DAS MONITOR 6 is a 2-way, passive loudspeaker comprising a 165mm woofer with a polypropylene cone and a 19mm aluminium-domed tweeter that radiates through a shallow axisymmetric horn. The cabinet has external dimensions of 335mm high, by 226mm wide, by 265mm deep, weighs 6.9kg and has a rear-facing port. The loudspeaker is supplied with a cloth grille that was removed for the tests. DAS recommend the use of amplifiers with a maximum power output of between 100W and 200W into 8Ω, and specify a maximum peak SPL at full power of 111dB.

The author regrets that, due to an equipment fault at the time of the measurements, the usual harmonic distortion plots are not presented in this review. Fig.1 shows the on-axis frequency response for the Monitor 6. The response is a bit uneven at high frequencies with a 5dB dip at 10kHz, but it lies within ±3dB from 70Hz to 8kHz and ±4dB from 65Hz to 20kHz. The low-frequency roll-off is approximately 4th-order with -10dB at about 40Hz. Figs 5 and 6 are the horizontal and vertical off-axis responses respectively. The horizontal directivity is well controlled with no evidence of side-lobes or mid-

frequency narrowing, but the vertical directivity shows the interference notch due to the spacing of the drivers appearing in the upward and downward directions. The 10kHz dip, evident in the on-axis response, is not present at 15 off-axis in any direction. The time-domain performance of the Monitor 6 is particularly good, with the step response (Fig.3) demonstrating excellent time-alignment of the drivers, and the acoustic source position (Fig.2) showing a shift of less than 2m at low frequencies—an unusually low figure for a ported design. The waterfall plot (Fig.7) further demonstrates the rapid decay at low frequencies, although there is some evidence of ringing in the mid-band between 400Hz and 900Hz. The cause of the dip in response at 10kHz may be apparent from the power cepstrum which is shown in Fig.4. A strong echo can be seen after about 100μs, which corresponds to a path length difference of 34mm—one wave length at 10kHz. This suggests that the 10kHz dip is due to diffraction from an edge, such as the mouth of the tweeter horn, as diffracted waves are phase-inverted relative to the direct wave. As noted above, the 10kHz dip is not present at angles away from the axis, which also points towards diffraction. Overall, the DAS Monitor 6 is a good performer. Its has a particularly good time-domain response, with accurate time align-

ment and minimal low-frequency acoustic source position shift, both of which suggest an accurate response to transients. Horizontal directivity is well controlled, but the crossover interference notch is evident both above and below axis. The uneven high frequency response is not a problem if listening is carried out slightly off-axis in the hori-

zontal plane (this applies to most loudspeakers which use axisymmetric horns). DAS specify harmonic distortion levels of around -35dB (1.8%) at low frequencies for an input of 10% of the nominal power handling rating. This figure reduces to -50dB (0.3%) at frequencies above 200Hz. These figures are typical for a loudspeaker of this type. ∎
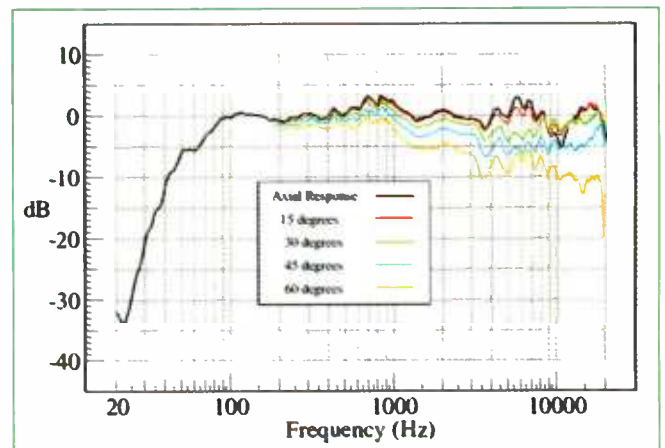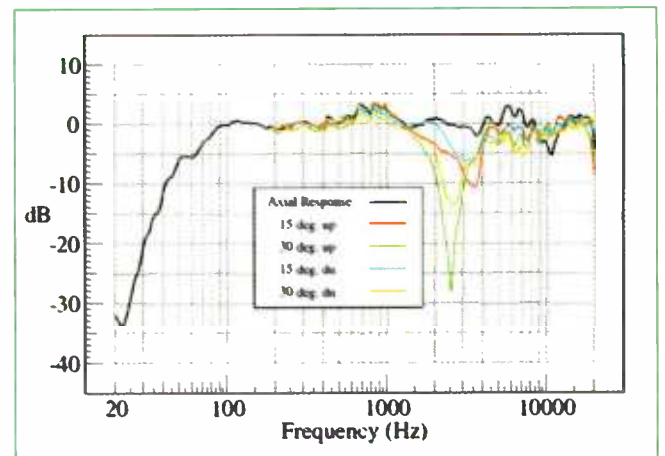
Fig.5: Horizontal Directivity
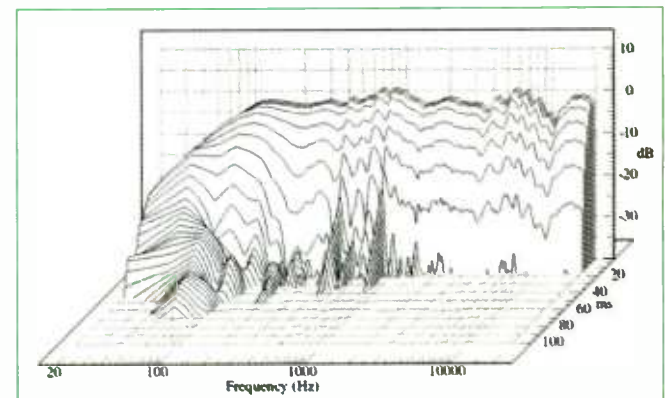

Fig.6: Vertical Directivity
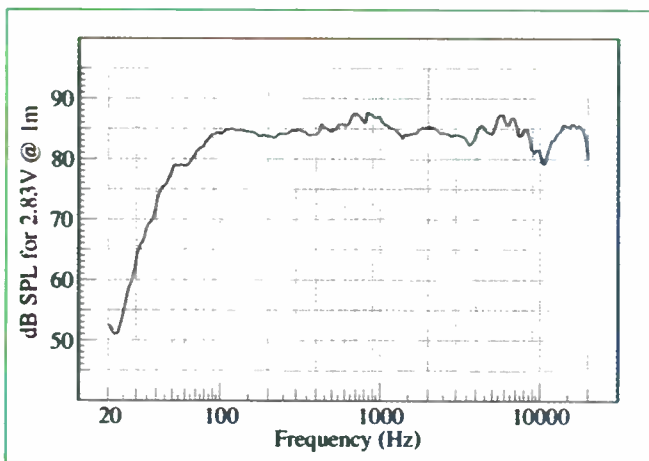

Fig.7: Waterfall

Fig 1: On-axis Frequency Response and Distortion
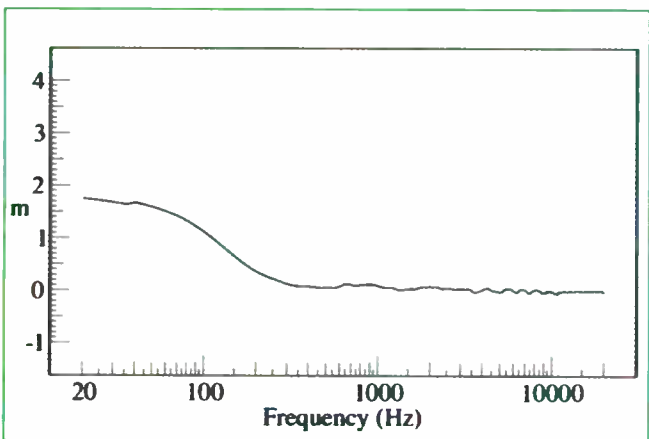
Fig.2: Acoustic Source
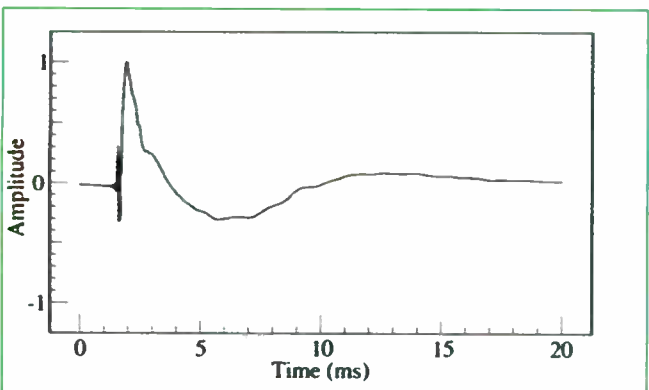
Fig.3: Step Response

Fig.4: Power Cepstrum

STUDIO SOUND

evolution

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NEW TECHNOLOGIES

Soundtracs nets

Soundtracs has released net.tracs as a networking environment for users of its products that permits the interchange of project information between consoles in a multiroom facility and to provide a convenient facility-wide archival and restoring system. A 19-inch rackmount server controller connects via CAT5 ethernet and deals in Soundtracs session files for relocation and restoration.

Soundtracs, UK. Tel: +44 1372 845600.

SIENA PCI

SEK'D has introduced the SIENA PCI audio card which has eight analogue inputs and outputs and 24/96 resolution. Two separate

MIDI I-O ports are included and multiple SIENA cards can be connected via a Sync-Bus. The card is Windows 95/98 compliant and comes with a complete driver set.

SEK'D, Germany. Tel: +49 7946 776 66.

Sea Sound Solo

The Solo EX recording system enables interactive monitoring of record and playback of signals with separate level controls. Essentially a PCI card supporting Windows and Mac, it has a separate break-out box that contains the A–D/D–A convertors and all interconnection facilities, including inputs for instruments and microphones with 24/96 conversion, channel inserts headphone amps, built-in mixing, SPDIF and MIDI in, out and through with zero latency input monitoring.

Seasound, US. Tel: +1 415 485 3900.

Compact flexibility

The Altair DA410 is a 1U-high distribution amp, zone mixer and mic splitter able to process two stereo input signals to obtain five stereo outputs with independent level control. As a zone mixer it can control five zones in stereo mode and ten in mono mode. Individually selected mic-line input preamps allow mic or line signal splitting to additional destinations in 4 x 8 or 1 x 10. The unit has a monitoring system with built-in microspeakers, headphone output and selectable vu meter. I-Os are electronically balanced with transformers an option as are two parametric EQ cards.

Altair, Spain. Tel: +34 918 043 265.

Domestic Interface

The Canford Pro-Interface Mk II is a bi-directional stereo interface for matching semi-pro or domestic equipment ▷

AEA R44C

Offering a real alternative to modern condenser microphones, this reissued ribbon brings the past up to date **Dave Foister** listens in

IT'S COMPETITION TIME. Of all the replicas of classic equipment, which dates back the furthest? How long ago could you have found equipment in use that still has a place today, not just as a means of getting a 'period' sound but as a quality tool alongside modern equipment? Valve microphones, valve equalisers, valve compressors are all very well, but in terms of the age of the originals they don't come close to the ribbon microphone. Now a Californian company that for 20 years has been servicing old microphones has come up with a replica of one of the classics—the RCA 44. Although the original was manufactured over many years, Audio Engineering Associates has gone all the way back to the very first models produced in 1936 to produce the AEA R44C.

Even if you didn't know what it was—and on the Eastern side of the Atlantic this could be forgiven as its use was never so widespread over here—this microphone would be instantly recognisable. It's as much of an icon as the BBC AXBT or the PGS/4038, both of which share the basic technology of the American model. In those days the ribbon was king, far and away the most sophisticated microphone technology available and capable of an audio fidelity that can stand up to close inspection today. Not long ago we looked at the currently available ribbons, many with an unbroken heritage dating back to the era of the 44, and it was a salutary reminder that the condenser was certainly not the first decent microphone design the industry had. In terms of noise, flatness of frequency response, and uniformity of polar pattern, to name but three, a good ribbon can rival almost any condenser.

But like most classics, the success of the RCA 44 was not predicated on clinical accuracy, but on a characteristic sound. This is why AEA has gone back to the earliest design, because, although its bass response is demonstrably less even than later refinements, its sound is regarded by adherents as superior. Because of its servicing facility AEA is in constant touch with many users of the originals, and could therefore easily decide which version to reproduce, much as AKG did with the C12VR.

The AEA R44C is, of course, enormous. It comes packed in a fabric pouch that is in turn inserted into a very large foam padded canvas carrying case, the whole thing being so big that when it arrived I had no idea that it might be a microphone. Its weight is as substantial as its size, thanks to the magnet essential for its operation. Its distinctive grille, with its angled faces allowing a glimpse of the working bits inside, sits atop a body identical with the original apart from the logo. The whole thing is suspended in a swivelling stirrup arrangement, whose big knurled nuts can only just be finger tightened enough to stop it moving unintentionally. The integral flexible cable emerging from the main body is clamped in the stirrup's base, and this base is the only means of mounting the microphone on a stand. Don't even think of trying to extend the thing on a boom—it'll only end in tears as the 44's weight brings the whole lot crashing to the ground.

The cable is reasonably long and terminated in an XLR: in the absence of instructions I didn't dare connect it to a phantom powered input, although, of course, it ought to be perfectly happy. The output level, like all ribbons, is relatively low, but since its noise is almost non-existent this is only a problem if your mic preamps aren't up to it. There is now an X variant higher output version of this mic available. The polar pattern appears to be a very smooth and uniform figure of eight, allowing me to place pairs of musicians across it and adjust their balance by getting them to move backwards and forwards.

This is the kind of approach that a microphone like this seems to positively encourage. I used it on a special project where we were trying to recreate the sound of an early jazz big band, and there was a sense in which this microphone's character, along with the avoidance of very close techniques, made it easier to find the sound we wanted. On the other hand, this should not be taken to imply that the 44's sound is in any way lacking the extremes: I used it on trumpets, saxes, trombones and percussion where I wanted a modern sound, and it was all there. Valves have warmth, and ribbons have it too, but in a different way. This microphone has a velvety smoothness, uncompromised by HF deficiencies, that would be hard to find anywhere else.

AEA's reproduction has won plaudits from engineers like Bruce Swedien, Allen Sides and Shawn Murphy, all with plentiful experience of the original and all saying the AEA version is as good or better. It may be tempting to think of it as a useful movie prop that happens to work, but this is a serious microphone that delivers something special. And it doesn't half impress the artists. ■

Contact

Audio Engineering Associates, US.
Tel: +1 626 798 9128.
Net: www.aera.com

Sennheiser MKH800

With mics capable of feeding fast and wide digital records thin on the ground, the MKH800 is a welcome innovation writes **Dave Foister**

IT IS ONE OF LIFE'S LITTLE MYSTERIES that the Sennheiser MKH80 has never truly become an industry standard, an icon like the U87, the 414 or the 4006. It is comfortably one of the world's half-dozen or so most expensive microphones, and with justification, as it is also one of the most flexible, high quality tools on the market. Yet outside a band of devotees it has not passed into the language. So confident is Sennheiser of its innate qualities, however, that the microphone has been upgraded still further, as Sennheiser's first bid to offer a microphone that makes sense of 24-96 recording.

It does not take much thought to realise that one illogical aspect of the quest for ever higher sampling rates, and the striving to deliver frequencies outside the conventional range of hearing, is the lack of those frequencies in much of the source material. There are some microphones—notably the small-diaphragm omnis from the likes of DPA and Earthworks —that have uniform frequency responses extending to 40kHz and beyond, but most of the other models we use are starting to roll off before they even reach the brickwall filter frequency of traditional CD sampling rates, and by the time we reach the area of the spectrum where the benefits of 96kHz sampling rates lie, there's really nothing much coming out of them to record. This is not the only point of higher sample rates, but if you accept the necessity to reproduce frequencies in that octave above 22kHz, the sad truth is that there often is not much there in the microphone output to begin with.

Hence the MKH800, which takes the design of the 80 and adds another octave. Since the original is not as familiar as perhaps it ought to be, a reminder of what it offers is in order. This is a side-fire condenser with an unusually small (half-inch) diaphragm, and a trademark set of four small rotary switches down the front of its body. It is usefully compact, with a body little wider than some simple end-fire models, and attaches happily to a stand with a conventional Sennheiser sprung clip. Its construction is certainly not lightweight, but its size means that it does not have the mass or awkward bulk to balance that bigger models do, and is very easy to position either singly or in pairs. There is an optional shock mount

available, but this too is conveniently compact.

The switches are what lends the microphone its unusual versatility. To begin with, it has no less than five polar pattern settings, inserting supercardioid and wide cardioid between the usual set of cardioid, omni and figure-of-eight. This facility makes it a particularly flexible all-rounder for various stereo configurations, from spaced to crossed to MS; in fact a pair of these makes an ideal MS coupling, partly because they are easy to set up physically, with close proximity between the capsules, and partly because the front-facing M microphone has such a good selection of patterns available to control the ambient pickup of the array.

Attenuation and bass roll-off are dealt with on two of the switches, again with extra flexibility as there are two levels of LF cut and two pad positions. The final switch adds the unusual facility of slight HF lift, again with two positions offering +3dB and +6dB at 8kHz. This degree of frequency contouring on the microphone itself is highly unusual, offering a useful alternative to EQ in tailoring the microphone to the acoustic while actually rigging it.

This much is all there on the original version. The MKH800 adds only the extended top end, which is now claimed to be flat all the way to 50kHz while retaining a low end down to 30Hz. This is achieved by means of a redesigned capsule, revised electronics, and a fresh look at the enclosure as its acoustic effects in the upper reaches become more significant.

Using the 800s I quickly came to understand why the microphone's adherents are so enthusiastic about it. It delivers a top-flight sound in a package that makes it adaptable for any situation. The term all-rounder is often taken to mean jack of all trades, with the implication that it is master of none, but the MKH800 can be master of almost any of them. Superbly quiet yet with a maximum SPL handling ability of 142dB, it is equally at home as a single microphone on an acoustic guitar, recording a soaring classical voice, or as a stereo array over a drum kit. For this I set up an MS pair and the benefits were immediately apparent, with an ideal combination of consistency, control and quality. It also showed well the HF capabilities of the microphone, which appear to have no audible limits and next to no colouration.

Every development in audio requires parallel developments in associated equipment. Just as surround requires surround processing, high-sampling recording requires extended frequency response in its sources. The MKH800 is one of few mics that can meet that requirement, and perhaps the only all-purpose microphones on the market that can deliver the spectrum modern recorders are designed to capture. ■

NEW TECHNOLOGIES

professional audio line levels. The Mk II is a portable version of the original 1U-high rackmount Pro-Interface. The rugged design has four XLR connectors on the front panel. Two are female, connecting to high-impedance, electronically balanced, line-level inputs, while the remaining two are male, fed from electronically balanced low-impedance, line level outputs. The rear panel has four phonos. All have associated multi-turn gain controls
Canford, UK. Tel: +44 191 418 1000.

Neutron upgrade
AVI has upgraded its NuNeutron monitors to Neutron III status with a number of improvements that are available as a pay for retrofit to existing models or for the same

£499 as the original. The crossover on the 2-way box has been moved down to 2.8kHz together with improvements to the circuitry.
AVI, UK. Tel: +44 1453 765682.

ESI2000 sampler
Successor to the popular ESI4000, E-mu's ESI2000 sampler is said to be the most affordable sampler from the company ever and features 64-voice polyphony, up to 128Mb of RAM, 64 digital 6-pole filters (19 different types) and DSP capabilities that include time compression and expansion, parametric EQ and digital tuning. The user-interface has ten programmable buttons which can be used to trigger samples while a Turbo board adds two more stereo sub outs, an effects stereo submix, two 24-bit effects processors and SPDIF I-O. The machine comes bundled with ten CD-ROMs of the official Emu sound library.
E-mu-Ensoniq, US. Tel: +1 831 438 1921.

New drivers
Digigram has released VX drivers for PC and Mac. On the former this includes an ASIO driver and current Wave for VX driver while for the latter it includes the ASIO driver and an update on SoundManager for VX driver.
Digigram, France. Tel: +33 4 76 52 4747.

Logic adds surround
Version 4.5 of Logic Audio adds surround functionality and a new class of audio objects called audio inputs that is said to further the value of the integrated real-time effects system. Studio Description Language aids the description of MIDI devices for more control over complex setups. The surround features permit freedom in the placement and movement of sounds from any channel strip with full automation. A sep-

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DIGITAL AUDIO SOLUTIONS ™

Lindos Electronics LSC24

Crossing the line between digital and analogue domains inevitably raises questions of quality. **Rob James** puts the LSC24 to the test

REGARDLESS OF THE INEXORABLE onslaught of digital technology, audio still demands precision analogue electronics. Successful convertor design requires meticulous attention to engineering detail and some black art. Even in digital-to-digital conversion there is no single 'correct' method, guaranteed to produce good results.

The name Lindos will be familiar to anyone who takes an interest in audio test equipment. The LSC24 Studio Convertor is Lindos' first foray into the fiercely competitive world of outboard signal processing. And Lindos is not the first manufacturer to crossover from gamekeeping to poaching.

LSC24 is a stereo convertor with 96kHz, 24-bit capabilities, offering a number of modes. Convert mode gives digital-to-digital sample rate, bit depth and format conversion. Record mode is intended for analogue-

to-digital conversion with an analogue output for monitoring purposes. ADDA gives independent analogue-to-digital and digital-to-analogue conversion while HQDA mode offers high-quality D–A conversion.

Housed in the ubiquitous 1U-high metal case, the unit has a smooth membrane across the front panel. The rear panel has XLRs for analogue I-O, AES A and B I-O and AES sync with BNCs for X256 wordclock and AES 3-id I-O. The last two double as SPDIF I-O. There is also a 9-pin sub-D 'data port' and IEC mains input. There is no mains switch; when did you last turn off the power on a rack unit?

Nor are there any knobs or buttons as such as all switches are of the membrane type, clearly delineated by grey keys—selection is indicated by LEDs above. The panel is logically laid out. Panel lockout switch and indicator on the left followed by mode select, analogue input gain block, the other parameter blocks and finally output gain. Input and output meters are 11-segment LED ladders. The input pair has peak hold and RESET key.

Analogue level adjustments are made using the increment-decrement keys below the large bright numeric LED displays. Levels are set in discrete, 1dB steps. Input ranges from 8dBu to 26dBu for 0dBFS. Output from -9dBu to 26dBu at 0dBFS. D–A performance is optimised for the '+18' and '+26' settings. Lindos has taken the unusual course of using resistor arrays, switched by relays, to set gain, achieving accuracy and solid tracking between channels. Lindos elected to use relays rather than solid-state switches to ensure minimum degradation. I think this is a better option than either knobs which get accidentally knocked or fiddly multi-turn pots which are really only suitable

for 'set and forget' purposes, although small clicks are sometimes audible between steps due to sudden gain changes and there is invariably a big one when you switch between 18dB and 19dB on the D–As. Since gain will normally only be adjusted when setting up this should not be problematic, but should be borne in mind for live work or high monitoring levels.

In A–D modes a soft limiter is available that begins to operate at -3dBFS. A red LED indicates limiting. Output word length is selectable between 16-bit, 18-bit, 20-bit or 24-bit. Noise shaping uses Lindos' own 'Minimally Audible' filter algorithm with an ultrasonic noise-shaping filter option for double sampling rates. Analogue monitoring is switchable before or after noise shaping for comparison. Sync source (referred to as DARS, or digital audio reference signal) is selectable between the AES11 (XLR) input, external times 256 (super) clock or incoming audio. There is no indication of valid signal on the selected sync source which caught me out once or twice.

The same selection of chip-sets is available to many manufacturers so the desirability of most designs is all down to implementation and factors such as user interface and design of noise shaping curves.

It is difficult, if not impossible, to achieve the full performance theoretically possible from 24 bits. The Lindos acquits itself honourably with quoted noise and THD figures well up to average for the current generation.

The LSC24 combines three functions in one workmanlike unit. Setup is simple and logical for the operator and with 'stepped attenuator' gain adjustments there is little excuse for getting encoding levels wrong. Interfacing is flexible with the inclusion of both single and dual 96kHz most welcome. The only strange omission is x1 wordclock.

At this level anything less than excellent performance would be surprising. The LSC24 does not disappoint. I liked the simple setup and subjective quality although, without similarly high -quality units to hand for comparison I cannot honestly say I noticed anything extraordinary about the sound. Performance is certainly superior to the convertors built into many consoles and recorders. Dither appears effective and smooth and I could not detect any obvious artefacts from the sample-rate conversion.

Analogue monitoring output in sample rate conversion mode adds to the flexibility. This combination of functions in one slim box

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arate fader controls LFE contribution per strip. Meanwhile Waveburner Pro for Mac builds on the original CD mastering software. The system integrates mastering plug ins and support for VST format plug ins. Plug ins include Fat EQ, compressor-limiter, multiband compressor and audio restoration tools. The system is 24-bit with dithering and noise shaping. AIFF and SDII files are supported in addition to WAV and MP3. Emagic has also introduced the EXS2424-bit sampler for Mac and PC which runs under Logic Audio and MicroLogic AV from V4.0. Sample assignments are performed in a editor in WAV, AIFF, SDII, and Akai with other standards underdevelopment. Sounds can be subject to subtractive synthesis processing, output selection and keyboard settings and velocity zoning.
Emagic, Germany. Tel: +49 4101 4765 44.

Low cost studio condenser
Canford have launched a large diaphragm condenser microphone. Despite its budget price the CSM41 is said to be manufactured to a high standard and features a 1-inch gold-sputtered diaphragm plus a cardioid pickup pattern, balanced transformer output and a machined brass enclosure. The mic also has an internally mounted bass cut switch. Frequency response is claimed

as 30Hz–20kHz with a sensitivity of 15mV/pa and a maximum SPL of 130dB. Canford supplies the CSM41 complete with elastic suspension and flight case.
Canford, UK. Tel: +44 191 418 1000.

Smallest mic
Countryman has introduced what it claims is the world's smallest mic with the B6 lavalier. Measuring one tenth of an inch in diameter the mic is targeted at broadcast and live applications. Impervious to moisture damage it can be worn on a collar or taped to a performer's face. Replaceable caps prevent perspiration and make-up from entering the mic but can also alter its colour and frequency response.
Countryman, US. Tel: +1 650 364 9988.

Voice processor
The VIP-digital is voice processor that uses chipcards to store personal setups of announcers and talent. DSP powered and bundled with Waves reverb, presets are stored on a smart card with system setup

Microboards StartREC

A CD-R duplicator that looks and acts like a piece of regular studio gear? **Tim Frost** takes a look

THIS IS WHERE studio duplicators grow up. Gone is the industrial CD duplicator with its mini-tower approach, folded metal casings, low-cost pressure sensitive decal control panels and off-the-shelf LCDs. In comes the 19-inch rackmount that looks like a pro DAT machine with a control panel that is pure audio industry and a back panel that really was designed for the studio.

StartREC comes in a 4U-high rack package that comfortably holds the control panel and its five CD drives—a dedicated 40x Plextor read drive and four 8x CD-R burners. This panel has the normal —well, normal for a recorder—play, record stop, forward, rewind, buttons, and level controls. The LCD has level meters and a sector that is used for control menus and status information. There is also a headphone socket and level control—another rarity on a CD duplicator.

The rear of the unit gives further evidence that the system has been designed from the ground up as a studio copier. Replacing the normally obligatory RS232 ports and keyboard connector are balanced analogue XLR input-outputs and unbalanced phono input-outputs, an AES-EBU XLR, coax SPDIF input, optical SPDIF (TOSlink) input-outputs and a socket for a remote control—an optional extra.

When mounted, it looks, and acts, like any other bit of studio kit and has the huge advantage over other duplicators because it is capable of creating and editing together a master from external sources, as well as duplicating discs. StartREC is effectively a master CD recorder and duplicator in one package.

The system has three separate modes of operation; Copy for copying complete CDs, Track Extraction for loading individual tracks or a complete list of tracks from a CD to the hard-disk and Audio for editing tracks on the internal 6Gb hard disk.

Each has its own control and menu system which is loaded from the internal hard disk separately. While this seems a long-winded way of going about things, it does simplify the way the system operates as you don't have to page through endless numbers of options.

Making copies is as straightforward as you would expect from any copier system now. StartREC will identify the type of disc being copied (it doesn't have to be audio disc) and

NEW TECHNOLOGIES

translated in to MTC and LTC. Providing a machine is transmitting constant time code the TimeMachine can be set to a Precision Follow Mode when connected to a sequencing or HD system in which it will follow fast forward and rewind. In jog-shuttle mode the system will show accurate machine position in its own timeline.

C-Lab, Germany. Tel: +49 406 944 000.

PostConform 2.0

Digidesign has released PostConform 2.0, an update of its autoconforming solution for Mac OS-based computers, which is now fully compatible with Pro Tools | 24 and Pro Tools | 24 MIX systems. 2.0 is a stand-alone application that supports Pro Tools 4.x and 5.x session file formats and HFS and HFS+ Mac OS disk formats. The software is compatible with the newer G3 and G4 Macintosh computers and can use USB connections for Machine Control and MIDI Time Code. It also supports the Digidesign Universal Slave Driver.

Digidesign, US. Tel: +1 650 842 7900.

Amp DSP module

QSC has introduced a digital signal processing module that offers two channels of DSP that attach to the back of most Data-Port-equipped QSC amps. Each channel has high-pass and low-pass crossover filters, high-pass and low-pass shelf filters, signal delay, compression, peak limiting,

go into disc-to-disc copy which simply makes direct copies from the master to the four slaves. Alternatively it will copy the contents of the disc to the internal hard disk—important when you are creating a run of discs and won't want to keep reloading the master. Sub indexes can be copied and the user has the choice of record speed—1x to 8x.

The Track Extract function executes the simple transfer of existing CD music tracks either piecemeal or the entire contents of a disc. It can transfer the Sub Index information and ignores the SCMS settings on the incoming content, although it does offer full control over the SCMS setting for the copies. This all runs smoothly and the 8x drives means that four copies can be made in well under ten minutes.

What makes StartREC really useful in the studio environment is that you no longer need an audio CD-R recorder to create the master disc to copy from. Using the Audio menu, anything from an outside source—analogue or digital—can be recorded onto the hard disc and then edited to top and tail tracks, move, or erase them. Audio quality on the analogue inputs appears excellent, although users are most likely to be coming into StartREC from one of the three digital inputs.

Like other CD-R audio recorders, the Record mode offers auto-mark to increment the track numbering when there is silence. Unlike consumer units, this unit offers full control of both the detection level (from -36db to -48db) and duration of the silence (3s to 9s) that will trigger the auto track numbering.

The logical process for creating a master is to use the Track Extract or Audio input menu to bring the tracks into the system and store them on the hard disk as separate tracks. Then any mistakes, wrong tracks, wrong orders of tracks, false starts or need for fine tuning dealt with using the edit functions. The start and finish points of a track can be altered by selecting an index point splitting the track, and deleting the unwanted bit—following standard MiniDisc editing procedures. Start-REC's editing functions also include track moving, erasing and fade-in and fade-out.

If there are any negatives it's the need to load the individual software menus separately rather than having an additional control pad. Also the system doesn't support CD-RW, although given that all the possible recording and editing functions are carried out on the internal hard disk, this is hardly a practical limitation.

Combining the master recording, basic editing and copy functions in the one rack-mounting box, with dedicated controls and input-outputs creates a more relevant copying tool for studio applications than a simple duplicator. Microboards is to be highly commended for approaching studio duplication afresh and moving CD-R audio duplication one step beyond the half-way-modified add-on-some-audio-functions approach that has largely categorised the market so far. ∎

Lucid SRC9624

The conflicts presented by the variety of digital sampling rates can limit quality and creativity. **Rob James** prepares for conversion

WHEN DIGITAL AUDIO BEGAN its inexorable rise, a number of practical, operational problems became apparent. From the outset, several sampling rates were in common use. Of these, 32kHz has dropped out of favour in production since its only advantage is economy of storage. Higher up the scale, 44.1kHz was adopted for CDs while, for many reasons, 48kHz became the 'professional' standard. Of late, the situation has been further complicated by the advent of double and quad-rate sampling. In sound-for-picture, life tends toward greater complexity largely due to the compromises inherent in NTSC television standards (like 29.97 drop frame) and the requirements of film running at 24 or 25 frames per second. To cut to the chase: if you wish to remain in the digital domain

throughout a production, a means of converting between sample rates is a necessity.

Of course there is the alternative of converting to analogue and reconverting to digital at the required sample rate. In some cases this has been a preferred option since early convertors were either seriously costly or not particularly transparent. One pro CD player actually used this technique as an economical method of accepting wordclock to sync with other equipment.

Lucid's SRC9624, then, is described as a 'high-definition sample-rate convertor'. It provides two independent channels of conversion between all the normal rates and, by using an appropriate external sync source, the pull-ups and downs required for TV and film work. Alternatively it can function as a single-channel convertor at the 'double' sampling rates. It also offers dither to handle changes in bit rate and can convert between physical formats.

The shallow 1U-high box has a finely sculpted brushed-alloy front panel. LED stacks indicate the selected functions and status while all parameter selection is via detented mini toggle switches allowing the shortest route to a desired parameter. Although the LED stacks are laid out vertically they are circular in function, so if the required selection is at the top and the current one at the bottom it only takes one click down to move back to the top. The mains switch is a slightly incongruous horizontal plastic rocker.

On the rear, four blocks of connectors provide the audio I-O each with TOSlink optical, BNC for AES3 or SPDIF (phono adaptors are supplied) and XLR for AES-EBU. All outputs are independent so may be used together. A further block has the sync connections, an XLR for AES-EBU and two BNCs for wordclock in and out. The wordclock input is

bridging and will require external termination if it is the last device in a chain. Wordclock output is always the same frequency as the selected sample rate of the audio output.

Four routeings are possible. The Independent routeing option offers the A and B inputs to the A and B outputs respectively. The Distribution option routes the A input to both outputs. The 96kHz Dual AES (source) requires a 2-wire AES signal on inputs A and B and merges the signals to give a single-wire output at the chosen rate via Output A while the 96kHz Dual AES (receiver) does the converse, accepting a single wire input on Input A and converting to two wire on Outputs A and B.

LEDs indicate the settings of the status bits in the input signals. Signal sources, output sample rate and output dither are selected using toggle switches. The output sample rate can be derived from audio Input A, either sync input or internally generated at 32kHz, 44.1kHz, 48kHz, 88.2kHz or 96kHz. Whenever an unlocked or invalid audio input or sync source is selected the relevant LED blinks. Dither is flat triangular PDF (probability density function). This was chosen since, if the resulting signal is subsequently redithered, it causes fewer problems than noise-shaping types.

As a function of the convertor chip employed (Crystal Semiconductor CS8420), the performance is relatively transparent at conversion ratios of 1.7:1 or less. At ratios higher than this—like the 2:1 required when going from 96kHz to 48kHz—the distortion performance is slightly poorer. As a subjective test on the 96kHz to 48kHz conversion I used some sine wave tones generated in Samplitude. I could not detect any noticeable increase in distortion between a 48kHz source with no conversion and a 96kHz source down converted to 48kHz.

In operation, the SRC9624 is quick and convenient. The toggle switches are positive and intuitive. Whenever parameters are changed the unit appears to mute the outputs, neatly avoiding high-level splats. Around half a second after any parameter change the panel LEDs blink, indicating that the setup has been stored. The same setup will be restored on a subsequent power up.

Leaving aside the future proofing of the high-bit-rate options, the two channels proved very useful at normal sampling rates when used as an interface with a DAT recorder. This allows source material from tapes at 44.1kHz and 48kHz to be loaded into a DAW and the finished product to be recorded back to DAT at both sampling rates without any repatching.

The SRC9624 is an attractive package. Its versatility and good looks will find it many converts. ■

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six parametric filters, 2:1 mixer, signal splitter, signal mute, 0.1dB attenuation steps, signal polarity reversal and selectable 1V/3V input sensitivity. DSP configuration is performed via software and RS232 and the module operates at 24-bit/48kHz.
QSC, US. Tel: +1 714 957 7100.

KT delays
Klark Teknik has unveiled two delay lines. The DN7454 is successor to the DN7204 while the DN7453 supersedes the DN7103. The former is a 2-in, 4-out device while the

latter has one input and three outputs. Each device provides 5.4s of delay while inputs have seven EQ filters which can be configured as fully parametric, high-low pass or high-low shelving. Outputs have the same EQ options with two additional all-pass filters for phase adjustment and a compressor/limiter.
KT, UK. Tel: +44 1562 741515.

Palmer
New Palmer amps include the 4-channel 1500LX4 (2x250W/2x500W 4Ω) and 2200LX4 (2x400W/2x700W 4Ω) with protection circuitry and built-in crossovers. Ratings in to 8Ω in bridged mono mode are 1x500W/1x1000W and 1x800/1x1400W respectively.
Adam Hall, Germany. Tel: +49 608 194 J 90.

TEF System 20 v4.0
TEF is now shipping TDS v4.0 Windows software for the TEF System 20. The software is available as an upgrade for TEF System 20 users and provides RT60, Alcons, and the ability to rapidly set delays and analyse room reflections. Measurements can be accurately accomplished even in noisy work areas. All TEF products are available factory-direct from Gold Line via their website.
Gold-Line, US. Tel: +1 203 938 2588.
Net: www.gold-line.com

Dolby's metadata box
Dolby's DP571 multichannel authoring tool accepts eight channels of PCM audio and then feeds the audio accompanied by a separate metadata bitstream to a Dolby DP571 Dolby E encoder for further distribution or a DP569 Dolby Digital encoder for transmission to the consumer. This allows producers to assign metadata parameters that will reach consumer playback systems via programmes with Dolby Digital and optimise presentation quality. The DP570 has multiple speaker selection, audio channel routeing, separate inputs for Dolby Surround Pro Logic decoding and 5.1 channel monitor configuration.
Dolby, US. Tel: +1 415 558 0200.

Luna 2496 DSP lands
Luna 2496 DSP is described as an uncompromising audio I-O interface system for Windows and Macs. It features a large num-

Universal Audio 1176LN

While most reissued kit revives the past, little is as accurate as UA's comp-limiter. **George Shilling** travels back in time

IT IS OVER two years since I reviewed the Purple Audio MC76, an enhanced recreation of the Urei 1176LN, my favourite desert-island compressor-limiter. Perhaps awakened by the continued demand for this design, the two sons of original designer, the late MT 'Bill' Putnam, sprang into action to reissue the original model under its original brand name. And a good job they appear to have made of it.

Every last detail has been faithfully reproduced; even the manual cover has been copied, including the textured paper. The spec sheets make authentic reading, with virtually every figure identical, and the circuit diagrams show a similar design, although they are no longer hand-drawn.

The front panel is black, recreating the look of the more favoured versions of the original 1176. The originals went through several revisions, designated A–G, this unit being based on the D/E black-faced models. Later units replaced the class-A output stage with a class-AB push-pull stage, and the transformer input was replaced with a differential op-amp circuit.

All controls feel similar, the unit featuring wonderfully large INPUT and OUTPUT knobs, and ATTACK and RELEASE pots, which are all smooth in operation, the ATTACK retaining its bypass click-off setting. This removes any compression, but leaves the gain controls active, turning the unit into a characterful line amplifier. The illuminated meter is similar in appearance to later Urei designs, with the re-created UA logo featured, and a recessed trim pot across the other side of the panel for zero calibration. Old-fashioned radio-bands-style push-buttons all work exactly the same as on the originals, except that they feel a little stiff and 'new'. Ratios of 4:1, 8:1, 12:1 and 20:1 are available with the central row of buttons, and on the right are selections for meter: Gain Reduction mode, Output Level at +4 or +8, and by pressing the bottom button the unit is powered off. The OFF legending had been mysteriously stickered with on-off symbols, no doubt to satisfy some pernickety modern safety rules. Although, talking of safety, upon removing the cover, I was surprised to see bare wires on the mains connections —a nuisance for any service engineer. However, the approach is admirable: the circuit layout looks similar in every respect, with plug-in transistors inserted into sockets mounted on the board. On the rear panel are tag connectors, exactly like the original, with connections for

signal and also remote metering, although I would imagine this to rarely be required. Apart from a fuse holding IEC mains socket the only addition to the original design is a pair of XLR signal connectors. The previously available Model 301 bolt-on box attached to the tag strip to provide XLRs, but many owners made their own modifications and attached sockets directly to the case. The phono socket for stereo operation is retained. This is labelled 1176SA, and the stereo adaptor must be connected between two units for stereo operation. The timing capacitors are in parallel, so doing this doubles the fastest speed of the Attack time, and stereo operation is fiddly to set up, especially between units of different vintage, due to differing transconductance in the FET. Perhaps the 1176SA circuitry could have been included inside the case, as with the Purple model.

In use the unit feels very much like one of the originals. I was fortunate to have the luxury of David Gilmour's Astoria studio, which boasts several original black-faced and grey Haeco-badged 1176s for comparison, not to mention excellent monitoring. The 1176 uses a FET as a variable resistor to control gain (there are no valves here), and this is the main reason for its unique character. All units were slightly different, but the new model's output gain pot was noticeably changed, with perhaps a slightly different value—I had to turn it further clockwise to match levels. However, in most other respects, operation is identical and the sound very similar, with perhaps a little extra clarity from the new unit. Vocals sounded wonderfully enhanced by this compressor, and in normal use it was hard to hear any difference between the new and old models. Attack and release characteristics seem to have been accurately retained, with an attack range of 20–800µs and release times of 50ms–1100ms. The only really obvious sonic difference was in 'overdrive' mode—the legendary technique of pressing all four Ratio buttons in simultaneously to give a really over-the-top distorted compression. The character of the new unit in this situation was quite different from any of the older models, with much brighter and nastier distortion, and less warmth, especially on fast settings. By backing off the Release speed, a more similar sound could be achieved, but there was still an obvious difference, and although the older units may have been somewhat out of alignment, this was the sound I preferred.

The 1176LN sounds great but the lower-priced Purple competitor is sonically very similar, and features many improvements over the original design. However, as a 'strictly authentic' official reissue with indisputable credentials, the UA is the box to have. ■

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ber of analogue and digital I-Os in 24-bit/96kHz, the new Ultra Low Latency Interface with a comprehensive driver set and three SHARC DSPs for audio processing as well as for mixing and surround sound software applications. The system hardware consists of a new PCI card and an external A–D/D–A convertor box. The card hosts stereo analogue and stereo SPDIF digital inputs and outputs. The convertor box adds another eight analogue I-Os. Luna can operate all of these channels in full 24/96. Furthermore, the system can be expanded with additional I-Os (up to 36 I-Os per unit —two units can be linked via S/TDM bus).
Creamware, Germany. Tel: +49 224 15 95 80.

OMR8 gets controller

DAR has introduced an edit controller for its OMR8 24-bit 8-track system that includes a weighted action wheel and dedicated

function keys. Two recently introduced features of the system include CD mastering and a new fine edit function. CD mastering is accessible from within the OMR8 internal editing software and enables tracks to be prepared for mastering with the OMR8 controlling the CD-R burning process. Fine edit permits two alternative takes to be viewed side by side and edited at waveform level with hi-res zoom.
DAR, UK. Tel: +44 1372 742 848.

RealVerb 5.1 ships

RealVerb 5.1 is a multichannel reverb plug in for Pro Tools with the ability to map reverb spatially for surround mixing and to morph between room shapes and textures. Rather than running parallel reverberators the package uses a single reverberator and crossfades the controls producing intermediate states with no zipper noise.
Kind of Loud, US. Tel: + 1 831 466 3737.

Broadcast mic

Soundelux has released the R-1 cardioid condenser. It features a 1-inch diaphragm with gold on 6-micron Mylar and transformerless FET.
Soundelux, US. Tel: +1 213 464 9601.

N/Dym wireless

The N/DYM series UHF wireless mic system is enhanced by Secure Phase diversity while the receiver has an adjustable squelch level, beltpack, and hand-held transmitters include Tx gain controls. Available in two hand-held versions, an omni lavalier, cardioid lavalier, a headset and guitar system.
EVI Audio, Germany. Tel: +49 9421 7060.

www.americanradiohistory.com

Steve Power

Interviewed the day after he finished working on Robbie Williams' third album, producer Steve Power talks with **George Shilling** about kit, career and characters

READILY TRADING HIS SEAT at a keyboard rig for a seat behind a mixing console, Steve Power's recording career really began riding shotgun for Mick Glossop.

'We got a deal with Virgin, but instead of spending the money on renting studios we built one in Liverpool called The Pink,' he recalls. 'Mick Glossop, whose engineering skills are renowned, produced our album and I assisted him. I was just mad keen —I was 18 and I took everything I could from him, which is a lot, because he is very knowledgeable.

'Before that I had done the band's 4-track demos and I was more into the studio than the keyboards. So when the band went on the road, I made a decision to leave in order to run the studio.'

What had seemed a simple move was, in fact, only the start of an eventful learning curve for the young Englishman. 'I'd only ever worked one way—I wasn't used to "You're the client, and I do what you say",' he explains. 'I just thought people came in and I organised them and got the best out of them. I didn't really think I was producing them but I suppose I was.

'People would come in and do six songs, record and mix in a day, which is very good practise. I was doing demos for Dead or Alive, Frankie Goes to Hollywood, The Lotus Eaters, and several other acts that didn't get anywhere but did get signed up—it was the usual thing of the masters not being as good as the demos.'

Power made the move to London and went straight into producing, working with Steve Lovell among others. Around 1986, when things went quiet for a few months, a chance meeting with another Liverpudlian put him back in the studio.

'One day I was walking past Battery Studios,' he says, 'and out walked Mike Score from A Flock of Seagulls—I knew him from Liverpool and he needed someone to finish engineering his album, so he invited me in and I finished it.'

Another of the producers working at Battery at the time subsequently asked him to engineer a Billy Ocean album that he was producing and suddenly Power found himself with a management contract for Zomba.

'I did The Railway Children, then it went quiet again for a bit around 1992–93 and so I studied classical guitar for two years. That really helped when I came back. I think that was part of the reason why I changed from wanting pristine multitracks to music that affected me. I suddenly realised that they don't buy the multitrack, they buy the stereo, and as long as the stereo's okay… You can get a lot more work done if you're not so cautious with every click and pop on your multitrack. On my multitracks now there's clicks and buzzes all over the place, but there are more spontaneous performances there.'

'As 1993 got underway, Power found himself working with Babylon Zoo on the album that spawned 'Spaceman'. Hard on its heels came Babybird's 'You're Gorgeous' ('Both records that everyone hated...'). And then Guy Chambers asked him to produce

Robbie Williams.

'All those years ago Guy had replaced me in the band.' explains a bemused Steve Power. 'I'd known him since he was 15. Rob wanted to be Oasis at the time and Chris Briggs, who had A&R'd World Party had the idea of bringing in Guy as someone who could work with Bobby and write with him. They had tried various people and it hadn't worked out, but they complement each other so well. Guy's much more of an artist, he's a bit scatterbrained, and he knew I'd be able to hold a net underneath him.

'Often they don't know when they've done a good demo.' he continues. 'I remember 'Strong' and 'No Regrets' for instance —I picked them out as singles straight away from the demos. and Guy didn't think 'Strong' was very good. As regards demos for the new album, when they were touring in America. Rob and Guy get bored very quickly—they've really got no patience, either of them—so they had popped into studios and written songs. There were about 35 or more songs that came through which were quickly culled down to 12, and the last three were written because Rob decided he wanted to get more "programmy" stuff, and they were just done about two weeks before recording started.'

Power professes not to have had a standard way of approaching making master recordings out of these demos.

'Because Bobby is such a large personality you don't have to do just one style of music. you can do anything and it's still him. Something like "Angels" is approached very traditionally. The whole band went out and played—acoustic guitar, piano, drums, bass—and Bob sang while the backing track went down. Of course everything was replaced except the drums.

'I don't spend long on the drum sound though—I find it easy to get drum sounds now, if you don't mic drums too close, as long as they're tuned and hit properly. Ambient miking depends on the song, but I'll maybe put up one ambient mic, I very rarely put up stereo ambience and try and get a big sound that sounds great but is useless within the song. With electric guitars I don't put ambient mics on them either, just one as close as I can get. and that's it, because if you have two you get phase things going on, and it's not as present, and you can always add something after.

'Mind you, I don't like reverb either. I've done almost complete mixes with no reverb, for a long time. If I do use reverb. I'll use a plate. I don't like the sound of Lexicon plate very much. The new one I tried briefly and it sounded nicer, and I tried the tc 6000, and I like the sound of that better. and I'm going to use that tomorrow doing surround mixes of 'Rock DJ' and a track not on the album, and six tracks from Robbie's Slane Castle gig for DVD. It has 5.1 reverbs and 5.1 compression. which I'm intrigued to try.'

The album was recorded 48-track analogue.

'For the last album we mixed to Genex 24-bit machine and half-inch tape, and compared every track at the mastering stage. The Genex won every time. The half-inch added warmth. but took away some of the clarity. which made you want to add some 5kHz to compensate. but with the Genex you didn't need to. So this time we didn't bother with the half-inch and just used the Genex.'

As regards mixing. Power prefers SSL E/G-series consoles to later models.

'When I tried the 9000j. I found it too clean for this rock track I was doing.' he explains. 'So what I've settled on now is Neves and Focusrites. and the DPA 3541 preamp [see sidebar]. I record with as good-quality mic amps as I can. and then I kind of like the crunching up that happens on the old SSL

for the mix. There's an SSL in Battery Studio 4 which has the black knob EQ. which is a slightly softer EQ. That's quite a special desk because it's got the crunch, but you don't find a lot of black-knob desks. You can use it without thinking, whereas when I went to the j—I can't be bothered learning a new computer system that doesn't seem to be based on the same principles. They should make a disk that emulates a G-series computer.'

Power's approach to mixing itself mixes work and play—a recipe he's found helps retain objectivity through the process.

'When I mix I go really quickly through it, and I can't go really quickly with the j-series. because I'm too busy picking the pen up and thinking, "what do I have to say to it?". I like simplicity, nothing getting in the way of just getting done what you want to do quickly. Generally I'll come in, do a bit, have some lunch, read some Ceefax about the football, go and do a bit more. But I do a lot quickly. in half-an-hour or an hour, but then I'll want to not hear it, and I'll go and watch telly for a bit, come back in. do another bit. have my dinner at seven and go home. And I'll take it home, and come in the next day and tweak it a bit, call Guy in and we'll put it down. So I seem to have spent two days on a mix, but I've actually not been doing a great deal most of the time.

'I think it's so important to keep yourself separate from it and objective—if you sit there all day listening to the thing, you can't retain your objectivity.

'Another thing I do to retain objectivity, which people who come to see me mix find very amusing is that I have every set of speakers in the world there. As soon as it sounds good on the KRKs. I'll put it on the AR18s that I've got on the floor, 'cos they were designed to be used on the floor, that's where they've got proper bass. So I switch >

< it over and go "Oh. I didn't notice that middle frequency sticking out a bit", and get rid of that. It sounds okay on them, so I'll switch it up to the PMCs that I borrowed to mix the album, and I'll think it could do with a bit more compression. It just makes me think of different things. If I switch to a different set it gives me a totally different viewpoint. like the situation where you take a mix somewhere else.

'I've got literally about nine different sets of speakers. I'll return to the previous speakers after making a change, to see if I've overdone it. So I get that instant "oh it doesn't sound like it did in the studio" straight away, before I've left the studio. I never use NS10s, by the way, because to me if it is right, it sounds terrible on NS10s, I can't mix to sound terrible. I spend a lot of time listening to other CDs on each set of speakers.

'A lot of what we do is going to be played on the radio. I do squash stuff a lot, and I don't like it, in a sense, but I know that the shop window is radio, and it does need squashing and it does need high end. I compress the mix with the SSL and a Prism Maselec. which is fantastic. On this album I had two sets of the 4-way mic amps, and I had the Prism EQ across my monitors, so I recorded a lot of the time without EQ, so I've never done anything that I'll regret. However, I'm screwing in a lot across the monitors. The quality of those things is absolutely incredible I think, the mic amps, the compressor and the EQ. I've never done anything wrong that I'll regret while I'm recording, however, it's loud and sounds great for the musicians who are working on it, so they can get off on it and get the

'WHEN I WAS AT METROPOLIS I was offered the DPA 3541 microphone kit to try out, and the thing I like most about it is the sound pressure handling. I like the sound of the [Neumann] 149, but you put it on a loud guitar and it can't handle it. And it can't actually handle Bobby when he's yelling. He's got a hell of a loud voice, and when he suddenly decides in the middle of a ballad that 'This is too down, I want a "YEAH YEAH YEAH" bit here,' and he'll suddenly do it completely out of the blue without warning anyone. However, I've found that even when I know he's gonna do it, on some mics like the 149, I'll have the mic amp all the way down, and he'll still distort the mic on the way in, because he's so bloody loud. The DPA can actually handle him, and it sounds kind of similar to the 149, probably because of the treble lift. I thought, if it can handle the volume, it can sound like a condenser on very loud guitar amps and snare drums, which it can. I sometimes like the sound of a 57 on the snare, but sometimes I would like to be able to get more like the sound of a 451, and with this thing you can, and you can get it close, which is a step forward, nobody else has managed that before. I ended up using it on acoustic guitars as well, and there its treble lift was very handy. And I like to use high quality mic amps, and the mic amp you get with it is very good and open.'

'I've got nine different sets of speakers. I'll return to the previous speakers after making a change, to see if I've overdone it.'

excitement, but I haven't done it to tape. So when I'm mixing I've got that, I've got their compressor there as well. Because if you mix and then you take it to the mastering room with a view to. "I'll put the top on then," it can change the balance. I know what I'm doing when I'm mixing—I'd say it's mainly for radio, but then a lot of bog-standard hi-fi's and ghetto blasters are quite like radios to me, and that is our selling point really, not to the hi-fi people. But there are tracks where you can get a bit more of that, generally the ballads.'

Robbie Williams is only the most recent notable artist to have come Power's way. Joe Cocker, Geri Halliwell (for Chris Briggs). The Dum Dums and new artist Rebecca Ryan have all helped keep him busy.

'Chris is the only A&R man you can play a monitor mix to.' Power enthuses. 'He once told me he'd like to sack all A&R men and bring in out-of-work record producers,

because they understand what goes on.'

Other notable achievements include mixes for various rock acts that have left Power hungry for more—in good time.

'I'd really like to work with Sou wax,' he confirms. 'They are a really talented Belgian band and I've mixed their next single. Also Feeder—I did some additional production and remix for two singles which got them on the radio. Then there's an artist called Rebecca Ryan on Zomba that I'm hopefully going to do after a break and after the DVD. But I'm definitely going to have a break first.'

With a career characterised by its stops and starts, a break might sound like a dangerous option.

'Usually there are a few things queuing up.' he says. 'and I can only do one of them. At the moment I'm in the good period but of course, I'll be at home for two weeks and then think, "Oh my God...." '■

Head case

In most conversations, monitoring invariably means loudspeakers but headphones play are valuable too.
Derek Johnson & **Debbie Poyser** audition the options

BUYING STUDIO HEADPHONES seems an uncomplicated proposition—and typically people put rather less thought into choosing their cans than their monitors. Yet most studio musicians and engineers spend a fair amount of time wearing headphones, so selecting the right ones is worth some effort.

First, there's the choice between enclosed and open types. Enclosed phones usually offer higher levels and increased bass perception, and as they allow little sound to escape they're the most appropriate for providing cue mixes, where any leakage could make it onto tape (or disk). They also let in little external sound, so are especially suitable for live engineering in noisy environments. Good open headphones are usually prone to less colouration than enclosed ones (making them more useful for mix checking) and are less fatiguing, but they allow more sound to escape and to enter. Semi-open phones provide a compromise between the strengths of both types.

Then there is impedance to consider: the best sound quality is produced by high-impedance headphones, but low-impedance phones offer higher listening levels (though they tend to more distortion) and so are often preferred by location recordists. In the studio, probably the most important thing is to ensure everyone has access to the same type of phones. If there are a mixture of types, some listeners could experience problems with volume levels.

Quality headphones from major manufacturers span a wide price range (from under £30 to almost £200 ex VAT in the UK). It's impossible to cover all models here, but we can at least provide a taster of what's available. Starting at the beginning of the alphabet, AKG models are affordable right across the range, and are used in studios world wide. At the lower end is the K70 (at £27), a lightweight, open-back, hi-fi-style phone with foam earpads, a 20Hz–20kHz frequency response and 100Ω impedance—a good bet if multiple sets are needed. Going up a step, the popular semi-open K240DF (£85) has a 600Ω impedance, 15Hz–20kHz frequency response, and a studio-grade build with deeply padded ear-cushions. If it is a quality fully enclosed phone you're after, the K270S (£110), with a 20Hz–28kHz frequency response and 75Ω impedance, could fit the bill.

Choosing between Audio Technica's headphones is easy, as AT markets just three models in the UK, all enclosed —the 66Ω ATHD40 and 60Ω ATHM40 Studiophones, and the 30Ω ATH910 Pro. The D40 and M40 (£85) are identical in appearance, with padded headband, cushioned earpieces, and a good upper frequency response of 28kHz. However, the D40 has LF extending down to 5Hz, compared to the M40's 20Hz. Both Studiophones are, usefully, fully serviceable, with earpieces, elements and cables being replaceable. The ATH910 Pro model (20Hz–22kHz, 40Ω, £68) is recommended for home listening and studio monitoring, but isn't field-serviceable.

Many of Beyerdynamic's headphones distinguish themselves by being modular, allowing defective parts to be removed and replaced individually. At the lower end of the range is the DT220 (£79), an enclosed phone with 20Hz–20kHz frequency response and 20Ω or 400Ω impedance. The fully serviceable, mid-priced DT100 (£109) is an enclosed model widely used in the studio industry. It has a solid construction and a 30Hz–20kHz frequency response, and is also available with a variety of impedances: 8Ω, 400Ω or 2000Ω. The DT150 (£129) is similar to the DT100 but has an extended frequency response of 5Hz–30kHz. Those looking for an open-backed phone might want to check out the DT990 Pro (£119), which has a remarkable 5Hz–35kHz frequency response and a 600Ω impedance. Beyer especially recommend the 990 for critical post and broadcast studio monitoring, but it and its enclosed version, the DT770, are accepted as fine all-rounders.

At least two of Fostex' four models of headphone are apparently gaining wide acceptance in US studios. Both the semi-open T20RP (£83) and fully enclosed T40RP (£89) are field-serviceable, courtesy of a screwed assembly. The T20 boasts a 20Hz–30kHz frequency response, while the T40's response is a slightly less extended 25Hz–20kHz. Both phones have a 50Ω impedance, deeply padded ear-cushions and padded headband. The 44Ω T5 (£46) and 70Ω T7 (£65), both with a 20Hz–20kHz frequency response, are described as good general-purpose phones, their main distinguishing characteristic being ultra-light construction for extended wearing comfort. **>**

< Sennheiser markets a large range of headphones, including enclosed pro models and open-backed hi-fi designs. The 32Ω HD500 (£60) is classed as a hi-fi phone, yet it has studio-friendly characteristics, including a more than respectable 14Hz–21kHz frequency response and unusually large 'circum- aural' earpieces which encircle the ear, with the phones sitting against the head rather than pressing on the ears. Open construction and low weight help ensure comfortable wear. From the stu- dio range comes the HD250 Linear II (£128), an enclosed phone which is apparently one of Sennheiser's best sell- ers. Again, the cushioned earpads are designed to fit around the ears, and Sennheiser claim that though the phones are enclosed they offer the feel of an open design. Frequency response is a creditable 10Hz–25kHz, and impedance is 300Ω. Then there's the HD25 (£136), a rugged enclosed headphone recommended for studio and live sound applications, with a 16Hz –22kHz frequency response, a choice of 70Ω or 600Ω impedance, and a con- ventional 'on-ear' design.

Sony produces many headphones, a few of which are dedicated studio devices. The professional MDR7502 (£60) is an affordable enclosed phone described as general-purpose, with lightweight construction, 45Ω imped- ance and 60Hz–16kHz frequency re- sponse. The MDR7505 (£95) is another enclosed model with an improved 10Hz–25kHz response and a 40Ω im- pedance. It's recommended for re- cording, remix and DJ use—boasting a swivelling earpiece for single-sided monitoring—and has a build quality described by Sony as capable of with- standing years of daily handling. The MDR7509s (£185) is termed a profes- sional reference monitoring headphone, with an appropriately wide frequency response (5Hz–30kHz), high power handling, an enclosed, cir- cumaural earpiece design, and robust construction.

Studios on a tight bud- get needing many sets of identical phones and pre- pared to be open-minded about brand-name may be interested to hear that Studiospares recommend the low-impedance Altai HVS22, at just £12. It has a 40Hz–18kHz frequency response, and Studiospares say its performance comes close to that of models costing six times the price. Something for every pocket, then... ■

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Mother, Jobs and speed

With Apple back on its feet, the Mac-PC race is on once again. **Martin Polon** explores the likely impact of forthcoming computer technology

HAVING FINALLY ENTERED the millennium, and having survived the Y2k miasma, we now face a new Millennium of computing and related technologies. We also find ourselves obsessed with processor speeds—Intel and AMD seem to be jumping over each other to reach gigahertz speeds and beyond for the PC.

By now, nearly every one is aware of the remarkable recovery of Apple Computer—ostensibly spurred on by the efforts of Apple founder Steve Jobs, returned to the fold for several years. The recovery was accentuated by the mid-April 2000 announcement of a two-for-one stock split, the first such action in over 13 years for the Cupertino-based company. The stock itself is selling at $120 per share pre-split (as of 20th April, 2000), or about ten times the asking price previously during the company's near-death experience. Apple has even managed to maintain it's share strength on the NASDAQ technology exchange, as that market place has sustained great wounds on many of the most significant technology sector and dot.com stocks listed.

Apple's revival has been based on carefully-implemented technology innovations and strong sales of the i-Mac, professional desktop G3 and now G4 and PowerBook computers. The success of these technologies has surpassed even the fastest and most recent innovations available on PCs. Despite the fact, that Apple desktop computers with G4 chips are still only rated at 450MHz-500MHz clock speeds (as opposed to the GHz range of Intel and AMD top-of-the-line chips), Apple computers consistently out perform their supposedly faster PC brethren.

Apple's Power PC chips have profited from their ability to perform more computing tasks, during each clock cycle than a PC microprocessor in a similar time frame. In addition, the Power PC G4 chips use a 'velocity engine' developed by Motorola and now adopted by IBM (Apple's two captive chip foundries) which operates as a vector processing unit. This so-called Altivec feature set means that graphical or multimedia content-heavy computer programs run more quickly and smoothly. Obviously, this also means that audio programs specifically designed to take advantage of the Altivec feature will offer greater speed and functionality than those that don't.

A further bonus is the co-operation finally restored between Apple, IBM and Motorola in terms of chip design, features, and chip production—a condition that had evaded the so-called 'AIM' triumvirate for several years, but now is finally operating again more like a finely tuned Swiss watch than not.

At any rate, when we measure the benefit of the G4-based Mac Power PC against that of an Intel or an AMD PC engine the difference is impressive. It is especially so when used to measure the performance difference between Macs and PCs operating in a power-hungry graphics processing environment, which is analogous to the demands of audio processing. In several recently published tests, the most important of which was Henry Norr's in the *San Francisco Chronicle*, one method of testing between the two computing platforms was to use Adobe's Photoshop (v5.5) as a common test medium. This was done specifically, because it is the one piece of power using software that is closer to a common denominator than almost any other computer program available. The importance of this is significant, because other specifically designed computer benchmarks are rather platform exact with various benchmarks available for the Mac platform and various benchmarks available for the PC platform. But

there are none that are not common to only the one platform or the other. In other words, these measurement tools that work with the PC won't work with the Mac—making it very difficult to measure between the two systems.

The tests were designed so that by using the same amounts of random access memory in both PCs and the Mac to be tested (256k), and specifying the same tasks and the identical image to be operated upon within Photoshop, the processing speeds yielded would represent identical task outcomes. It was clear after the tests were completed, that the Mac computer, using a processor ostensibly half as fast as the chips on the PC, outperformed the PCs on the graphics heavy transactions in two-thirds of the time required by the PCs.

What is so fascinating about the direction indicated for the future of Macintosh computing, is the research advances and chip fabrication techniques produced recently by IBM's research labs. The IBM semiconductor research facilities have won five Nobel prizes for work done in semiconductor physics. That is an almost unbelievable number for the research output of one company. But IBM is not an ordinary company and the breakthroughs that have been recently announced, promise to increase the power and the speed of computers using IBM's microprocessors as well as reducing the heat generated and the size of the chips. The advances are as follows:

The future will see use of multiple local clocks on board IBM-designed microprocessor chips including the future range of Power PC chips, as opposed to the current practice of using a single central clock to run the entire chip. This new technique is known as Interlocked Pipelined CMOS (IPCMOS) technology The use of multiple clocks will allow faster elements of a chip to run at their own speed—not having to wait for a system-wide clock cycle to proceed. The technique removes the current limitation of the fastest elements on a microprocessor having to wait for the slowest elements and could eventually yield operations in the 3bn to 4.5bn cycles/second using conventional or unconventional silicon circuitry.

Second, it is the use of another method of insulating size-reduced chips from the Internal interference caused by the closeness of connections known as silicon-on-insulator or SOI. This technology will have an immediate impact on chip fabrication and may influence succeeding generations of chips.

Third, is the continuation of the move from aluminium-based chips to copper-based chips. This advance too, came from IBM's research laboratories with the initial announcement made in 1997. The use of copper allows circuits to be made physically thinner without the loss of any ability to move electrons about

the chip and to run the copper chips cooler and to use less electricity. IBM and the other Apple partner Motorola, pioneered this technology as well and other chip makers are also beginning to transition to copper this year. In fact, it would be safe to say that the year 2000 will be known in the semiconductor industry as the 'run' to copper. It is clear that despite a history of using aluminium as a microprocessor and other chip substrate in some form for the last 30–40 years, the transition to copper will be complete for any new technology products sooner than later. It is only in the realm of high current (non-computing) products that aluminium chip substrates will hang on in the short term. The multiple demands in computing and computer devices of speed, size, heat and reduced power consumption will guarantee the ascent of copper.

Fourth, IBM now proposes to build power PC chips with circuits of only 0.13 microns in width. This is approximately 30% smaller than the current 0.18 micron width in use in today's microprocessor chips. The new chips will also use an insulation spacing of only 0.1 micron between two individual circuits. Although the smaller spacing promises speed advances to as high a clock rate as 3MHz or better, the need to maintain inter-circuit isolation becomes that much more important. This is especially true, since the space between on-chip wiring and circuits could be measured in the thousands or even the hundreds of atoms. Intercircuit electrical interference must be suppressed or else any chip using these micro dimensions will become unreliable. Current chips from all chipmakers use glass-like silicon-dioxide-based insulation. That method of insulating is not appropriate to the incredibly small dimensions of the 0.13 micron chips.

Fifth, to insulate for the new chips, IBM has developed the use of an organic polymer material known as a 'low-k dielectric' or SiLK which is baked on between layers of more conventional oxide materials in a sandwich. IBM has learned how to manage the incredible difficulty of using polymer material, so it can be layered and sculpted to ultra-precise specifications around the copper circuitry. All of this to shield the millions of individual copper circuits on the microprocessor chip, reducing the electrical crosstalk between circuits that could denigrate chip performance (speed) and waste electrical power on the chip. The low-k (or low capacitance) plastic material is a semiconductor dielectric produced by Dow Chemical Co to IBM's proprietary specifications. Curiously, the Apple computer partner Motorola is the other major chip maker to be working with SiLK, although its plans for using the organic polymer material do not indicate chip production until 2002 at the earliest. IBM chips

using this and some of, if not all of, the other new technologies discussed above will be available for commercial usage beginning in 2001.

The development of all of this extra-ordinary microprocessor technology is indeed challenging, but where does that leave those of us in the audio industry who have to make on-going decisions for the future as to platforms and functionality? The first issue that appears clear is that both IBM and Motorola will continue in one form or another to serve Apple's bidding for microprocessor chips for Macintosh computers. Whether the two companies exist in rigid logic-step or simply operate in relatively parallel positions, they both are married to the Power PC chip and that is not going to change soon or easily or even at all.

It is also clear that the lead IBM has taken with these microprocessor fabrication developments will place it one to two generations ahead of the Intel-AMD camp in the development of PC microprocessors. And, as we have already seen of the lead held today by the supposedly inferior Power PC chips (inferior in terms of raw clock numbers rather than throughput), the future will be measured not in raw clocked numbers but by the use of multiple chip clocks—the several insulation techniques used to reduce chip die size and increase speed—to the gains made in chip fabrication implementation.

In other words, the speed advantage held by Power PC chips over supposedly faster Gigahertz-range PC chips, will in all likelihood remain. And as to how long two generations of chip implementation will represent, it is possible to look at the current rule-of-thumb that says one chip generation equals one year. This may not hold true in the compressed universe of chip fabrication and development in the new Millennium, but it is clear that whatever happens in the Intel-AMD camp, IBM will not be sitting on its hands.

It is estimated by analysts in the microprocessor industry, that IBM's lead in technology should yield GHz Power PC chips by the end of this year or the beginning of next and potentially two and three GHz chips in forthcoming years. It is not unlikely that by 2005, the average Power PC chip will be running at 5GHz or better. However, despite this very pretty picture there is one problem. Despite the advances that have been made and which have yielded powerful and significant improvements in video editing and manipulation technology, audio has lagged far behind on both the PC and Macintosh platform. Audio software has to be written to take advantage of these incredible future advances in chip design. That this will happen is very likely, but when and with what standards remains the huge question mark hanging over the audio industry. ■

Surround Sound
Microphone Recording

The second part of our surround sound investigation highlights a study in microphone placement. **Dave Foister** follows a series of purist Swiss recording sessions

STEREO HAS BEEN with us for more than a century, and we're still finding new ways of recording it. Generations of engineers and producers have explored an enormous spectrum of techniques, and no clear winner has emerged—and never will, since the concept of a perfect multipurpose technique is nonsense. If there's a rule, it's along the lines of horses for courses, or perhaps better still there are no rules—if it works in a given situation, do it.

By comparison, surround is a newcomer. Even allowing for the quad explorations of the sixties and seventies, and the adoption of Ambisonics and other methods by small bands of enthusiasts, it is a fraction of the age of 2-channel stereo, and widespread use of surround intended for a real consumer market is still in its infancy. The consumer format that has emerged as the *de facto* standard may not be what music recordists would have chosen, but the cinema-dictated 5-channel layout appears to be here to stay. Idealists continue to complain that it's inappropriate for music, and that yet another opportunity has been missed, another second-best system has won, while pragmatists are rolling their sleeves up and finding out how to make it work. While manufacturers are beginning to offer products that address 5.1 recording, effects and mastering, astute recording outfits are trying various approaches to miking an acoustic event, paralleling the decades of work spent identifying the strengths of the various stereo techniques. A particularly enthusiastic team is Swiss company Sound Arts AG.

Sound Arts is a group of former Detmold Tonmeisters who have established a niche for themselves in location recording, production and editing for the classical CD market, including their own Pan Classics label. The team of Clement Spiess, Koichiro Hattori, Jens Jamin, and Simon Fox share out the roles of producer, balance engineer and editor in every possible combination depending on the nature of the job. Their relationship with Lucerne's Arts and Conference Centre was highlighted in our review of the SPL-Brauner Atmos 5.1 microphone array last year, and it allows them to explore various methods within a known, acoustically successful environment. The SPL recording was one of a series carried out in the Lucerne hall, as well as in a favourite church up in the mountains outside Lucerne. This time the focus of my visit was a recording of the Central Swiss Youth Orchestra, with a programme including Bruckner's 6th symphony, and the central surround element this time was to be the Soundfield Mk.V microphone along with its new surround partner, the SP451 processor. This eagerly awaited adjunct to the established microphone system is still in development, and a prototype fitted with the first output format card was kindly lent by Soundfield Research for the purposes of the trial.

In its way, Sound Arts' fundamental aim is simple—to produce a realistic impression of an acoustic performance from the point of view of an ideally placed member of the audience. There is no attempt to do anything fancy with the surround channels, as anything more than recreating the sense of the original space would distract from the central purpose. At the same time it becomes manifestly clear on first hearing of a good surround recording of this type that the rear channels add enor-

mously to the depth and perspective of the sounds in front of the listener, removing the sense of listening through a window that stereo could be said to give.

Few microphone arrays designed specifically for surround recording have ever been available. There was for a time a special quadraphonic version of the Neumann SM69, with four capsules instead of the usual two; there is the SPL system referred to earlier; and there has been for many years the Soundfield microphone, designed for full spherical surround reproduction, although its surround capabilities have often played second fiddle to its flexibility and sheer quality as a stereo system and even as a mono microphone. I know of users who have owned a Soundfield for years and never done any surround recording with it. The fact remains, however, that its tetrahedral capsule array (not one in each corner as many imagine) has the capability of capturing a complete spherical soundfield, although few decoders capable of delivering this have ever been produced. On the other hand, surround without height of the kind we are now becoming accustomed to is second nature to the Soundfield, and Soundfield Research is now beginning to offer us real-world processors to exploit this potential.

The device loaned to Sound Arts for this recording was the SP451, a 1U-high processor that takes the B-format signals from the microphone control box and delivers multiple surround channels in whatever format has been chosen. There are eight output XLRs on the back, allowing delivery of all kinds of standards and layouts. The actual choice of how the signals are presented is

dependent on a card fitted in one of two slots in the chassis. At the moment there is only one such card available, and this is therefore what was used. The fact that this card has the name Microphone Array Pattern (MAP) gives a strong hint as to what the approach is at this stage.

The conventional stereo output of a Soundfield is derived by mixing the B-format signals in a manner similar to MS techniques, but with considerably more flexibility. As a quick reminder, these B-format signals comprise four virtual microphone signals: three figure-of-eights pointing forward, sideways and up, and an omni. Mixing the front (X) and side (Y) gives figure-of-eights pointing in any direction, and adding the omni (W) to these modifies the polar pattern to any first-order configuration you want. It is thus possible to synthesise any number of microphones of any pattern and pointing in any direction from a single Soundfield, and indeed the Mark III could do four simultaneously in a symmetrical array. The SP451's MAP card takes this a stage further, and creates five cardioid microphones pointing nominally in the directions of a standard 5-speaker surround layout. Controls are provided for altering the widths of the front LR and rear pairs, for altering the polar patterns of the rear-facing microphones, and for adjusting the relative levels.

It doesn't take much thought to work out that for music recording, the centre microphone is going to seriously reduce the apparent width of the frontal image the microphone is picking up: our feeling was that if it was used at all, it should be at least 10–12dB below the levels of the others, and this was confirmed by Soundfield as the intended use afterwards. Its purpose is for recordings where a strong front centre image is required, like a dialogue recording.

Having established this, the sound stage the microphone produced was impressively natural in a restrained sort of way. Those wanting particularly noticeable ping-pong effects would be disappointed, but the same is true of coincident cardioids in stereo, which never give as spacious a result as figure-of-eights. At the same time, the complete lack of any antiphase components or arrival-time comb filtering created a stable and convincing image, tolerant of moving around the listening space, and capable of being collapsed into stereo or mono without any ill effects. The adjustments were interesting and useful; the width controls did what one would have hoped, particularly once the centre microphone had been appropriately adjusted, and the polar pattern adjustment on the back pair had the expected effect of putting more signal on the rear channels. Turning it towards omni brought the frontal image forward into the room, while the other direction, by effectively putting front signal out of phase on the backs, exaggerated the front-back separation and enhanced the sense of space.

A major benefit of B-format work is that all this kind of adjustment can take place in postproduction, just as MS allows later adjustment of stereo images. Recording the straight B-format from a Mk.V, an ST250, or an Ambisonics mixing system requires only four channels (or even three if the vertical component, not used in the SP451, is discarded), and the entire range of adjustment and processing is available exactly as it would have been on the microphone live. This is useful enough in simple stereo, where a location B-format recording can have all the stereo configuration decisions made in a known control room after the event, and the additional power here to control the surround stage in such detail and so simply could become a huge advantage.

Those of us who have heard true Ambisonics derived from a single Soundfield will know that there is much more to be had from this microphone, including the facility to map an extraordinarily natural surround field on to whatever loudspeaker layout is available. It is to be hoped that future developments of the SP451 will exploit Soundfield's technology to the full and include decoding to what has become known as G-format, which is five Ambisonically-decoded speaker feeds intended for the 'standard' layout in use here. Meanwhile the 5-cardioid MAP card is a significant step in the right direction and another useful addition to the small >

Sound Arts' Simon Fox and Koichiro Hattori (seated) at the Yamaha 02R with Soundfield SP451 and MkV controller to the left. Manger Zerobox 109 speakers in the background
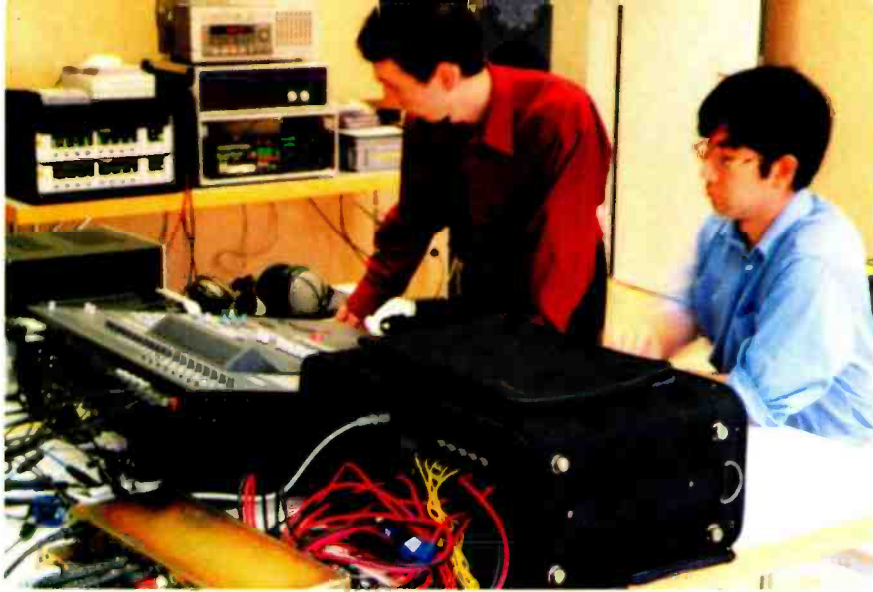
Fox and Hattori with Studer mic preamps, Genex and Tascam DA-88 recorder to hand

< surround armoury. By the way guys, it's spatial. not spacial.

Having spent time playing with the Soundfield. we went on to review some of the other surround trials Sound Arts has undertaken, played over an extraordinary set of five Manger Zerobox 109 loudspeakers, which use a new design of lightweight broadband transducer to minimise the transient problems associated with the mechanics of conventional drivers. These innovative speakers have attracted considerable attention in their native Germany, and did an excellent job of conveying what we were hearing in a natural, transparent way, undoubtedly helping with the judgement of the spatial aspects. We began with the final surround mix of the Atmos 5.1 recording, complete with surround-panned spot microphones, that I had heard in its raw state on my previous visit. This had used a large array of spots throughout the orchestra, and the care taken in the mix with the main SPL array had produced a very impressive and involving recording. Sometimes the balance between instrumental detail and a proper sense of the recording space is hard to achieve even in stereo, but this setup certainly worked well,

with the SPL providing virtually all the rear information while the spots augmented the frontal space and depth.

Panning individual microphones into a convincing surround image was of course nothing new to Sound Arts, who had previously experimented with elaborate layouts of discrete microphones of various kinds. The very first such experiment, in December 1998. was on a recording of a piano concerto using a remarkably simple setup with only eight microphones in total. In a sense this was the surround equivalent of a semi-purist stereo recording that might have been done with a Decca tree plus some close miking of the piano. Here five Neumann KM143s were used in a roughly pentagonal layout above the audience and orchestra; the rears were about 8m apart and 10m up, several rows back over the audience and facing backwards, while the three fronts were 3m above the stage and spaced by 2m or 3m. The detail of the piano was handled by three further 143s in a small tree arrangement. The result was a very vivid impression of the venue's acoustic, with excellent rendition of the localisation of the orchestral instruments and good clarity on the solo piano.

This basic idea was expanded on for a further session six months later, where two variants on a larger rig were tried. Here an out-front ORTF-style pair of Schoeps cardioids formed the basis of the front, augmented by a 6m-spaced pair of B&K omnis and multiple spot microphones. The back was handled by two rear-facing KM143s in line with the ORTF pair, and two Neumann KM130 omnis 5m or 6m further back. Two versions of this rig were used in the course of the recording: one as it stood with simple mixing, and one with delays added to all the closer microphones according to their carefully measured distances from the main pair. This technique of attempting to time-align the various sources is of course common practice in stereo, but in this case it actually seemed to work against the intended result. Both versions were very impressive, and both enveloped the listener in a persuasive surround image; but the one without the delays seemed more open, more natural, more clearly localised and generally more satisfying. In fact the very symptom the delays should have cured —muddying and smearing of the image —seemed to be worsened by their addition.

Sound Arts has been through many permutations of the idea of natural surround recordings. from simple specialised arrays to complex ad-hoc layouts, and two lessons appear to emerge. One is that experiment is often rewarded by surprising results; the other is that there is no 'right' way to do it —different methods have different strengths and different weaknesses. Both reinforce the idea that the final decision as to how to approach each recording is a matter of taste, judgement, knowledge and experience. Bit like stereo then. ■

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product placement...

The most critical part of any audio production has always been the final mix and in today's world that means being able to work with surround sound. Until recently this type of mixing power was available only to a select few. But not any longer...

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The
EQ
GUIDE

Dividing its time between remedial and creative duties, the outboard equaliser is an essential studio tool. **Derek Johnson** & **Debbie Poyser** audition the options

EQUALISATION has become a recording and mastering mainstay and the popularity of outboard units has only increased with the rise of digital recording and digital consoles. Studio models available range from the cost-effective and functional to the esoteric, where it's probably fair to say that if you have to ask the price you can't afford it. Design approaches vary equally widely. We can only provide a speedy tour, and we've had to exclude combination devices.

The United Kingdom has an illustrious history in EQ, and British companies offer a variety of equalisers at different price points and with different approaches. The idiosyncratic Joemeek proffers the VC5 Meequalizer (at £254, UK exVAT), a dual-channel device aiming to offer a classic sound with simple facilities—shelving treble and bass bands, and a swept mid band. Cut or boost of 16–18dB is available, but there's no bandwidth control. Another model in a niche that spans the project and professional studio is the 5013 (£399) from TL Audio. This dual-channel, 4-band parametric uses hybrid tube/solid-state circuitry and offers overlapping bands with up to 15dB of cut or boost in each band and Q variable between 0.5 and 5. Drawmer's take on tube equalisation is the £995 1961 dual-channel parametric, which augments its bass, low-mid, high-mid and treble bands with adjustable high-pass and low-pass filters. Each main band has six switchable, overlapping frequency choices, with Q variable between 0.3 and 3, and up to 18dB of cut or boost available. Klark Teknik is a venerable British company who made its name with graphic EQs, but are also famous for flexible, affordable and long-lived parametrics. The single-channel 405 (£595) is a 5-band device, with all bands fully parametric and covering a 20Hz–20kHz frequency range. Bandwidth is variable between 0.08 and 1.2, and cut-boost of -25dB to +15dB is available. It also features high- and low-pass filters. The DN410 (£999) is a dual-channel version of the 405.

Two Oram contenders are the HD-EQ2 (£3,534) and the limited-edition HD-35 (£4,276). Electronically similar (though the EQ2 has a sculpted front panel and more precise calibration), both are dual-channel devices offering six parametric bands plus HF and LF filters. Also available from Oram is the Octa EQ (£1,850), providing eight channels of 4-band parametric EQ. From Focusrite come the Red 2 (£2,195) dual-channel 4-band model (with a transformer-coupled design offering two parametric mid bands, shelving HF and LF bands, and high-pass and low-pass filters) and the dual-channel Blue 315 Isomorphic Mastering Equaliser (£3,995). The latter is a 4-band device with stepped rotary switches and the same circuitry and components as the legendary Focusrite ISA 110. Yet another opportunity to buy British is offered by Trident-MTA, with the iX2 EQ module. Though this 16-channel EQ, featuring four overlapping frequency bands per channel, is part of the Intermix modular mixing system, it could fit into any mixing situation.

A newer name to analogue signal processing is Cambridge-based digital convertor manufacturer Prism Sound, who teamed up with equipment designer and producer-engineer Leif Masses to create the Maselec Master Series MEA-2 4-band mastering equaliser (£3,950). This 2-channel device has stepped controls for precise settings, and four bands grouped as two overlapping pairs, each pair being dedicated to low or high frequencies. Each frequency control offers 21 settings, making a total of 84 choices per channel. Gain in the range of ±8dB is adjustable in 10 steps, and the bandwidth control for each band selects between five preset curves and a shelving option.

Across the pond there's once more a wealth of choice, including parametric EQs from the man who introduced the idea to the world. George Massenburg Labs markets two parametrics: the reference-standard, hand-built 8200 (£3,195), a dual-channel, 5-band model, and the 9500 program equaliser (£6,295) with detented controls. Both have discrete transistor circuitry, GML's servo stabilisation to correct DC offsets, top-quality passive components, and a transformerless, DC-coupled design. Another successful American export is Manley Labs, the company which has access to the passive EQ circuitry featured in the legendary Pultec EQs. The valve-equipped 3-band Pultec EQP-1A is available in single (£1,300) or dual-channel versions, and as mastering models featuring detented controls. Manley even offers an all-tube, dedicated mid-frequency EQ (£1,060) said to be ideal for mid-heavy instruments such as guitars. The latest EQ is the Massive Passive (£2,995), a stereo, 4-band mastering processor with additional high-pass and low-pass filters.

US designers certainly seem to favour valves: tubes are used in equalisers ranging from the Aphex 109 (£340), a cost-effective parametric featuring 'Tubessence' valve circuitry and dual 2-band or mono 4-band operation, right up to EQs costing £2,000 plus. The EQP-200B (£1,995) from Summit is an example of the latter type. It's a 2-channel, 4-band passive parametric programme equaliser utilising tubes as well as solid-state electronics, a transformerless signal path, and an unusual configuration. This comprises an HF shelf attenuator, a mid-high band offering boost control at eight frequencies, an LF band with boost and cut (available simultaneously) at five frequencies, and an LF shelving filter. The same company's EQF-100 (£2,995), is a single-channel, full-range passive device, with four overlapping EQ bands followed by a valve amp stage. Millennia Media, >

< too, uses valves and solid-state electronics, in a rather unusual way. Its £1,950 NSEQ2 dual-channel, 4-band parametric EQ offers switchable valve and solid-state amplifier stages in one unit, for an instant choice of sound character, plus an impressively wide frequency response.

Over in the tubeless camp is California-based Avalon Designs, who offer two 4-band, dual-mono parametrics based around class-A amplifiers, a DC-coupled, transformerless design, and passive-active filter topologies.

The 2055 (£2,635) has passive lower and upper bands with shelving or peak-dip characteristics, and two fully variable active mid bands. The substantial 2077 (£8,615) is recommended for mastering, and to that end it has switched controls. Even Summit, previously known for valve designs, is branching out, with the new Element 78 EQ-200 (£3,495). Summit used the talents of Rupert Neve in the design of the discrete Class A transformer-coupled circuitry of this solid-state 4-band parametric, and added MIDI, 25 memories, and support for Pro Tools with the 'Extension 78' software control panel.

Many ways to spend your Euros are offered by companies such a SPL, Z-Systems and Behringer. SPL's Qure parametric EQ (£999) features a mysterious 'Qure' control which appears to be derived from some of their Vitalizer

enhancement technology. The dual-channel, 3-band Qure also features variable-frequency high-pass and low-pass filters, and a valve stage.

There are not many digital equalisers in the world, yet Z-Systems, makes two. The Z-Q6 mastering EQ (£6,750) can operate at up to 96kHz with 24-bit resolution, and provides six channels of 4-band EQ—ideal for surround mixing —or two channels of 18-band EQ. The £2,495 Z-Q1 is aimed at mastering engineers, providing four parametric and two shelving bands, and can run in stereo or dual-channel mono. This 24-bit device is designed to behave like an analogue EQ, offering real-time adjustment without pops, clicks or digital artefacts.

Behringer addresses the lower-cost end of the market with the sub-£100 solid-state Ultra-Q Pro PEQ2200, but climb the price ladder somewhat with the valve-enhanced Tube Ultra-Q T1951 (£560). The PEQ2200 offers five fully parametric bands plus tunable high-pass and low-pass filters. The four overlapping bands of the stereo T1951 are fully parametric, though the high and low bands can be used as shelving filters.

A quick trip north of the border brings us to Denmark, home of Tube Tech, whose valve processors share a retro-style, functional appearance and large precision rotary controls. The ME1B single-channel parametric offers three passive EQ bands, with five selectable

centre frequencies in the LF band, 11 in the mid band, and five in the HF band. Also on the Tube-Tech roster are the PE1C 3-band passive mono program EQ, and the EQ1A 3-band single-channel EQ. ■

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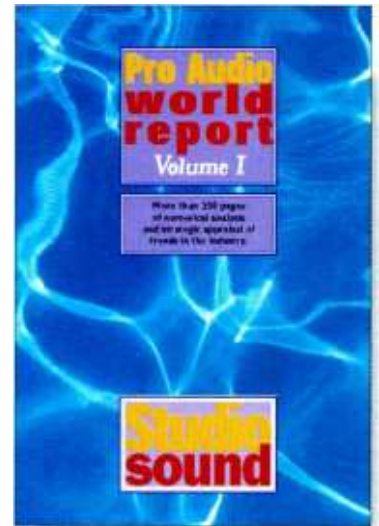
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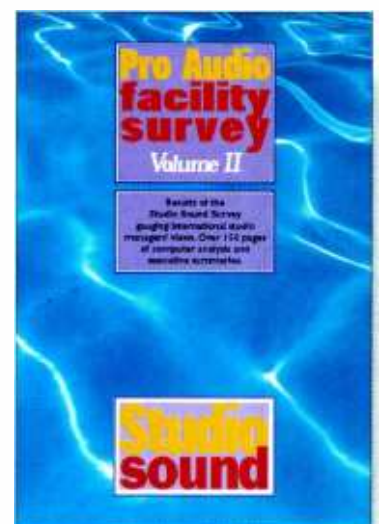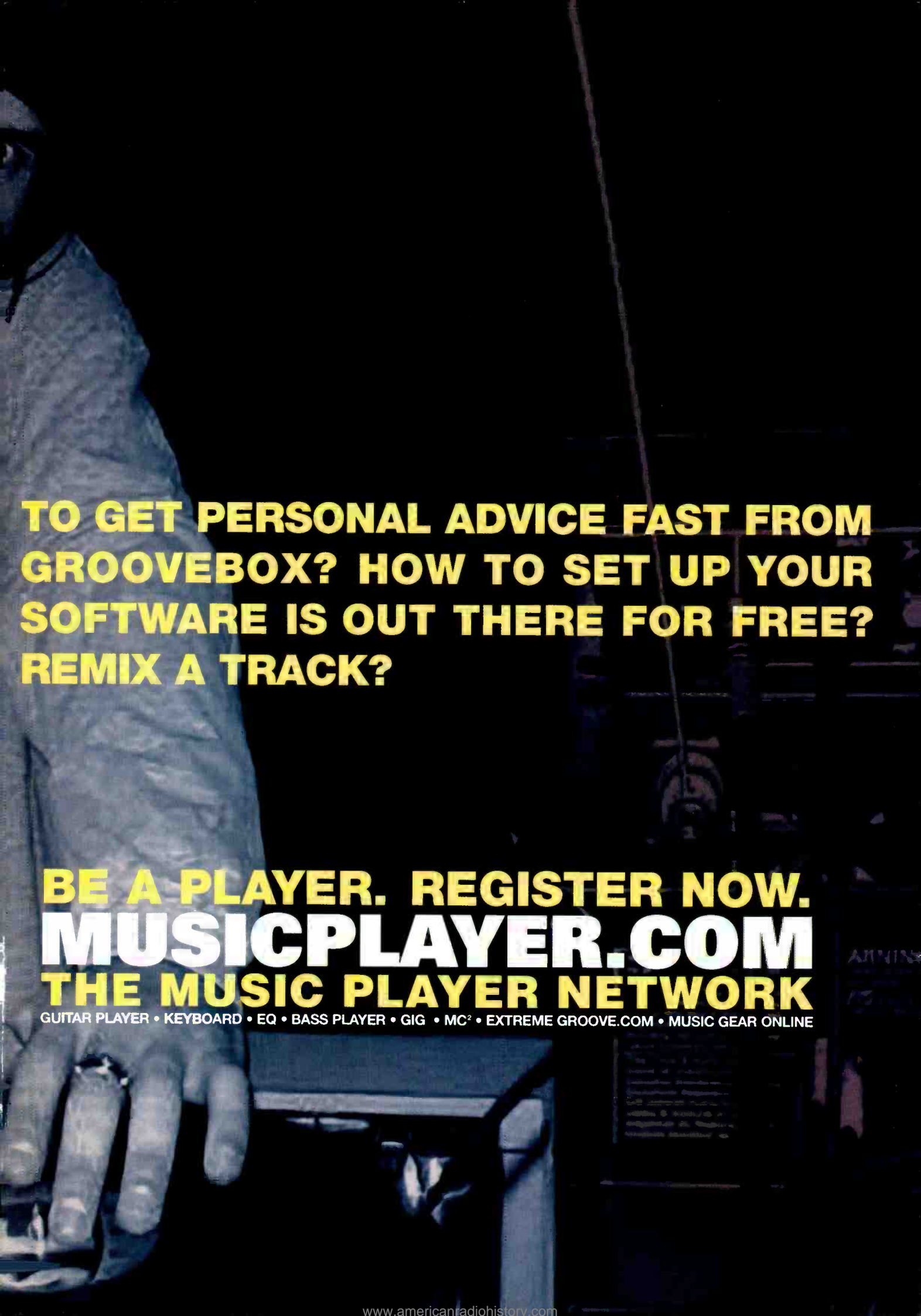
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Radio Days

Increased computing power has taken broadcast automation systems into the 21st century. **Kevin Hilton** explores the options and opportunities

IN THE EARLY DAYS of computing, mechanical computers took up huge amounts of carefully air-conditioned space. Today, more powerful devices sit quietly in the corner of many living rooms. In the same way, time-controlled, mechanically driven radio automation systems were housed in special equipment rooms, away from the main studios. Their modern, server-based counterparts are also consigned to separate racks, but they take up considerably less space and could comfortably run the entire output of an station if necessary.

With the evolution of computer processors and soundcards, the radio station in a broom cupboard is now a reality. As is commonplace today, the Internet is adding a new dimension: systems developed for that medium have found their way into the mainstream and have forced the prices down, bringing automation to users who would have previously considered it a luxury. In a market that ranges from expensive, fully functioned, complete solutions to software packages for under £100, radio automation is a classic example of users getting exactly what they pay for.

During last year's Sound Equipment Broadcasting Show (SBES) in Birmingham, UK, Jutel introduced a component version of its On Air-applications package. This can now be used for playing out schedules from the ENPS desktop, aside from its established jingle facilities. Included in this is the Cartwall, plus AutoPlayer for automatic play-out of playlists, the Quick Edit audio editor,

Matrix Controllers and the new DAB/WWW package, featuring the PAD server.

DAVID, at one time part of the Harman Group, is now owned by fellow German company Management Data Software Engineering. At the end of last year this software house completed a 2-year project to design and install a full digital production framework for all seven channels of the Hessischen Rundfunk network. Management Data has also been involved in initial pilot projects for the digitalisation of the BBC World Service's Russian language channel and is collaborating with US company b-i-t-s Broadcasting Technology Services. This year it expanded the scope of its automation activities by merging with Omni-Bus Systems, a British specialist in integrated TV studio control.

As much as manufacturers may want customers to use their system and their system only, end-users have become more sophisticated and shown that they do not necessarily want to be tied to one platform. Plug-in technology has made it easier for operators to pick and choose the applications or components they want. A classic example is the RCS Selector music play-list package, which has become almost the generic term for such systems. Many stations will use this in conjunction with other automation systems, preferring not to use RCS's own automation package or the scheduler of their chosen system.

A company banking heavily on plug-ins is Mediatron. Working on familiar drag-and-drop lines, Mediatron's plug-ins work under either Windows NT 4.0 or Windows 2000. Current plug-ins include: HotControl, a hotkey and short-cut play-back module that replaces cart machines; X-FadeEdit for creating cross-fades and segueways; VoiceTrack, a module for recording and triggering linking announcements; and the Digital Radio control module, offering the potential of text and graphics on

domestic receivers.

Most of these can be used as stand-alone applications and do not have to be used in conjunction with mediatron's full automation systems, the AirControl NT and AirEdit NT. AirControl NT is a full studio software package for the digitalisation and automation of radio stations and is designed to run programming on a 24-hour basis. Music, jingles, news, commercials and other sound elements are sourced in CD-quality from RAID5 hard-disk arrays, running at 32-bit on the Windows NT 4.0 platform.

All manufacturers of radio automation systems have built their reputation on one element, as with RCS and Selector, or have become the main supplier in either their home country or a specific part of the radio market. While aimed largely at the high-end, large scale sector, Netia Digital Audio and Dalet Digital Media Systems, coincidentally both French, are arguably two of the biggest names in automated radio.

Netia has consolidated its position in the last year by becoming part of the Belgian EVS Broadcast Equipment group and taking over another French operation, Audio Follow. The former move strengthens Netia in the TV market, as it, unusually for a radio automation company, also works in the vision field. Netia and Audio Follow had been business partners since 1993; a formal merger was regarded as prudent, as both systems work on the NT platform. Audio Follow's products have now been absorbed into Netia's Radio Assist range.

Netia has its own set of broadcasting modules, marketed under the Odyssey umbrella name. These incorporate Audio Follow's 'Air' prefix and include the Air-DDO play-out system for network centres; the Air-PlayList for themed stations, which can be used in automated or manual modes; the Air-HotKey; Air-CartStack; and Air-Jingle. Users of Netia equipment include Radio France, BBC World Service and the Australian Broadcasting Corporation.

Dalet has an equally long list of users, among them the BBC, CNN, Radio Switzerland International and BFBS. Central to the company's philosophy is the Dalet 5.1, which is able to acquire audio from a variety of sources, including satellites, CDs, digital files, and live recordings. All content is stored in a common database and assigned titles for correct identification.

One system that developed from collaboration and licensing is the Sony BMS. Swedish software house BSS was looking for a hardware platform, which was supplied by the multinational in 1998. Latest developments for BMS include a new SQL database, based on ODBC (Open DataBase Connectivity) for Windows and a new edit function.

ODBC acts as a translation device, which allows different data sources to communicate via SQL. The BMS editor

has been able to open a range of file formats (MP3, MPEG Layer II, AVI) for some time, using ActiveShow technology built into Windows. But this was only enabled it to 'write' (or mixdown) completed edits to a PCM WAV or Broadcast Wave file. This has now been expanded, giving the ability to use any file format supported in the Windows ACM (Audio Compression Manager).

For production purposes, the system offers the Mix Editor and Surfer applications for editing and preparing material. Dalet also has an optional music scheduler, while the whole system can work in live or fully automated modes. For the future, Dalet has included a multimedia content database and a variety of optional broadcast modules. Webcasting is covered by the On-Air Now! function, enabling broadcasters to offer e-commerce and content data (for example, song titles, album covers) that complement the on-air programming.

A big name, yet a relative newcomer to radio automation is Fairlight On Air Systems, formed when the DAW manufacturer acquired Vamos Media Solutions, based in the Netherlands, and fellow Australian company Ogenic On Air Systems. Central to its range is CoStar, based around hard disk RAID storage, semi-online CD jukebox arrays or offline CD-ROM/DLT tape backups linked to a self-tuning database engine. As a modular system, users have a choice of various components: Quick Recorder, Multi Channel Recorder, Looping Recorder, Single Track Editor for journalists and editors, CoSTAR Multitrack Editor for specialised usage, News Wire third-party systems interface, CoSTAR internal wire reader and the CoSTAR NewsCollector. A recent major installation was at Radio New Zealand's new Broadcasting House complex in Auckland.

Windows NT has become the operating system of choice for an increasing number of audio applications, so radio automation is shifting towards this platform as more new releases are purely software packages that work on any PC infrastructure. Although it can operate with servers running Novell or Unix, Studer's DigiMedia uses NT 4.0 as its main platform and, like a great many automation systems, employs Digigram professional audio soundboards, including the PCX9, PCX11, PCX11+, PCX80, PCX20 and PCXPocket.

Another convert to NT is Enco Systems, an American manufacturer that is apparently making a new push into Europe. It's main contender is the DADpro32 digital audio delivery system, which will also run on Windows 98. The system can be used for live assist, full automation, on-air presentation, production and inventory management.

The array of radio automation software on the market is almost bewildering. There are those that are well known in most national markets, but there are

plenty more that have carved out a niche on a local basis. Computer Concept Corporation is one of the longer serving developers, recognised for the DCS. This has evolved into Maestro 2.0, which uses apt-x, MPEG, or linear audio formats and can integrate with a user's preferred music scheduler.

On Air Digital USA has introduced version two of its Ultimate Digital Studio (UDS), which is designed to work with live presenters, but can also run as a completely automated device. At the core of the UDS II is the RS-HD (Radio Suite Hard Drive system), a Linux-based digital audio server. This stores all events on hard drive or combines with external CD changers.

A venerable name in broadcast automation is Smart Broadcast Systems. Its radio system is the Smartcaster, a hardware rack and software combination that employs MPEG or Gen2000 files controlled by Windows 98, 95, NT or DOS. The software is currently in its second generation; a play-list scheduler, Digital Programme Director, is included in the package.

Both stand-alone and integrated products are produced by Zenon Media of Germany. Stand-alones include the Air Check hard-disk logger and Auto Recorder; integrated units are the Send 32 scheduler and the Audio-Cast complete one-channel broadcast system. VCS has a family of automation products under the dira (digital radio) banner, including the OACTRL On Air Controller and the On Air jingle machine, cart player and scheduler.

Two of the best known British manufacturers for small to medium radio stations are Barrcode, which first came to note for the Brian editor, and ProCass, developer of the WAVcart. For wider automation Barrcode offers the BCSCS (Barrcode Storecasting System) and ProCass the SoundBox, an MPEG compression-based system that can run small stations on a 24-hour basis. Software house P Squared produces the Myriad V2.0 play-out and automation package. This comprises three components: Myriadmin, Myriad Live and Q-NXT. A fourth element, Voice Custom, is also available. Another significant seller in the UK is Radiomation, which has a strong grip on its native Republic of Ireland market, being used by the country's independent news provider to feed most commercial radio stations.

A great many software packages currently available for automation either began life aimed at Internet radio or are also intended for other music sequencing applications, like live DJing and music scheduling in clubs and bars. Web Jockey by HGYS caused a stir at last year's SBES, where it was demonstrated by UK distributor Alice Soundtech. Retailing for around £70, Version 2 is due later in the year, retailing at approximately £600, and will offer more

features for conventional radio.

Web Jockey is not alone. In no particular order, and taking a very deep breath, also available are: Radio 2000 digital jukebox and Internet radio system; Raduga V3.0 low-cost automation; Keogh Software's RMS WaveCast 3.0 for automated play-out of digital audio files; Supersonic software driven audio rack; Virtual DJ, using Microsoft Direct Xtech; a trio of systems from Greece—AK Radio Suite, Hotradio and Jazler; BasS (Broadcast Automated Support Systems) Windows and Mac-based server and play-out tools; Discosoft MP3; DRS 2006; Etre Windows-based software that can interface with 'leading components'; LAN International's Sound Manager, with a range of components including OnAir and Cartwall; OZZ 2000, offering three modules for studio operations, planning departments and systems engineers; Pristine Systems' Rapid Fire hard disk studio system and MusicPlus scheduler; Pro Studio Radio Host integrated software, including the Heavy Rotation scheduler; Tiesseci Broadcast Systems' ts2000, a networked or stand-alone system running on NT or Windows 2000; Win Antenna from France; Italy's Winjay; the Spanish XSystem; SBZ AIRMIX sound automation PRO and AIRMIX sound automation DJ; and Radio Edition's Megamix, equipped with Soundsoft MP3 TAG ID file labelling technology.

Unsurprisingly, some manufacturers are dismissive of the cheaper software solutions. Those involved with these packages admit they are limited in certain respects, but say it is a matter of what the user wants to do. In a perceived digital world, radio is neither as digit-based nor as automated as might be thought. Large-scale radio stations, particularly the cash-rich groups, can afford the high-end systems in a big way. Many smaller stations either use low-cost software automation or still have not progressed that far. Techno-dreams must be tempered with the revelation that, apparently, 80% of American radio stations still run off cart.

Lower cost technology is becoming critical to radio automation. As off-the-shelf soundcards fall in price and rise in quality, there has been a move away from Digigram. MPEG and apt-X compression could see a similar migration to other formats, although compression in general could be a less common part of this technology.

This is due to a shift towards linear audio. Manufacturers are already addressing the issue, with RCS's Master Control NTV14.3 supporting the format. Barrcode will introduce linear versions of existing products at SBES in November. The radio market has always been an idiosyncratic one and it seems that its future will once ahead be mapped out on a wet day in Birmingham. ∎

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Circo Beat's control room   Fingers' DVD studio

Aires
and
Graces

Buenos Aires is more than the Tango, with two major recording facilities illustrating the future of the most European of Latin America cities reports **Dan Daley**

BUENOS AIRES IS QUITE PROUD of its European heritage. That's immediately evident in the architecture of the sprawling city's downtown section, where sleek stores sell even sleeker Euro couture and furniture designs. Where much of the rest of South America remains an adobe and glass-and-steel hybrid of structural design, part Castilian, part Indian, part Bauhaus, BA, as the locals call it, combines classical post-Renaissance, post-colonial and postproduction harmoniously amid post-modern and twisty European lanes feeding into wide boulevards, including Avenue 9 de Julio, at 120m the widest avenue in the world. In short, BA takes some chances, but knows whence it came.

That sense of history seems to give it a good sense of where it's going, as well. Two facilities in BA illustrate how Argentina and possibly the rest of Latin America will handle the future. One facility is resting its hopes on DVD, using the recording business foundation it was once based on as a support mechanism for its launch into the world of digital media formats. The other is your straight-ahead rock 'n' roll studio that is undergoing the increasingly common shift from private domain of a rock luminary to making a living in the day-to-day world of for-hire facilities.

Fingers Arts Group is rooted in the recording studio business, one started modestly enough by owner Sergio Molho back in 1986 out of a small space with a Soundcraft recording console and a lot of MIDI gear. However, Molho, a husky, gregarious BA native whose ear

is glued to a cellphone that never seems to stop buzzing, found ways to follow the money in a business that was rapidly becoming the purview of everyone and anyone who could afford the increasingly affordable new generation of pro audio equipment. After MIDI had become a commodity, he moved into video editing and audio postproduction, shifting the company's focus from music recording (rock, not Latino; another way that BA stays more Euro than South American) to corporate clients such as advertising agencies. That, in turn, led to the addition of animation, graphics and digital audio editing, and later led to the creation of a joint venture with US media facility designer John Storyk's Walters-Storyk Design Group in 1994, after Molho had moved his expanded studio business into new digs in BA's Villa Crespo neighbourhood.

'The studio business is changing all over the world,' Molho says. 'And even though this is Argentina, we're not immune from the forces of evolution in the business. We've had to adapt the facility to changes in the market and changes in technology. Yeah, we follow the money. But that's what business is about. When a lot of people can make records on their own equipment, then music studios have to move into areas where people can't do things at home.'

That mind-set has now led Molho to turn yet another technological corner and establish what is South America's first independent DVD authoring facility last year. While the new format has barely a footprint in Argentina and not

much more of a presence in all of the continent (only Brazil and Mexico thus far). Molho is betting that DVD will become a major force in his region, although not in quite the way it's doing so in Europe and North America.

'Argentina is what I would call Hollywood-dependent,' observes Molho from behind his desk, where he sits framed by a Kandinsky print from the Guggenheim Museum. 'I suppose it's understandable to some degree—with 35m people, Argentina is a relatively small market for movies. But I knew that movies weren't the way to introduce an authoring business to Argentina.'

What Fingers Arts Group—the name refers to the musical adroitness of Molho's cousin, a musician who lives in the States—did instead was to leverage its already large base of corporate clients, which it had developed during its transition to postproduction, and introduce them to DVD as the interactive successor to VHS for corporate promotional videos. Although the awareness of DVD in the advertising and other corporate ranks was virtually nil in 1999, within six months Fingers Arts had created two DVD presentations, one for a real estate developer, the other for the BA branch of an Italian advertising firm. During that time, it also did the DVD authoring for the first domestically produced Argentine film to go to DVD, *Mia Alma* (*Dear Alma*).

However, Molho doesn't believe that the advent of DVD is going to be any sort of deliverance for the Argentine film industry or the audio post business which feeds it. The Argentine movie industry puts out only about a dozen new titles a year. Molho isn't kidding himself that Argentina is destined to replace Hollywood, Bombay or Hong Kong any time soon. Rather, he says, the theatrical titles that will start fill- **>**

< ing Argentina's DVD pipelines will come from catalogue films, bringing domestic output from 12 to about 40 locally produced new films per year within a few years, still a very small number.

Fingers Art Group is fitted like a jig-saw puzzle into a 2-storey space above retail shops on the street level below. At the top of a steep staircase is a warren of small yet well-apportioned studios and offices containing the DVD authoring suite and a multimedia presentation room back to back, with Molho's and other offices on the other side of the reception area. The second floor, accessed via a spiral staircase, holds the audio studio that was the original business at this location, fitted with a Soundcraft analogue console and Panasonic digital mixer, as well as video editing and multimedia graphics suites using Adobe Photoshop and Quark Xpress, and a small cafe and lounge.

The DVD suite is based around a Sonic Solutions DVD Creator system residing on a Macintosh computer. This is fed by an array of media devices, including a Tascam DA-30 DAT deck and a DA-88 digital 8-track for audio tracks, and a Sony Beta SP video deck. Two Pioneer 244 DVD decks serves as playback for quality control and presentations in the adjacent presentation room. Both suites are fitted with Speakercraft audio monitors which, like the Denon power amps that run them, are decidedly quotidian as far as audiophile pieces go. 'We decided that we wanted to be able to do the authoring and monitor the results under the same kind of playback environments that most of the work would be seen in,' explains Augustin Ignacio Alvarez, Fingers' DVD manager and sole authoring engineer. (At the moment; Molho expects to add two more by year's end in order to have the facility running a full three shifts. Currently, Fingers has a total of six technicians working here at various media disciplines.) The rooms sound good and tight, however, helped

Fingers' video room

by the placement of shallow RPG diffusers on the rear walls, creating an articulate and accurate surround audio playback environment. And while most of the audio components of DVD productions are created out of house and brought in by the client, Fingers uses a Digidesign Pro Tools system to create its multichannel mixes. Despite its compactness, the entire facility has each media suite linked via Ethernet.

The authoring projects have been relatively simple and straightforward thus far. *Mia Alma*, for instance, has a middling degree of interactivity, including interviews with the producer and the cast, as well as cinema and broadcast trailers, a music video clip, and Portuguese and English subtitling. The subtitles are translated by another company in BA and delivered to Fingers as text files with time code embedded. The film's audio came in on the DA-88, with tracks 1-6 holding the surround mix and stereo on track 7 and 8. The dialogue tracks are mixed in with the film score; there are no separate M&E tracks, which is standard operating procedure in Hollywood to facilitate foreign versioning dubs. 'They don't do separate music and effects tracks in Argentina because the

films don't get out of the country much,' Alvarez explains. 'That would change the entire culture of film audio post-production in Argentina. But that may happen at some point as the film industry here grows.'

And while Molho continues to pursue music recording to some extent, he believes that the region's future will be based on a wider range of media, and he is acting accordingly. 'We think corporate work is going to be the big growth area for audio,' he states.

Rock music has been the anarchistic muse to a few generations of social, political and cultural hellions, and it did not pass Argentina by.

ESTUDIOS CIRCO BEAT started as the private facility for Argentine rock legend Fito Paez, but in many ways it is the logical conclusion of how rock gained its foothold in Argentina, a narrative that has all the elements of made-for-TV movie. A war movie, actually, dating back to the 1982 Falklands War, better known here as *La Guerra Malvinas*, the name by which Argentina knows the disputed islands.

'Rock 'n' roll had always been persecuted by the military governments of

Argentina while I lived here,' recalls Alan Colbert, a native of Buenos Aires who left for the States in the late eighties, working in the music business in New York and Los Angeles and now back in BA as the label liaison between Warner Music and Paez. 'If you went to a rock concert here in the seventies, you could easily be arrested. It wasn't a place that was very hospitable to rock 'n' roll.'

Up until that point, says Colbert, the local rock music industry was small and based largely upon garage bands playing cover songs of western artists —especially British ones, since Buenos Aires is culturally, architecturally and emotionally a European city, and the UK was the centre of much of rock's development in the seventies and eighties. And though Spanish is the language of Argentina, those bands sang American and British rock songs in English, trying to get every note and inflection perfect, like any adolescent garage band of the time anywhere in the world.

The way things were then, with rock essentially on a short leash and locked in the cocoon of English that prevented it from infusing itself into Argentine culture, was just where the ruling military junta wanted it, to the extent that it tolerated it at all. Besides, they had other things to think about, including a faltering economy that was producing quadruple-digit inflation on an annual basis, and political unrest that was inducing increasing violent—and increasingly lethal—police crackdowns on dissidents.

It was a time in Argentine history when the studio that Colbert and I were sitting in and having this conversation would not have been possible. Fito Paez opened Estudios Circo Beat in December, 1997, and named it after one of his best-selling albums. A protege of Argentine rock legend Charly [sic] Garcia, with whom he played keyboards and guitar before stepping out to his own multi-platinum career, Paez originally planned it as purely a personal recording facility. However, as has been the case

in many instances, building a complex and expensive personal facility leads to it being opened eventually for outside rentals, which is the road that Circo Beat is now taking, after a redesign of the large space, located in the industrial Paternal barrio of Buenos Aries, by the Walter-Storyk Design Group.

The main recording space is 12m x 50m with a 7m ceiling rising to a small skylight and surrounded by several iso booths. Motorised acoustical cloud panels can change the room's acoustical properties dramatically: from a decay of 0.4ms to over 0.8ms. The control room, which houses a 56-input SSL G+ console with Massenberg automation and a vintage 10:2 Neve sidecar mixer, is heavily diffused, and loaded with Genelec 1038 monitors.

The design was limited to some degree by the building itself, which is long and somewhat narrow—it was originally meant to be a television production studio before Paez purchased it. Storyk recalls that the first move was to get the control room lowered. 'I needed the height,' he explains. 'Then I angled the control room a little in the space to give a new axis to the entire facility, which also provided acoustic high frequency ray reflection benefits with the angled surfaces in the studio. We also knew that at some point the

facility might start to be rented out, and that was one of the reasons for getting the control room sunken and made a little bit larger than we might have done if it was just a project studio.

The big issue in the studio was variable acoustics. Fito needs live big band recording potential—he uses a great number of strings and acoustic instruments on some of his recordings as well as typical rock setup. We were able to get a 100% RT60 (reverberation time) swing from 0.4s to 0.8s with a system of moving wall panels—hinged variable surfaces as well as motorised ceiling clouds. And all of the rooms are isolated using a combination of floated and split slabs. We actually made our own isolation product in Argentina from materials that we can get locally.'

Power is always a concern in a Third World environment, and Circo Beat was no exception. 'Power was a screw up, that's for sure,' Storyk sighs. 'In general, Third World power situations are not that difficult; most of the problems were and are political—getting it installed and so on. Cleaning up the power was not that big a problem. It's getting the lines in properly that's the main issue in these situations.'

In addition to work by regional recording artists, including Los Fabulosos Cadillacs and Ilya Kuryakin & >

Circo Beat's control room

the Valderamas. Circo Beat has also propelled Paez's film-scoring career, and a projector and drop-down screen are part of the design in the main studio. The facility is gaining an international reputation, helped by the fact that Paez' most recent record was produced by Phil Ramone and engineered by Frank Filipetti. Upstairs, the fittings for a prince of the musical realm are evident in a huge, manor house-like dining-conference room, next to a gourmet kitchen.

But back to the narrative, because it was in that dining room that the conversation with Colbert took place, and where, surrounded by racks of vintage Argentinean, Chilean, Californian and French wines, the story of how rock 'n' roll came out of the closet in Argentina reached its conclusion.

'What changed it all was the Malvinas War,' says Colbert. 'The government was in trouble, economically and politically, and war was a good way to deflect criticism and attention from those issues, as it often is for many authoritarian governments, and a way to rally the people around a central issue. So as part of the war effort, the government decreed that in the future, all broadcasting had to be in Spanish —no English. Especially not English, because the war was with England. And that turned out to be the best thing that ever happened to rock 'n' roll here, because all of a sudden, radio and television needed a lot of Spanish music to fill up the airwaves. That meant that the musicians of Argentina—including the rock musicians—had to start generating new music. The bands began to write their own songs instead of covering English and American rock songs. And now that they could do that in Spanish and have an outlet for them, that really pushed the development of a rock culture here forward. And what had once been a business that stayed in small clubs was suddenly filling soccer stadiums. That was the beginnings of Argentine rock.'

And Argentine rock music never stopped to look back. After the military government fell in the wake of the Malvinas War disaster, Argentine rock artists found a wider reception for their music throughout South America, and several of them, such as Los Fabulosos Cadillacs, are making inroads into the North American music market. Rock's revolutionary roots run deeply in Buenos Aires. The walls of the buildings in Paez' working-class neighbourhood are covered with political graffiti. And, up in his equipment storage loft, he keeps a portrait of Cuban revolutionary leader Che Guevera hung on the wall, overlooking the guitars and drums and amplifiers. Like some saintly icon, Che's visage confers a blessing on the equipment, reminding it that has a larger mission in the world than fraternity parties and Saturday night soirees. ■

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US: The musical bomb

Without some major reorganisation, the destruction of the established music market is at hand writes **Dan Daley**

IMAGINE THAT the atomic bomb had been invented before the aeroplane. (Trust me, I know where I'm going with this.) You have a weapon of unimaginable destruction, but are incapable of delivering it. Along come the Wright brothers, who provide you with just the delivery system to whomp whomever it is you feel needs whomping. You needn't buy a bad Hollywood script to arrive at a similar scenario in the music business at the moment. Just flick on your computer and watch the music business as we know it crumble around you.

If you look at the world from the perspective of the Recording Industry Association of America, you'd view MP3 as a slew of atomic bombs, going off randomly, chipping away at the fortress of your business. Then add Napster to the equation. If the RIAA went crazy over MP3, it went apeshit over the delivery mechanism that is Napster, a software engine developed by a bright but bored 19-year-old college dropout that enables people to find exactly the music file they want, retrieve it from one of the estimated 10m hard drives of other Napster users and store it on their own hard drives, from which it can be retrieved by any of those users. This is precision bombing at its finest.

There is a twist to the MP3-Napster story that affects the professional audio world, and in doing so will perhaps hasten the erosion of the music industry: MP3 has caught on in a huge way in the commercial music business. Audio post facilities across the States are using MP3 to send draughts and finals of commercials, promo bumpers, spots and other commercial audio bits to advertising agencies for reviews and approvals.

But it also seems that that same record industry which so reviles MP3 for ripping music off the Internet is now also using the format in a similar fashion to the audio post business: they are taking in new releases, formatting them for MP3, and using the Net to send them to such destinations as radio programming directors to get feedback on which cuts should be the first single off the album, or sending them to prospective directors to bid on doing the music video, or distributing within the company preproduction tracks before an album is even finished. These files might have been conveniently labelled for internal use, such as 'Green Day Nu Release' or 'Sting Single' or 'Oasis demo'.

Enter Napster. Go up there sometime and enter words like 'demo' or 'advance' and see what it pulls up. Could be anything from a garage band in Des Moines to the next Springsteen track, months before it was due. And once Napsters get their hands on it, it begins to make the rounds of 10m hard drives.

According to a report in the Los Angeles Times, one major-label president reported that one of his artists recently made verbal threats to a company employee that she would be blamed if the act's album turned up on Napster before the scheduled release date. Other artists, reportedly Radiohead among them, are demanding that no advance copies of albums be sent to journalists or others in the promotional chain. The labels are reportedly scrambling to devise and implement security measures to avoid leaks.

It would be funny, if it weren't also so amazing. What's being created out of this unstoppable use of new distribution technologies is a definite disincentive to make music for commercial gain. At least, in the way that it has been done for the last 100 years or so. In the same way that personal recording technology created the horizon for the studio business the way it was since the forties, MP3 and Napster and Gnutella and other such software are putting nails in the coffin of the music business as we know it.

No longer will large conglomerated media entities have easy strangleholds on the way music is created, marketed and distributed. Combine that with the proposed sale of Universal's music holdings, acquired and owned by Seagram's, to a French corporation, and Sony's long-

Europe: Listening and hearing

Listening tests carried out in London on the Verance watermarking system have done nothing to quell engineers' fears writes **Barry Fox**

AFTER A CONCERTED awareness campaign conducted over the Internet by UK-based engineer Tony Faulkner, the SDMI and Verance finally gave the first demonstration of their analogue watermarking system outside the US in early July. Around 30 engineers and a few journalists were invited to Sony's studios in Whitfield Street and given the chance to do comparative listening tests on four pieces of music, with and without watermarking.

Paul Jessop, technical director of the IFPI represented the Secure Digital Music Initiative; David Liebowitz, Chairman of Verance Corporation, was there too. But no-one was present from 4C Entity, the consortium of IBM, Intel, Matsushita-Panasonic and Toshiba which ran the US tests in the summer of 1999 on behalf of the SDMI, and on whose say-so DVD-Audio and probably also SA-CD will be watermarked. Panasonic has already launched DVD-A players in the US, over the 4th July weekend.

'The testing for DVD-A is done and complete' Leibowitz insists. 'From October all DVD-A players must have Verance circuitry built-in.'

The 4C tests which led to this decision were conducted last year in US studios, on 50 'golden ears' who had been chosen by the five major record labels. So far only Denny Purcell of GeorgeTown Mastering, Dave Smith (Sony Music) and Mark Wilder (Sony Music) have been prepared to identify themselves. Although DVD-A provides a sampling rate of 192kHz, 4C's test material was coded only at 96kHz/24-bit standard.

After 4C chose Verance as the winning system, 4C created a licensing corporation called LMI which will now administer the licences and collect huge royalties on behalf of Verance.

So why no 4C in London? 'They were busy with other commitments, they send their apologies and have left me to carry the flag', said Liebowitz. Yes, there were good personal reasons why one member of 4C could not be present. But why no-one? In the words of Pretty Woman, Bad Mistake.

Astonishingly, the testing process depended on a Compaq laptop PC in the listening room with a hard drive that whirred louder than quiet passages of the music and any subtle ambience. The drive could not be switched off during the tests. How can anyone who is serious about testing, organise a test with such an obvious distraction?

Faulkner flew back from Malaysia to be in London for the tests. 'I'm 49 years old, I've got a cold and I had jet lag', he said after participating the day after flying in. 'I scored 75%, correctly identifying the watermarking in six out of eight cases. This system is not transparent, and anyone who says so just isn't telling the truth.'

James Mallinson is a Grammy-winning producer for all the major record labels and independents. 'The tests were so badly handled that they are totally invalid. I don't know whether it's because they don't know about sound quality or were trying to pull the wool over our eyes.'

'The first musical piece, Debussy piano was an old analogue tape with hiss and very constrained frequency range. Halfway through I refused to continue because I said there was no point in the test. The Petrushka recording was some of the worst recorded sound I have ever heard; analogue hiss and you could hear the azimuth flexing, as you do from old tapes.

'It was quite impossible to judge a system with that kind of material and DVD audio and DSD are all about high quality.

'Then they played some New World

rumoured desires to unload their entertainment properties, and you are watching the unstated reaction of the corporate mega-entities to this situation: divestiture of entertainment holdings due to a diminishing ability to wring revenues out of them. Not bad for an afternoon's work by a disaffected teenager.

It goes further. Once the broadband pipeline becomes more widespread, this scenario begins to apply itself to films and television. Don't take my word for it: listen to Jack Valenti, president and CEO of the Motion Picture Association of America (MPAA), as quoted in the Miami Herald: 'If Napster can facilitate and encourage the distribution of pirated sound recordings, then what's to stop it from doing the same to movies, software, books, magazines, newspapers, television, photographs or video games?' The answer, Jack, is nothing.

It would be a cheap shot to say that the high-end marketers of music are playing a charade—decrying MP3 and Napster while simultaneously using it as an internal marketing tool, and crying again when it turns around and bites them in the shorts. But the two points I will leave you with are that it was audio which established the pattern for what will likely become the most radical restructuring of the modern entertainment business since its inception, and that the field it was played out upon was the one delineated by American marketing concepts and technology. Keep an eye on how this plays out over here. It's how it's going to go everywhere else.

music which was appallingly badly recorded, with no low end, no high end above around 10kHz, and heavily compressed. It was of magnificently low quality. Then they played some dreadfully recorded pop. It was a joke, a complete joke. I was really quite cross because I thought they had wasted my time. These tests were totally invalid.'

World-renowned hi-fi reviewer Martin Colloms had to invite himself. Says Colloms: 'I withdrew from the tests. The quality of the material was appalling. It was sub-Walkman standard. They were using very old analogue material, the Petrushka dates back to 1962. The piano had a dynamic range of only around 40dB. It was awful music, lifeless and anaemic. There was a mess of hiss. The material had been copied from CD-R onto computer hard disk, and then processed through a 96-24 convertor, so it didn't start as 96-24. The room acoustic was bad, with metalwork ringing. The PC hard drive was whirring in my ear and a transformer was mechanically humming. I told Sony it was an embarrassment to their company and I couldn't avoid wondering whether they had chosen material that was bad to conceal what the system was doing.'

Adds Faulkner: 'If the system is audible now with a 2-bit copy management payload, how will it sound with a 72-bit full identifier payload?'

Reframing the airwaves

From being a specific forum on broadcasting issues **Kevin Hilton** signals a widening discussion of delivery matters

THIS IS THE END, the end of broadcast as we have all known it since the mid-nineties. In the last year, much has been discussed as to how 'traditional' broadcasting practices and companies will change through the influence of new technologies and techniques. There will undoubtedly be a long-term evolution, but in the immediate future, long-established broadcast ways will form the basis of all activities.

What is changing is this column. From being a specific broadcast comment forum to allow me to take relevant broadcasting issues and make irrelevant, obscure remarks about them, it will become a wider-ranging discussion of delivery matters. About which I will make irrelevant, obscure observations. It is reassuring to know that while things change, they don't change that much.

Some of the issues that will now be discussed regularly in this space have already been examined in past outings, whether in detail or just in passing. Digital technology has blurred the edges between many areas: where there was once a clear divide between production, postproduction, distribution and transmission, the core use of digits has reduced the entire process to merely the movement of data.

Different front-end interfaces remind us that there are differences between the various stages, that they are different skills; but, ultimately, it is possible to imagine a workstation that combines all the functions, connected to a black box that will then direct where the finished product goes. Production and postproduction techniques are highly developed and while they will continue to grow in sophistication, delivery of the programme or creative content is the area where most attention is currently directed.

With a plethora of outlets—digital terrestrial, cable and satellite television, digital radio, broadband communications and the Internet (including mobile 'phones, personal organisers, WAP devices et al) —delivery is arguably as important as the creation of what is carried. This is a massive turnaround in thinking. At one time, not very long ago, transmission was the broadcast journalism equivalent of the black spot. We all knew that it was absolutely necessary—otherwise the creative stuff wouldn't go anywhere—but that did not make up for the overall dullness of the subject. It did not help that transmitters looked like filing cabinets. The only commission more mind-numbing was one concerning routeing and switching.

But things change and now those involved in transmission, routeing, switching and matrixing—the technological meek—are very likely to inherit the earth. It would have been hard to imagine that NTL, once merely the engineering

division of a now almost forgotten regulatory body, would become a brand name splashed over newspapers, TV and radio as 'the complete communications company'. But it has. Likewise, national telcos—France Telecom is a prime example —are central to the distribution of sporting events like Euro 2000 and are poised to be crucial to webcasting.

Delivery is important; it is even, to use an eighties word, 'sexy'. Naturally, without good content, delivery is just a means to an end. Hence NTL taking shares in digital radio multiplexes and others buying into production companies. In the light of all this, it would appear that the BBC's decision to privatise and sell off its transmission networks was hopelessly shortsighted. But if one considers that conventional terrestrial transmission is not the future, then the Corporation did not completely strand itself—particularly as it retained the landline infrastructure between the transmitters.

In the spirit of free trade, the BBC made its studios and facilities business a wholly owned subsidiary—BBC Resources—with the implication that it would eventually privatise it. At the beginning of July, new director-general Greg Dyke announced the findings of his review of Resources. Privatisation is out and more efficient and cheaper working is in, with the loss of 200 jobs. Some elements of Resources will be returned to BBC Production, while a new subsidiary, BBC Technology, will be formed, subject to government approval.

'We want to change parts of Resources and make it more profitable,' Dyke told a strangely under attended press conference. Making a profit out of public service broadcasting was central to the approach of Dyke's predecessor, John Birt, and while the new incumbent has not rejected that, he is set to almost completely undo the philosophy that became known as Birtism. 'We are trying to get rid of the dafter parts of the internal market without losing the good parts,' Dyke stated bluntly.

One of the good parts is obviously delivery. BBC Technology will centre on 'BBC technological and distribution expertise', with the ultimate aim being to generate additional income to spend on programmes and services. (The subtext is getting enough cash to win back sport from the commercial sector.) The new subsidiary will comprise approximately 1,300 personnel, experienced in broadcast, telephony, IT, on-line technology and communications. The BBC projects that Technology will generate between £150m to £200m over the next six years.

Such projections emphasise the importance of delivery and show how much broadcasting is set to change. Delivery being used to underwrite programme making—who would have thought it?

Ampex
ATR100 Series

In the second part
of his eulogy on the
venerable ATR mastering
machine, **Tony Arnold**
gives more pointers
and fixes

ALWAYS MAKE SURE that your ATR
has tape lifter sleeves fitted over the
tape lifter arms, otherwise a master
can be severely damaged due to wear from
the original arms. Machines are often updated
from ½-inch to ¼-inch without changing tape
lifter sleeves or fitting them at all, and users
then wonder why they have a severe scratch
down the middle of a ¼-inch tape or the edges
are curled on ¼-inch and ½-inch tapes.

Another further warning accompanies
another tip. If you wish to run your capstan
for adjustment or repair then you will have to
place it in an extender card that is usually
housed in slot No. 5. All you have to do is to
switch the ATR off and remove PWA No. 9.
This is the Reel Servo Card and you will now
have all transport functions without having to
load the tape. That's the tip, now for the warn-
ing. Ampex provided a clip on the extender
card to prevent whichever PWA you are ser-
vicing from falling out, but now that the cap-
stan PWA and the servo are removed from
the machine it has been known for the cap-
stan to be fitted accidentally into the wrong
slot with a huge amount of resulting damage.
To stop this happening we make up dummy
cards from plastic and fit these into the empty
slots not being serviced.

I will repeat again that you should never
remove or refit cards while the machine is
switched on and this includes the headblock.
When removing or refitting the ATR head-
block always turn the head release screw
clockwise as this will prevent the allen screw
from coming loose from the lower T section.

Users often ask for low-profile TO3 tran-
sistors to replace Q5, Q6, Q8, Q9 and Q10 on
the main heat sync chassis and the reason
these were used was to prevent the collector
from grounding through the on-off switch
arm. We fix this by fitting thick sleeving over
the arm to isolate this problem.

The 4-speed dual EQ Padnet assembly is
designed as a plug-in compatible accessory
for the ATR100. It may interchanged with the
standard Padnet assembly and requires no
modification to the standard ATR100 or to the
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The 4-speed Padnet can be configured in
two modes by simple switch selections. In
conjunction with the existing jumper options
for master bias operation on the audio

control PWA, it provides an extremely
flexible equalising and biasing system.

In the 4-speed configuration each of the
four speeds (30ips, 15ips, 7.5ips and 3.75ips)
is individually biased, level controlled and
equalised. The equalisation at each speed may
be set to any required standard.

In the 2-speed configuration, as with the
standard Padnet, any two speeds of the four
available may be selected. However, each of
the two speeds selected has two separate
equalisers and level controls available. The
selection of which equaliser is in use is made
by the EQII switch on the audio control PWA.
In addition the audio control PWA also pro-
vides individual master bias adjustments for
each speed-equaliser.

With this Padnet it is thus possible to con-
figure the full 4-speed capability of the ATR100,
or, having selected the 2-speed dual equaliser
configuration, to align the two equalisers, lev-
els, and biasing for two major equalisation stan-
dards (that is NAB and IEC) or to align the
system to use either of two types of tape at
each speed with no restric-
tions on operating levels,
equalisation standards, or
biasing. In addition to inde-
pendent reproduce and
record gain presets, and the
normal repro HF, LF, record
HF, front panel equalisations,
there is a second record
equaliser, 'Record shelf'
which, in conjunction with
the normal record HF
equaliser, permits flatter over-
all response to be obtained
with a wider range of tape
and biasing conditions.

You can change the over-
lays on the appropriate PWA
to suit 4-speed or 2-speed
and if you have the wrong
overlay showing you should
find the correct one behind
the existing one. You simply
remove the small screw, and
slide the two overlays out
vertically to change. On the
Audio Control PWA, you will
also have to remove the
printed board as the bias
switch will prevent removal.

Some confusion does
exist on how to set the DIL
switches on the 4-speed
Padnet. For the 2-speed
setup (Fig.1), mount the
appropriate front overlay
and ensure the audio con-
trol PWA is set as required.
Set either of the four posi-

tion slide switches (S2 or S3) to the '45' posi-
tion. For each speed in turn, determine if
record low frequency boost is required (NAB).
Switch S1.1 through S1.4 to select this, for
30ips through to 3.75ips respectively. Any
switch in the on position provides no LF boost
(IEC) as required by AES 30ips or EIC-CCIR
15ips and 7.5ips. For NAB 15 7.5ips and IEC-
NAB 3.75ips standards, set the appropriate
switch to off and this will provide the stan-
dard 3180 low frequency record boost.

For the 4-speed setup (Fig.1), mount the
appropriate front overlay on the Padnet and
the ensure audio control PWA is jumpered cor-
rectly for the speed pair desired. Set S2 to the
position corresponding to the higher speed
of the required speed pair.

Set S3 to the position corresponding to the
lower speed of the required speed pair. If S2
and S3 are set to the same speed, the illegal
speed indicator next to the transport speed
select switch will illuminate indicating
improper operation. Set S1.1 through S1/4 to
the correct positions determined by the equal-
isation standards to be used at each speed,
and-or the EQ I or II selection.

S1/1 and S1/2 are for EQ I, high-speed and
low-speed respectively and S1.3 and S1/4 are
for EQ II, high-speed and low-speed respec-
tively. As before, the on position for each
switch of S1 is for no LF record boost (AES
30ips, IEC-CCIR 15 7.5ips) and the off posi-
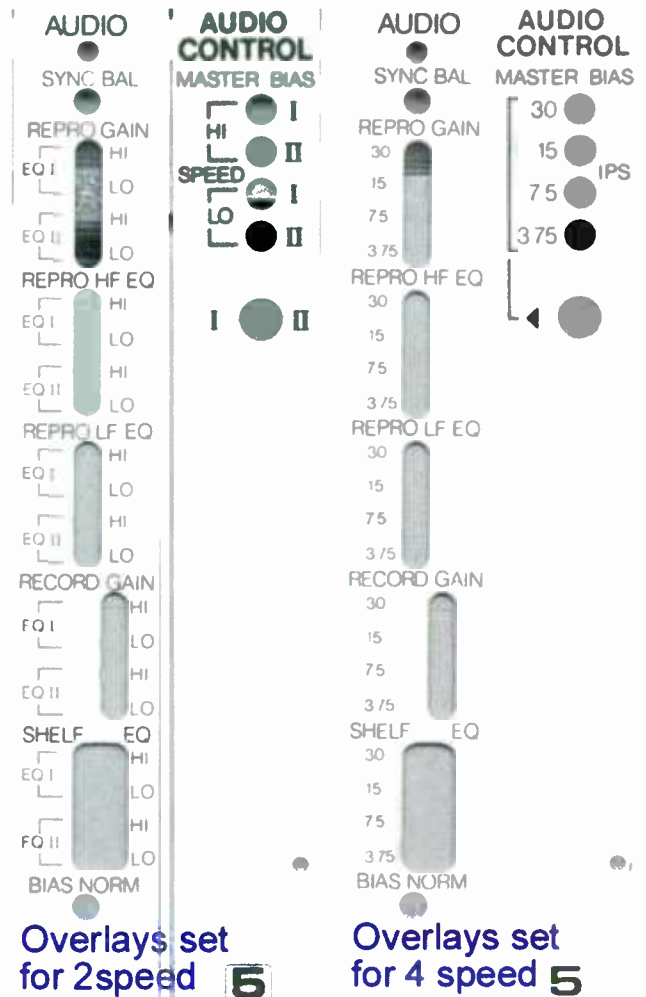tion provides 3180 LF record boost (NAB
15 7.5ips, and NAB-IEC 3.75ips).

**Overlays set
for 2speed** 5

Fig.1a

**Overlays set
for 4 speed** 5

Fig.1b

These 4-speed Padnets are still available from Courthouse Facilities or M Tupper (Tel: +44 1729 552428).

Many second-hand ATRs are purchased without a service manual and most machines were shipped to only take 10½-inch reels, new owners often wonder which type of ATR accommodates 14-inch reels. There is not space in this article to explain how to modify your machine to take the larger reel-size, but all the relevant information is in the service manual. Photo copies of service manuals are available from us and in time these will be available on CD-ROM.

For the conversion from ¼-inch to ½-inch mastering operation, apart from the cost of the heads the other expensive items are the upgraded tape tension guides. If you have the early type of guide then you will have to change the whole assembly, but if you have the later type just the extra length shaft and fixing screw need to be fitted to your existing Ceramic guides and this brings the cost down dramatically.

When removing ICs, often the print itself is damaged, and with the print being on both sides of the PWA it makes it difficult to see if you have made good contact with the upper print which will sit below the new IC. We have had great success by using wire wound type DIL Sockets and when using this type of DIL you are able raise the socket and solder in place to see the print around the legs of the socket.

On the early version of the control panel switches the press-buttons shed a powder that gets into the switches causing intermittent malfunction. Ampex later made a protection sheet to trap the dust and it's worth checking to see if you have this update and it leads to longer switch life.

This is the completion of my comments on the ATR 100 Series, next month I'll be giving tips on the MM1200 series. If you have any questions or problems please email me as I hope I can be of some help. ■

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A more in-depth edition of this feature with extra figures is available on our website

Transforms and Duality

Exploiting the duality of digital signal analysis gives us an essential insight into the working of many audio processes. **John Watkinson** explains transforms and prediction

A COMPLETE DESCRIPTION of a signal in the frequency domain consists of frequency and phase information, whereas a complete description in the time domain consists of a waveform. Transforms allow either description to be used. In general, any phenomenon in audio that can be explained in the frequency domain can also be explained in the time domain and vice versa. Indeed if it cannot there is something wrong with our understanding.

The duality of transforms provides an interesting insight into what is happening in common processes. In general the product of equivalent parameters on either side of a transform remains constant, so that if one increases, the other must fall. Fig.1 shows the example of a low-pass filter. As an ideal low-pass filter has a rectangular frequency response, the impulse response will be the transform of a rectangle which is a sinx/x curve. At Fig.1a is shown a filter with a wider bandwidth, which has a narrow impulse response. At Fig.1b is shown a filter of narrower bandwidth that has a wide impulse response. This is duality in action.

If we want to know the effect of a filter on a signal, we can use either the frequency or the time domains. In Fig.1c the output spectrum is obtained by multiplying the input spectrum by the response of the filter. If these responses are measured logarithmically in dB, the multiplication is achieved by adding. However, in the time domain, the effect of a filter on the waveform can be calculated by convolving the waveform with the impulse response of the filter as in Fig.1d. A good example of convolution is the tracking process of a vinyl disc player (anyone remember vinyl discs?) The output waveform of the pickup is the convolution of the shape of the stylus and the shape of the groove. As we might expect, narrowing the impulse response of the stylus, by making it elliptical instead of round, would extend the frequency response

and reduce the distortion.

Fig.2 shows some examples of duality. At Fig.2a to increase the width of the frequency response of a tape head, the gap has to be made smaller. At Fig.2b to improve the width of the directivity pattern of a loudspeaker or microphone, the diaphragm has to be made smaller. At Fig.2c the narrower a pulse gets, the broader its spectrum becomes.

Duality is like paranoia, the more paranoid you get, the more threats you perceive. Thus the more familiar you become with duality, the more examples come to mind. The compact disc player has a storage density that is a function of the spot size of the laser pickup. We can also apply transforms to light. Fourier optics holds that a lens is a filter to spatial frequencies. As detail in an image increases, the light goes further and further from the optical axis toward the perimeter of the lens. Thus the resolution is limited by the aperture. In order to increase the capacity of CD to make DVD, the aperture of the lens has to go up. This has the effect of increasing the spatial bandwidth of the lens, so by duality it can focus to a smaller spot on the disc. The intensity function of the read-out spot then convolves with the track geometry to give the replay waveform.

Duality also holds for sampled systems. A sampling process is periodic in the time domain. This results in a spectrum that is periodic in the frequency domain. If the time between the samples is reduced, the bandwidth of the system rises. Fig.3a shows that a continuous time signal has a continuous spectrum whereas at Fig.3b the frequency transform of a sampled signal is also discrete. In other words sampled signals can only be analysed into a finite number of frequencies. The more accurate the frequency analysis has to be, the more samples are needed in the block. Making the block longer reduces the ability to locate a transient in time. This is the Heisenberg inequality again. The Heisenberg inequality is the ultimate case of duality because when infinite accuracy is achieved in one domain, there is no accuracy at all in the other.

Block length switching is common in audio coding. For example in MPEG Layer 2 coding, usually the blocks contain 1152 samples, corresponding to a time period of 24ms. During transients, the block length drops to 384 samples, increasing the time resolution by a factor of three, but reducing the frequency resolution by the same factor.
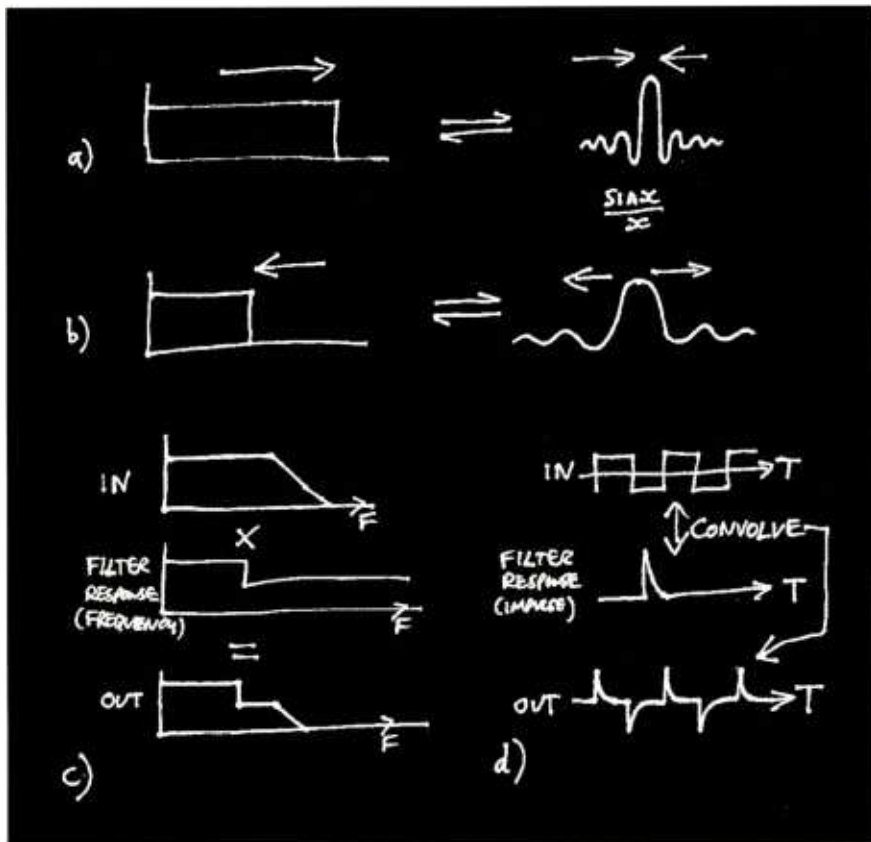

Fig.1a): Wide frequency response has narrower impulse response.
Fig.1b): narrow frequency response has wide impulse response.
Fig.1c): in frequency domain, input spectrum is multiplied by filter response to give output spectrum.
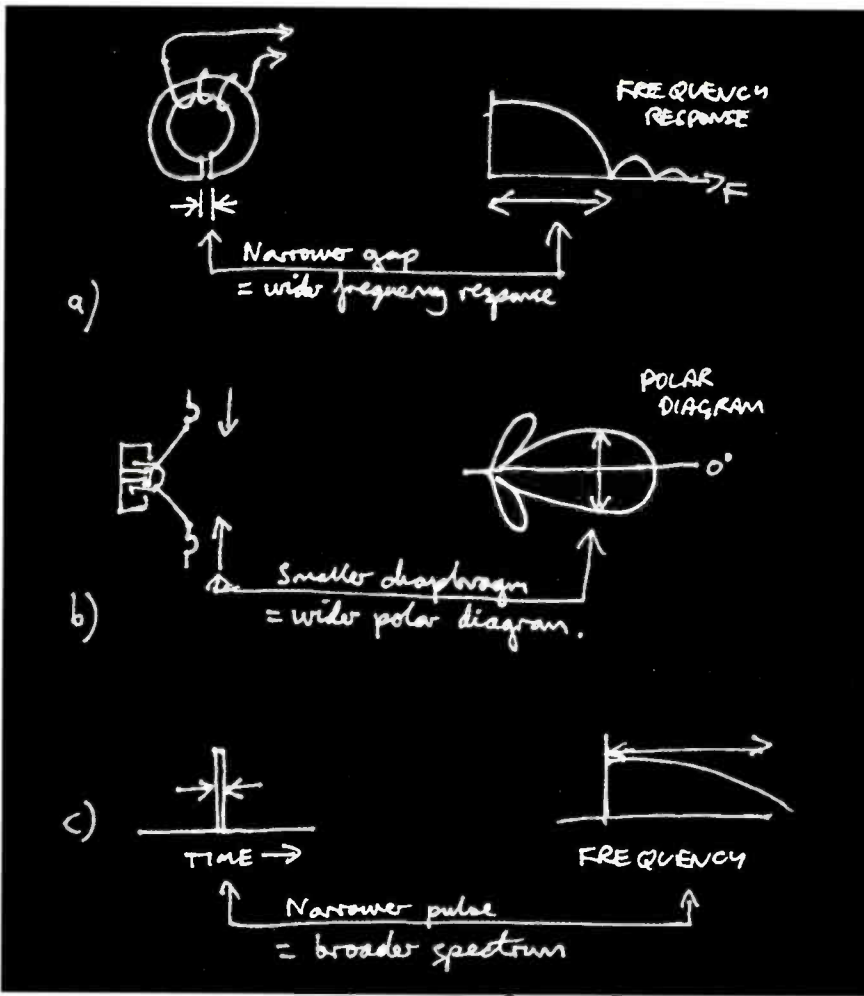Fig.1d): in time domain input waveform is convolved with filter impulse response.

Fig.2: Three examples of 'Less is more'

In compression, the transmitted bit rate can be reduced by the use of a number of tools. One of these is prediction. Fig.4a shows that in a time domain predictor the system uses knowledge of earlier sample values to try to anticipate what the value of the next sample will be. In the real world it will always be in error, and this is handled by subtracting the prediction from the actual value and sending the difference. The more accurate the prediction, the smaller will be the size of the error and the lower the bit rate can be. In the decoder an identical predictor is used, and the addition of the prediction error results in the original value once more.

Predictors can be simple or complex,
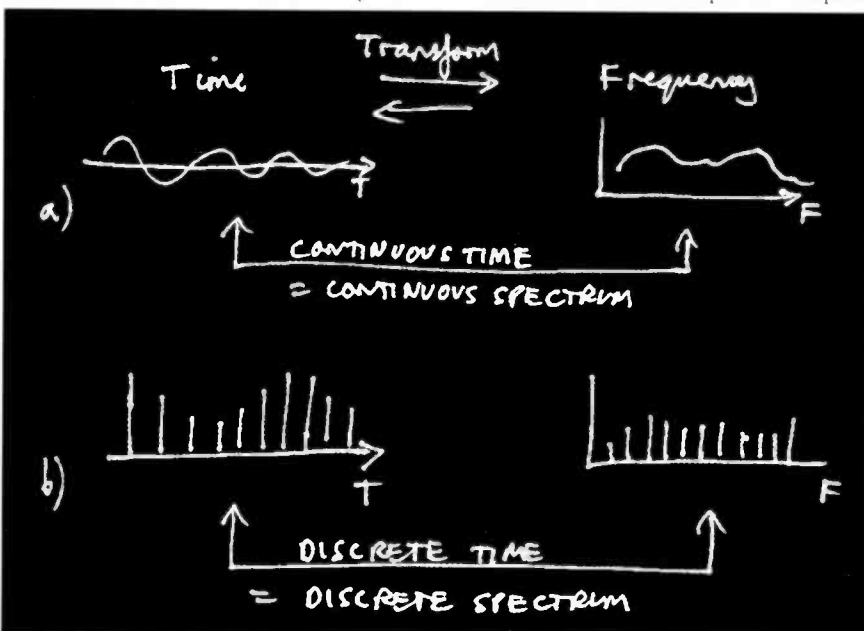
Fig.3: A continuous signal a) has a continuous spectrum whereas a sampled b) has a discrete spectrum
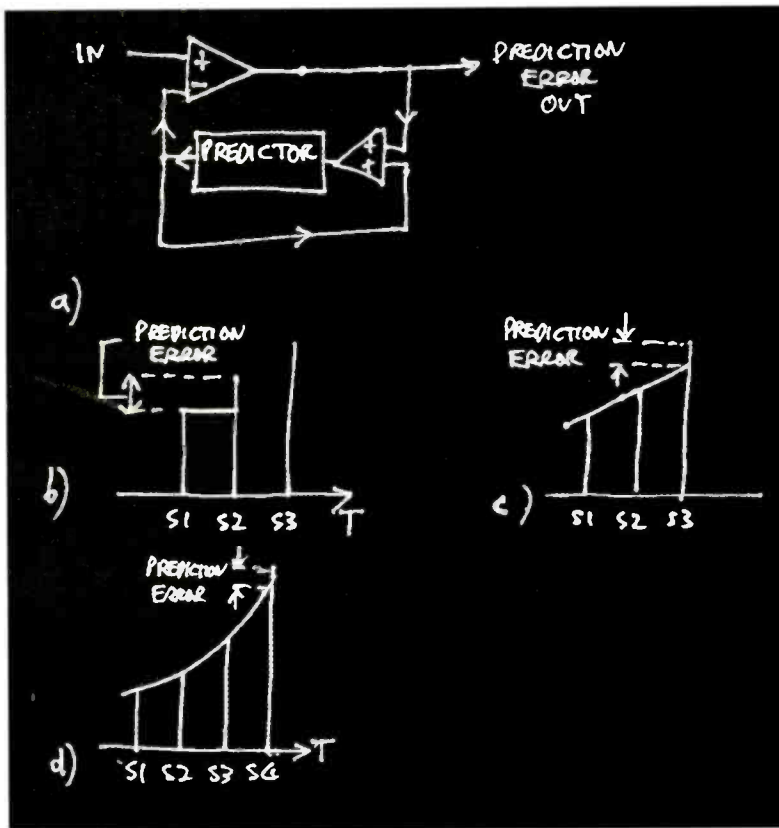
Fig.4: Time domain prediction. At a) the prediction error is sent to the decoder. Comparison with previous sample b) results in differential coding. Using two samples c) the slope can be used, whereas with three samples d) the curvature is included.
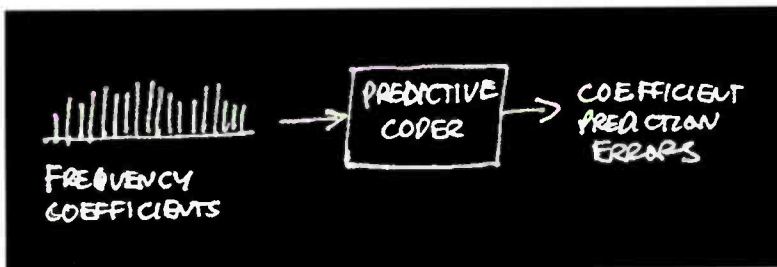
Fig.5: Predictive coding can also be used in the frequency domain where the coefficient's value is estimated from the ones before
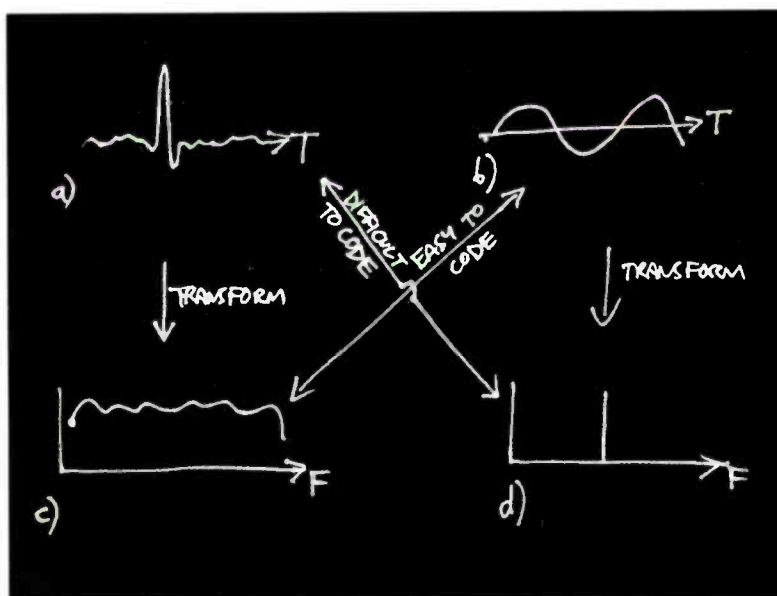
Fig.6: Transient at a) is not anticipated by predictive coder, whereas broad spectrum of transient at c) is easy to predict. Sine wave has opposite effect. Waveform is easy to predict b) but spectrum contains a spike d)

as long as the same predictor is present at both ends of the system. A stupid predictor might simply assume that the current sample has the same value as the previous one. This results in differential coding as in Fig.4b. A smarter coder might look at two previous samples to estimate the slope of the waveform as in Fig.4c. Using three previous samples allows the slope and the curvature of the waveform to be taken into consideration as in Fig.4d. Predictors of this kind work well on waveforms such as sine waves, but in the presence of transients the prediction fails. Essentially the predictor doesn't see a transient coming, because it doesn't follow from what went before.

Another type of prediction that can be used is in the frequency domain. Fig.5 shows that after a transform the result is a block of coefficients. By scanning the coefficients from one end of the block to the other, a predictor can be used to estimate the value of the current coefficient from the value of previous ones. Again any number of previous coefficients can be used to make the prediction more accurate. Predictors of this kind work best on broad or uniform spectra where many frequencies are present in the audio signal. Prediction in the frequency domain works poorly in the case of discrete tones because these make the spectrum spikey and the predictor fails to anticipate a discrete tone at one frequency because there is no energy in nearby frequencies.

Combining Figs.4 and 5 give another interesting example of transform duality shown in Fig.6. At Fig.6a in the time domain, a predictor fails to anticipate a transient because it is not a function of the earlier part of the waveform. At Fig.6b a time-domain predictor works well on a sine wave because it contains no transients and the slope and curvature change smoothly. If we transform into the frequency domain we obtain an opposite characteristic. Fig.6c shows that in the frequency domain, the transient of Fig.6a has a broad spectrum, consistent with Fig.2c. A predictor working in the frequency domain finds the spectrum of Fig.6c easy to handle because it is reasonably smooth. In the same way Fig.6d shows that the spectrum of the sine wave of Fig.6b is a spike or singularity. This is difficult for a spectral predictor.

Thus the duality can be seen in predictors. A transient waveform that is difficult for a temporal predictor is easy for a frequency domain predictor, and a stationary waveform in the time domain that is easy for a temporal predictor is difficult for a frequency domain predictor.

This all sounds like a lot of abstract theory but it has a practical application. In MPEG Advance Audio Coding (AAC) both temporal and frequency-domain prediction are used. The incoming audio is subject to a frequency transform, and the temporal predictor attempts to anticipate what coefficient values will be by using the values from earlier blocks. On stationary material this will work well. Additionally the frequency domain predictor operates within each transform block by trying to predict each coefficient from ones of lower frequency. This works well on transient material. The AAC coder is free to use either of these techniques and it may even try both to see which results in the most effective compression. AAC achieves a better subjective quality than Layers 1,2 or 3, which is no bad thing, and it does so by tangibly exploiting transform duality. ∎

Passing the buck

The mutterings of discontent over new audio distribution channels is becoming a roar. **Peter Filieu** ponders the position of the artist and producer

MP3, DOWNLOADING, streaming, broadband, security and business models are becoming an increasingly familiar part of our vocabulary. The impact of our brave, new digital future is often described in terms of enthusiasm, excitement and opportunity but tinged with caution over security. Often such rhetoric masks a combination of apprehension (not to say fear), a lack of clear understanding (not to say ignorance) and suspicion (not to say cynicism) resulting from the atmosphere of corporate uncertainty and an apparent lack of priority being given to the artists' and producers' desire and indeed, duty to ensure their fans have access to the very best in music at top audio quality.

Much of the talk is of increased choice, 'free' music, cutting out the (big bad) record companies in favour of (small bad) music distribution dotcoms and download directory services. The prospect of genuine ecommerce security is the straw to which we are all clinging but the speed with which new technologies arrive and are adopted casts a dark cloud over everyone's future. The SDMI standard seems to be the only game in town but there is still a measure of scepticism as to whether its implementation can stem the Napster terror. Much of the current confusion arises from the vast and disparate range of services and business models on offer. Sadly, many of the first players suffer from a 'chicken and egg' market-share dilemma —the need to acquire a critical mass of content before they can attract sufficient consumer attention to fulfil their promises to artists. This has led to there being little filtering of quality from dross. Currently, consumers are drowned in so much, usually substandard, choice that realistic profitable commercial volumes are hard to come by. This may change when the majors go on stream, but at the moment a new artist can expect to lose best tracks as free download samples and collecting relatively insignificant mail-order revenue only for the first couple of quarters.

Maybe we should agree to reduce the quality of all 'free' downloads and apply impregnable security to all high-quality distributions. Producers and artists are becoming increasingly concerned about how and at what stage security systems are applied. The old chestnut of inaudible watermarking still poses serious questions. Only recently have proprietary systems become commercially available and the very issue of competing regimes splitting up the total available catalogue can only add further problems for the consumer as well as causing nightmares for the producer. Listening tests have almost universally failed to satisfy the best expectations of audio professionals and the expression 'trade-off' is

being bandied around. Maybe artists and producers will be forced to sacrifice some quality in favour of business but the idea that the imposition of a variety of packages, each depending upon particular commercial alliances, as an intrinsically postproduction process is regarded by many discerning producers as untenable. In any event, the producers and the artists are not even in the loop.

Producer revenue streams are affected in parallel to artist revenues, but the situation for the producer is more critical. The question of performance revenues derived from 'related rights' is still in the balance for producers and remixers, but there is an issue of even greater importance affecting all performers. While secure downloads can be compared to sales of physical products,

> Producers and artists could find the quality of their recordings in jeopardy as a result of decisions made by company execs who in their enthusiasm to protect our business feel unable to trust those who make the music

different industry sectors have differing opinions on whether web-cast streaming is analogous to broadcast or whether as broadcasts they should benefit (or suffer) from related rights revenues. It seems the future of conventional broadcast is set to lie in web-casting. There are over 3,000 'stations' available all over the world, some rebroadcasting existing programmes and others providing new focused services. Web-casting organisations usually pay for music use, but not necessarily on a measured census basis even though on-line details of what is playing is available to the punter. Collection societies like PRS, ASCAP and BMI have historically relied upon measurement of usage to calculate distributions of licence income to composers and publishers. But the proliferation of users expected in the digital distribution environment threatens to expose an inability or possibly an unwillingness to afford the cost of the necessary data processing. The same applies to performance income to a greater extent. How PPL and PAMRA plan to handle web-casts is still up in the air. Even the question of whether web-casts qualify in the same way as conventional broadcast is a point at issue.

Another move in the US has the RIAA launching a royalty collection and distribution system of its own. Although at an early

stage of development, fears are already being voiced over whether the US record companies are the right people to handle what is estimated to be revenue in excess of US$400m from digital distribution sources and what or even whether performers will get any of it.

Key is metadata—the information about who played what, wrote what or owns what and how a recording was made—that feeds all ecommerce systems. Collecting and handling this information has always been sector specific and piecemeal—record companies, publishers and the MU for example only wanting data that suits their own needs. While there is broad acknowledgement of the need to develop a cost-effective and accurate metadata system, there is resistance to spending money on developing it on a cross-sector basis. As the studio producer is the one individual in the recording process who can provide accurate and at-source performance and asset protection data, facilitating the extra work should be a worthwhile investment. Sadly this seems not to be the case. Indeed, there have even been efforts to cynically transfer data collection responsibilities under 'take it or leave it' contractual terms.
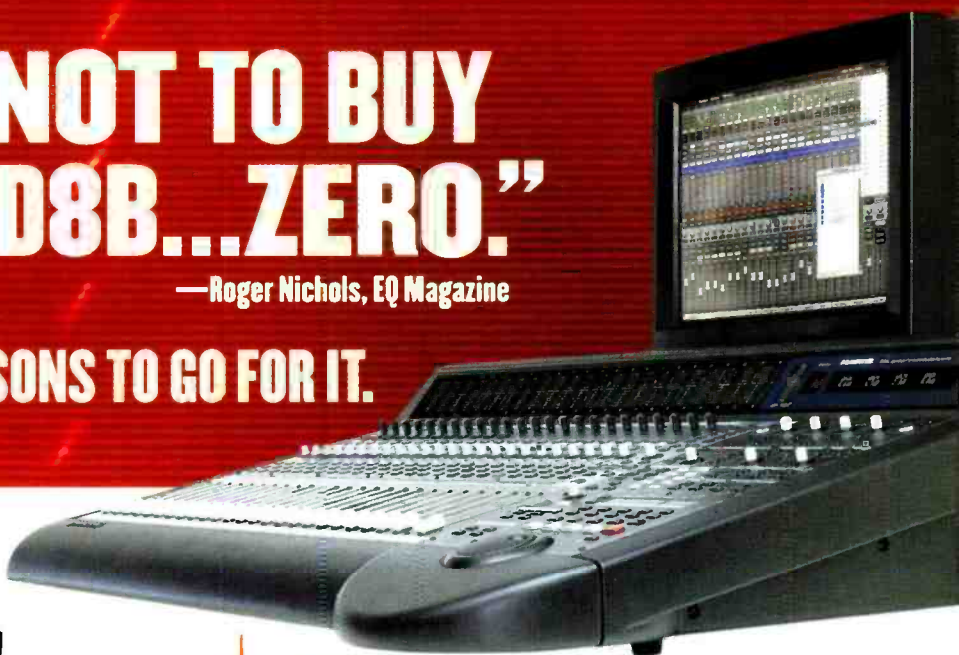
A true perspective of the current 'music on the Internet' situation would probably benefit from a regular reminder that, compared to, say, commercial flight, the Web, even including broadband, is much closer to Lindburg's 'Spirit of America' than to Concorde. There may be new business models on the horizon that will place the studio producer in an altogether different position but our traditional role as bridge between inspiration and product will remain pivotal no matter how music is distributed. However, both producers and artists could find the quality of their recordings in jeopardy as a result of decisions made by company execs who in their enthusiasm to protect our business feel unable to trust the contributions and experience of those who make the music at its core.

On balance, the marketing and career building added-value provided by experienced record companies (be they virtual or not) is likely to remain the most attractive option both for ambitious artists and discerning consumers. It remains to be seen whether the record company sector has the vision to include the studio producer community as full participants in future solutions. If they don't, the reduction in the cost of quality audio production and the convergence of the interests of producers and artists will encourage a vast increase in rights owning producers, management and production houses who licence their recordings to marketing and distribution entities that used to be called record companies. ■

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