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Stage Tec A–D
Mackie HR824
Helios EQ1
Wendt X2
dbx 386

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Studio sound *online edition*

Studio Sound May 2000

Have you heard the news?
When you next visit the *Studio Sound* web-site simply enter your email in the box and get our weekly news update

June 2000   Studio Sound

A question of outlook

A NUMBER OF INTERESTING POINTS are raised in our look at the prognosis of the DAW in this issue. While the devil's advocate is clearly being played on page 57 it is the changing nature of the environment in which the DAW operates that surfaces as the issue rather than a shortcoming of power and application.

In too many people's minds the cliched view of the DAW still remains a picture of an inclined and button-rich worksurface dominated by a green screen, and you could argue quite convincingly that the essential operational elements required to achieve DAW status have remained largely unchanged over the years. Bolt-ons and enhancements, such as larger and faster storage and processing which have led most significantly to increases in tracks, have only improved on the fundamental duties. Yet the stand-alone nature of these audio monoliths has slowly been eroded as you'll find representations tied with sequencing, as widely available and profession-indiscriminate desktop utilities, and increasingly buried within larger picture editing motherships. It is the last of these that could be regarded as the fulfilment of what some see as the intermediate solution of our state-of-the-art combined and intergrated DAW-digital mixer-nonlinear, video-playback package.

When you create a DVD you are dealing in a variety of different media simultaneously with none of the artificial craft and discipline divisions that exist at other stages of its production chain. Elsewhere there are broadcast programmes produced in video-edit suites without resort to a dedicated audio professional and you will find many Soundscape and Yamaha cards working well outside the realm of audio rooms.

Not very many years ago I ploughed a furrow on the impending impact of picture systems with audio capability on the sound operator's lot. Opinions were divided on whether such developments were a fuss about nothing or a threat. Economic pressures will cause the fuss that ensures that previously disparate processes will become combined. Whether this is a threat depends entirely on your outlook and whether you want to take advantage of it.

Zenon Schoepe, executive editor

Strings 'n' things

WHETHER DANCE MUSIC was the catalyst for the rise of the European project studio, or whether the project studio was an essential ingredient in the rise and rise of European dance music, is a moot point. Either way, the two are inextricably linked as one of the Nemeses of major recording studios. The damage done to the lower end of the commercial recording market by Tascam's first Portastudio is as nothing when compared to the sustained onslaught of successful records made in project rooms rather than bigger commercial facilities. But the swing back towards traditional recording venues seems to be gathering momentum.

It is easy to identify the elements that made dance a bedroom gift: drums sampled from vinyl or prepackaged in a drum machine, bass lines perfectly delivered by a sequencer, piano and strings from countless synthesiser modules packed with 'dance' patches... Only the topline vocal of the mainstream dance anthem promises anything traditional—maybe. But given dance's success and proliferation, the question is: how to imbue such a formulaic sound with renewed appeal? And the current answer seems to be a return to the old-fashioned acoustic sounds that the genre had initially so readily discarded.

Let's start with last year's UK chart success of Touch and Go's 'Would You...' —an otherwise unremarkable single lifted in part by the plaintive female plea 'Would you go to bed with me?', but mostly by a trumpet lick light-years beyond the capabilities of any synth or sampler. More recently, Moloko's 'The Time is Now' topped the UK charts—an unashamed dance track reliant on a token sequenced bass drum, acoustic piano and guitars, and lavish strings arranged by Audrey Riley. Since then, MJ Cole's charting 'Crazy Love' has sought to further the cause of UK garage music with a captivating samples string lick that could only have been more beguiling had it been delivered by a real string section.

Admittedly, most of Moloko's *Things to Make and Do* album (wasn't that an Angletraxx single from the seventies?) is unashamedly electronic, as is William Orbit's interpretation of Samuel Barber's 'Adagio for Strings', but the message seems clear—dance music is offering to make peace over studio practice. And we all know that there's no substitute for a good mic, a good preamp and the expertise to use them. Or, for that matter, for a talented violinist...

Tim Goodyer, editor

The universal favourite
even

VS-1880

- 18 track digital recording workstation with 288 virtual tracks
- 28 channel, fully automated digital mixer (18 tracks + 10 input channels)
- 12 gigabyte hard drive gives over 80 hours recording time in 24-bit MTP mode at 44.1khz
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- 24-bit A/D-D/A convertors ensure optimum sound quality
- CD mastering is now even faster, and image files can be stored directly on the hard drive
- Emagic Logic VS software [PC/Mac] available free to registered VS users [excluding VS-840 series]

VS-1880

If music is a universal language, then this is where you write the story. These are the blank pages - you provide the pen and the inspiration.

Once you're ready to record your music there are things you should look for which will help you achieve your vision: reliability, clarity, reputation, intuition, flexibility, power. Where do you start?

The Roland VS-1880 18-track hard-disk recorder catapults your music into the professional domain by delivering true 24-bit audio, via world-class effects processors, in a powerhouse package that has seen the V-studio concept sell over 150,000 units world-wide.

www.americanradiohistory.com

■ Israel's leading studios have recently ordered a total of four Studer 928 analogue consoles for use in applications including TV broadcast and re-mote recording. The first 928 has been ordered by a 56-input Globus Group GG of Israel and is the largest 928 in Israel. Jerusalem Capital Studios has ordered a 48-input 928 with modific-ations and customised functions speci-fic to JCS while United Studios of Israel has purchased two 36-input 928s for broadcast transmission—one desk will be installed in a remote truck, the sec-ond console will be installed in Studio 6 and be used for TV broadcasting.
**Studer, Switzerland. \
Net: www.studer.ch**
■ LA's Todd-AO post house has pur-chased Tascam 13 MMR-8s and 24 MMP-16s for use in its various facilities throughout Los Angeles. The facility, which has been posting top motion pictures and television series since 1952, now employs over 50 MM-series units some of which have seen recent action on *American Beauty*, *Gladiator*, *South Park*, and a DVD re-release of *The Exorcist*. Todd-AO is presently remastering many of Stanley Kubrick's films, in 7.1 or 5.1 formats. Meanwhile, LA's new Intersound's new audio and video post facility has install-ed an 88-input, 24-channel Euphonix System 5 for its Stage 1. The facility houses 7 editorial suites and three ADR-Foley stages, Avid Symphony and Media Composer on-line and Avid Media Composer off-line bays and specialises in sound mixing, sound design, foreign language dubbing and subtitling, MPEG encoding, on-line and off-line editing, plus various services for the feature film, television and ancillary markets. The console will be interfaced to digital sources, recorders and work-stations, including DA-88 MDMs, Pro Tools and Tascam MMR-8s which are housed in a central machine room.
**Todd-AO, US. Tel: +1 213 962 4000.
Intersound, US.
Net: www.intersound.com
Euphonix, US.
Net: www.euphonix.com
Tascam, US.
Net:www.tascam.com**
■ Austria's Studio Baumann has install-ed a D&R Octagon mixing console to compliment its Pro Tools 5.0 system, 32-track ADAT XT20 and Otari MX80 analogue multitrack. Located in the town of Steinakirchen, Baumann spe-cialises in the recording of acoustical instruments and has built up a solid reputation over the last 14 years. Recently Italian star Albano Carrisi recorded and mixed his latest album *Ancora In Volo* at the facility.
**Tonstudio Baumann, Austria.
Tel: +43 7488 76638.
D&R Electronica, The Netherlands.
Tel: +31 294 418014.**
■ UK-based Carlton Television's Abingdon Studios has taken delivery of a 36-channel Calrec S2 analogue console for use on live Central News bulletins and the early evening regional news. The S2 replaces a Sony MPX2000 and is to be installed by studio installation specialists, Ian P Kinloch.
**Carlton TV, UK.
Tel: +44 20 7722 8111.
Calrec Audio, UK.
Tel: +44 1422 842159.**
■ Chicago-based Radio Spirits has ordered four Spectral Design Audio-

Cube 3-II mastering and restoration systems for its specialist restoration and redistribution of 'old time radio broad-casts' Programme material consists primarily of broadcasts from the thirties, forties and fifties, that are restored by Radio Spirits and then redistributed via syndicated broadcast, as well as on CD and cassette. All four AudioCubes will share storage resources on a 100Mbit Ethernet network, and are configured identically. VPI's selected for each Audio-Cube include DeClicker, DeNoiser, Loudness Maximizer, Spectralizer, De-Crackler, DeScratcher, DeClipper, DeBuZZ, and Analog EQ. In addition to the four AudioCubes, Radio Spirits will be installing Spectral's CD Inspector.
**Spectral Design, Germany.
Tel: +49 421 22 1440.**
■ British producer and Lightning Seeds front-man, Ian Broudie's has taken delivery of an Audient ASP8024 ana-logue console, Otari RADAR and a Digidesign Pro Tools system from Stirling Syco for his Richmond-based studio, The Barge. The console will be used primarily for tracking.
Expotus, UK. Tel: +44 1923 252998.
■ Canada's Salter Street Digital audio and video postproduction division of Salter Street Films, has installed a Fair-light Merlin in its Halifax, Nova Scotia facility. The Merlin has replaced an MFX3-plus in the facility's mixing theatre, where it is operating as both a dubber and a digital editing system. One of its first uses is for the acclaimed current affairs program *The Awful Truth* with host Michael Moore. In Montreal, the Media Principia film production com-pany has ordered a 64-input, 32-bus Euphonix System 5 digital console for installation in its new Mix Room. Owner Daniel Langlois, founder of SoftImage, is concentrating on new creative techniques for motion-picture production and exhibition.
**Fairlight, US. Tel: +1 213 460 4884.
Euphonix, US. Tel: +1 415 855 0400.**
■ London's Soho has seen two new White Mark designed post facilities open: Wild Tracks Studio 7 is one of a 2-room suite recently completed in the basement of the company's Greek Street premises and is the first room designed from the outset to support Dolby EX. The control room is fitted with a 6.1 Genelec monitor system set up by Matt Dobson with BSS 366 Omnidrive crossovers. 4MC is opening newly completed post suites at its Wardour Street premises. These are configured as two near identical control rooms with attached voice-over/Foley booths. The complex also includes a mastering facility equipped with 5.1 surround monitoring.
**Wild Tracks, UK.
Tel: +44 20 7734 6331.
4MC, UK. Tel: +44 20 7878 0000.
White Mark, UK.
Net: www.whitemark.com**
■ Birmingham-based Waterside Sound Studios has installed CEDAR CRX decrackler, DHX dehisser and BRX+ debuzzer processors. The British facility has worked on recent productions for the BBC, ITV and Channel 4 Television including Top Gear, The Cook Report: Millionaires and Embarrassing Illnesses.
**Waterside Sound, UK.
Tel: +44 121 633 3545.
CEDAR Audio, UK.
Tel: +44 1223 414117.**

▲ **India: AMS Neve claimed collateral credit for Hitendra Ghosh's success at this year's Ux Zee Cine Awards in India. Honoured with Best Recording Award for his mix of the film *Janwar*—starring Kshay Kumar, Karishma Kapoor and Shilpa Shetty—Ghosh (pictured seated) used the AMS Neve Logic 2 console installed at Gaurav Digital in Mumbai to secure his award which follows work on over 1,000 feature films and the honour of numerous other awards.**

Sennheiser Scholars

UK: With the support of Senn-heiser UK, the British National Film and Television School has inaugurated a student scholarship scheme for students of the 2-year NFTS Diploma specialising in screen sound. The award covers fees and provides an allowance towards living and travelling expenses. The first recipient is Robert Bourke from Dublin, whose final year studies will include shooting an animation, fiction and a documentary short. Locations will include Nevada, Utah, LA and San Francisco while dubbing and post work will involve surround sound and the NFTS' Harrison console, Audio

Vision, Pro Tools and Fairlight systems. Sennheiser UK MD Paul Whiting commented: 'We are extremely pleased to award the very first Sennheiser NFTS Schol-arship to a student of Robert's out-standing calibre. I think that his achievements at the NFTS are rep-resentative of the reasons for our funding initiative here.'

Production on the Net

US: Euphonix has announced the launch of Listen-In, an Internet-enabled remote monitoring ser-vice available to users of the System 5 digital console. 'The L s-ten-In Service allows customers of our System 5 facilities to offer a quality live, secure access to a

▲ **UK: London's West End postproduction community recently saw the opening of Qtwo and a new room at Goldcrest. Qtwo (pictured) offers a good location for recordings and live broadcasts as well as addressing Internet radio, temporary broadcast radio stations and consultancy services 'across the broadcast industry spectrum'. It is backed by the Unique Broadcasting Group and was set up by the Unique Facilities operation. Goldcrest, meanwhile, has knocked together control and live rooms to create a new theatre equipped with an Otari Elite film console for mixing, Foley and dialogue replacement applications. The 80-channel Elite uses the PicMix monitor routeing system and is paired with Otari's RADAR nonlinear recorder-editor**

control room mix via a high-speed Internet connection,' explains Scott Silfvast, Euphonix founder and Chief Product Officer. 'Not everyone involved in the creative process of recording and mixing can be located physically in the control room at the same time. But a producer accessing the Internet from anywhere in the world—and through a secure link —can now listen-in to the mix as it's happening live in the studio.'

John McGleenan, facility director at One Union Recording Studios adds, 'Listen-In will give my clients a safe and secure way to hear a very high-quality audio mix of their TV and radio spots over the Internet and offer real-time direction from a remote location. They can monitor any System 5 studio live, to approve a mix or voice-over session as if they were in the studio.'

'Audio production is breaking out of its traditional boundaries,' agrees Barry Margerum, Euphonix CEO and president. 'No longer does creative talent need to travel to a central production facility; the Internet provides an exciting environment for collaborative endeavours.'

NBC on Dolby E

**US:** American National broadcaster NBC is to use Dolby E and associated technology for the distribution of multichannel and stereo for its with DTV programming.

NBC will use Dolby E encoders at the Network to encode audio for which Dolby will provide technical assistance to NBC and its affiliates to ensure a smooth transition to multichannel sound. NBC began using Dolby Digital equipment for DTV beginning with the first high-definition broadcast of the *Tonight* Show with Jay Leno in April of 1999. 'New Dolby E products provide the technology we need to broadcast multichannel audio out to our audiences,' said Jim Starzynski, principal engineer, technical planning and engineering, NBC.

Dolby E-encoded audio does not reach the consumer; it is re-encoded with Dolby Digital, the coding used for ATSC DTV broadcasts, DVD-Video discs, and other consumer media, just prior to transmission. 'Dolby's help and technical expertise will be valuable in supporting NBC's goal to provide the best technical programming to our viewers and listeners,' Starzynski concluded.

Fairlight firsts

US: First sales of the Fairlight Prodigy integrated digital-audio production system follow its North American launch at NAB
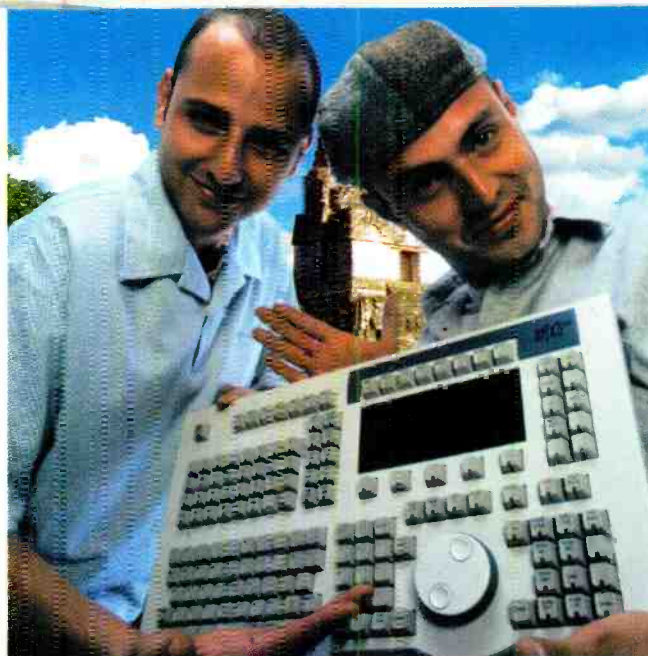
▲ Mexico: Leading commercial postproduction facility, Fiction, is installing the country's first combination of the Fairlight MFX3-plus workstation and ViVid digital nonlinear video recorder. The facility, founded two years ago by industry veteran Scot Evans, selected the Fairlight MFX3plus as the platform to launch its new audio services division for which the majority of work comes from commercials for television broadcast and cinema trailers for clients including Coca-Cola and Ford.

2000. In the US, the first two Prodigys will go to Commercial Recording of Cleveland and the third to North Hollywood-based Pacific Soundwaves. Soundwaves specialises in episodic television for clients including HBO and The Disney Channel while Commercial Recording is a leading postproduction facility handling commercial spot work for the likes of Goodyear, Applebee's, and McDonald's. Both use Fairlight systems, Pacific Soundwaves having two MFX3plus workstations and Commercial Recording two FAME2 systems linked via a MediaLink server.

▲ Germany: Leading facilities house, TVN, has invested E11m in two new digital AV remote trucks equipped with 120-input Studer D950 consoles (pictured outside the Olympic Stadium in Munich). Already established with most of the major TV stations in Germany, the increasing need for remote productions led TVN to design and build the ultra-modern remote truck in cooperation with Toutenkamion (France) and Protec/Bingen (Germany). Thomas Hamann, sound engineer responsible for the concept of the audio section of the truck, stated: 'We were looking not only for an adequate high-tech digital mixing console, but also for the one offering a future-proof investment'. TVN, Germany. Tel: +49 511 12 12 450.

www.americanradiohistory.com

US: Manhattan's Photomag post house has added a second Soundtracs DPC-II digital console in its downtown Photomag@The Anx facility. The recently refitted Studio A will see the new DPC-II working alongside a Fairlight MFX3+ and is 5.1 equipped. Photomag 'Partners in Sound', US. Tel: +1 212 687 9030.

UK: Sir George Martin has been presented with serial number 001 of the ISA 110 Limited Edition re-issue mic pre and EQ from Focusrite. The presentation acknowledged the relationship between London's AIR Studios and Focusrite, which dates back to 1985; 'I am delighted to see the new special edition of the wonderful Focusrite module, which Rupert Neve designed for Air Studios many years ago,' said Sir George. 'It is something we use constantly today and its unique design is very popular with the leading engineers of the world' AIR Studios boasts a large collection of Focusrite including the original 16 vertically racked Focusrite ISA 110 modules for the custom Neve console that were installed in 1988. The ISA 110 was modelled on Rupert Neve's 31106 EQ module designed for the A7971 by Rupert Neve, Sir George Martin and Geoff Emerick. AIR also owns four original ISA 110 modules which are used as outboard mic-pres and EQs.

Studio Sound   **www.prostudio.com/studiosound**

Search for a star

Net: Two new web-sites intended to smooth the search for suitable studio and record company contacts have just taken to the air. Hobomusic.com allows record companies, personnel, and recording studios (on locations, price, availability, equipment) can all be searched giving what is called 'an impartial family tree' of the recording industry offering contacts with 'confidence and reliability'. HitQuarters .com, meanwhile, claims to be 'the most detailed database of names, companies, addresses, numbers, recommendations, links, charts, contacts, classifieds and so on, ever made within the music business, free of charge'. HitQuarters' mission is 'to direct you to the right persons by providing information, advice and recommendations, thanks to our uniquely designed track-record search, the HitTracker, which reveals who is worth dealing with'.

www.hobomusic.com
www.hitquarters.com

High Gain

AFTER OVER 29 YEARS supplying professional audio equipment to studios, broadcasters governments and the military, Sonifex thought the time right to offer *Studio Sound* readers a chance to win a selection from its Redbox range—an exclusive gold livery version of it's distribution amplifier.

Redboxes began production in 1999 and are intended for use in various applications in radio stations, TV studios, video and recording suites. Having just passed the 1,000 sales mark, the range includes analogue units such as the RB-MA2 dual mic amp, RB-SM2 stereo-to-mono convertor, RB-SL2 2-channel limiter and RB-HD6 6-way headphone distribution amplifier, and digital boxes such as the RB-SC1 sample-rate convertor, RB-ADDA audio convertor, RB-DDA6A AES-EBU distribution amp and the new RB-MM1 mix minus generator. The complete list can be found at www.sonifex.co.uk

Studio Sound's competition prize is also a new unit called the RB-DA6, a high-performance 6-way stereo distribution amplifier that enables a stereo source to be directed to up to six stereo individually-buffered outputs. Best of all, we have three RB-DA6s—serial numbers 1000, 1001, and 1002—to give away to you.

In the time-honoured tradition, all you have to do is to answer three simple questions about Sonifex and its Redboxes. And for a little further entertainment, there's a tie-breaker.

▶ 1. In what year was Sonifex founded?

▶ 2. How many stereo audio outputs does the RB-DA6 have?

▶ 3. Sonifex also manufacture a portable hard-disk recorder—what is its name?

▶ Tie-breaker: Suggest a new product for the Redbox series and the reason why.

To enter, you can either email your answers to sscompetition@unmf.com, fax them to +44 20 7407 7102 or send them on a postcard to Sonifex Competition, Studio Sound Magazine, Miller Freeman Entertainment, 8 Montague Close, London SE1 9UR, UK before Friday, 28th July 2000. All entries will be subject to an infallible new checking system hotlinked to the Southwark recycling depot. Good luck.

World Wide Audio, without leaving your studio.

Rocket Network takes audio production beyond the boundaries of studio walls, making connections that let you work with anyone, anywhere, anytime. It's like a global multi-track.

On-line Flexibility.

Rocket Network uses the Internet to allow professionals to work together on audio productions without having to be in the same physical space. Instead of shipping tapes from place to place or renting high-capacity phone lines, you log into your Internet Recording Studio, where Rocket Network handles the details of passing your parts to others and vice versa. That leaves you free to concentrate on capturing the perfect take, using your own local system to record and edit. Whenever you're ready for others to hear your audio or MIDI parts, you simply post your work to the Internet Recording Studio, automatically updating everyone else's session.

Full Audio Fidelity.

With Rocket Network, there's no compromise in audio quality—the system handles files in a vast range of formats and compression levels, all the way up to uncompressed 24 bit/96kHz. And you don't need access to a super-fast connection; DSL or T1 is great, but you can also work productively over a humble 28.8 dial-up. The system supports multiple user-defined presets for posting and receiving, and handles all conversions, letting everyone participate in their own preferred format. That means you can conduct a session in a speedy, low bit-rate "draft" mode, then move on while the final parts are posted in the background at full-fidelity.

Professional Tools.

Through partnerships with leading audio developers, Rocket Network is bringing RocketPower™ to the professional tools you already use, starting with Steinberg Cubase VST and Emagic Logic Audio. A multi-level permission system lets you control access to your Internet Recording Studio. And our RocketControl™ client offers built-in chat capabilities, so everyone in the session can chime in with feedback as the project takes shape. The Rocket Network Web site offers additional resources and services for audio collaboration.

A Powerful Connection.

Rocket Network adds a new level of freedom to creative collaboration, allowing you to choose your team—singers, musicians, voice–talent, composers, engineers, producers—based on who's right for the project, wherever they happen to be. With full fidelity, plus anytime, anywhere productivity, Rocket Network is a powerful new connection to the world of audio production.

Escape the boundaries of your studio walls.

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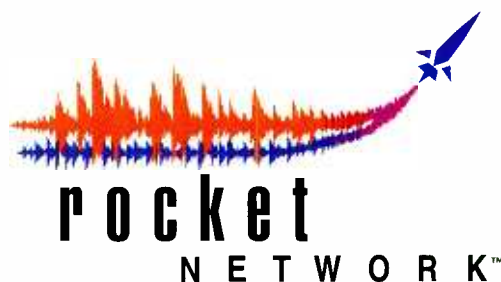
digidesign
A division of Avid   emagic   Euphonix   gvox   Steinberg

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June

3–6
Nightwave
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Rimini, Italy.
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Fiera di Rimini.
Tel: +39 541 711 711.
Net: www.fierarimini.it

6–9
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Cablesat 2000 and
Professional Audio
Technology 2000**
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Exhibition Services.
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Net:www.broadcast-asia.com

7–9
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Intex, Shanghai, China.
Contact: P&O Events.
Tel: +44 (0)20 7370 8231.
Fax: +44 (0)20 7370 8143.
Email: shanghai@eco.co.uk

9–18
**Synthèse 2000
International Festival
of Music and Creations
Electronic**
Palais Jacques Coeur,
Bourges, France.
Contact: Institute de Musique
Electroacoustique de Bourge.
Tel: +33 2 48 20 41 87.
Email: ime-bourges@gmeb.fr
Net: www.gmeb.fr

13
**Lecture:
Sound Systems
Control Networks**
Baden Powell House,

South Kensington House,
London, SW7, UK.
Contact: AES.
Tel: +44 01628 663725.
Email: uk@aes.org

13–15
**ICCE 2000:
International
Conference on
Consumer Electronics**
Los Angeles Airport Marriott,
5855 West Century Boulevard,
Los Angeles, California, USA.
Contact: Diane D Williams,
Institute of Electrical and
Electronics Engineers.
Tel: +1 716 392 3862.
Email: d.williams@ieee.org
Net: www.icce.org

13–15
Replitech
The Moscone Centre,
San Francisco, California, US.
Tel: +1 914 328 9157.
Fax: +1 914 328 2020.
Email: kipi.event@kipi.com
Net: www.REPLITECH.com

20–22
Pro AV Africa
Contact: FEI.
Tel: 0118 977 0611.

22–23
**Broadcast Content
Management 2000**
The British Library Conference
Centre, London, UK.
Contact: Miller Freeman
Entertainment, David Roberts
Tel: +44 (0)20 7940 8561.
Email: david.roberts@unmf.com

23
**Creative Industries,
new perspectives
for a new century**
Sheffield Hallam University, UK.
Contact: Jon Pyle
Tel: +44 114 225 4187.
Net: www.shu.ac.uk/creative

26–39
Cinema Expo
Amsterdam, The Netherlands

28–30
**Real Conference
Europe 2000**
Disneyland, Paris, France.

Contact: Noiseworks.
Tel: +44 1628 628 080.
Email: sharon@noiseworks.com
Net: www.europe.real.com
/conference2000

July

3–5
Radio Festival 2000
Scottish Exhibition
& Conference Centre,
Glasgow, Scotland, UK
Contact: Cate Gibbert,
The Radio Academy.
Tel: +44 (0)20 7255 2010.
Email: festival@radioacademy.org
Net: www.radioacademy.org
/festival
Net: www.secc.co.uk

6–8
PALA Philippines 2000
World Trade Centre Metro
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Email: jchin@iirx.com.sg

21–23
Summer Namm
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August

23–26
BIRTV 2000
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Email: shanghai@eco.co.uk

September

4–7
IECEP 2000
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www.prostudio.com/studiosound   Studio Sound

Fostex D1624

Latest in Fostex' digital recorder line is the D1624 which continues a train of thought and comes complete with fast-and-wide recording capabilities. **Rob James** punches in

THE D1624 CONTINUES the Fostex tradition of workmanlike, cost-effective recorders. In many ways this machine is a development of the D-160 and, like its predecessor, it operates as a 16-track digital recorder at 44.1kHz or 48kHz sampling rates but also offers eight tracks at 96kHz 24-bit. In keeping with this, the A–D and D–A convertors have been suitably uprated to 24-bit sigma-delta 128x oversampling from the previous 18-bit A–Ds and 20-bit D–As.

The external package is pretty much unchanged from the D-160. A 3U-high rack-mounting case fitted with a 3½-inch hard-drive caddy and the socketry is topped off with a removable control panel in the Fostex house colours. The front of the main unit contains just a power switch, recessed 15-pin D-connector for the control panel, removable disk slot and four steel mushrooms. The control panel is firmly attached to the machine by slotting it over these. The idea is a good one, instead of an expensive accessory remote

control all you need is an extension cable in order to site the panel up to 10m away.

The chunky metal cased disk carrier is supplied empty. Fostex offers extra empty caddies at reasonable cost and a short list of tested E-IDE drives is in the manual although I have no doubt many alternatives would be satisfactory. The SCSI connection is purely for backing up or exporting data from the internal drive. On the D-160 a suitable SCSI drive could be used directly. Another change, definitely for the better is .WAV file import and export. This is accomplished using a DOS (FAT16) formatted SCSI drive. If you wish to load files originated on other systems the restriction is the file name. This must be a maximum of six characters with a 2-digit track number and the .WAV extension.

The internal drive is formatted to suit the bit depth and sampling rate required. Bit depths and sampling rates cannot be mixed on the same drive even in separate projects. The other major operational choice is whether to initiate the Multiple Undo function. The trade-off is a simple one: with the function on, all recordings and edits are retained taking up disk space, with it off only the last recording or edit is kept—giving one level of undo-redo.

The D1624 uses Fostex' FDMS-3 v2 (Fostex Disk Management System) This is limited to a maximum of 99 program files per disk. Recording can take place anywhere in a

24-hour 'window' on any track up to the maximum capacity of the disk. Apart from the 8 or 16 'real' tracks there are 8 additional tracks. Complete 'real' tracks may be swapped with additional tracks to free space for further recording, individually or in 8 track banks.

The D1624 records individual files for each recorded 'event' and it is possible to insert silence between existing events. Silence is counted as an event—this is important since the maximum number of events per track is 512, although recording is not possible after 507 events on a given track. When editing, each edit point splits existing events, further increasing the number. The number of events can be monitored using a menu. Fostex suggests a number of strategies to reduce the number of events.

At 44.1kHz or 48kHz recordings may be up to 16 tracks wide if you have either 16 digital sources or 8 analogue and 8 digital sources. At 96kHz the limit is 8. Punch-in recording can be either automatic or manual from the control panel or optional foot switch. Rehearse mode switches the monitoring from replay to source in the familiar manner. However, when you actually go for it, the track(s) monitoring switches to replay about 2s after the actual punch-out. After punching out the transport must be stopped before further recording is possible. There is a delay of a few seconds after stopping before you can do anything else while 'housekeeping' takes place.

Monitor switching is otherwise accomplished by arming a track for recording and pressing the REC key. Alternatively, pressing the TRACK SHIFT and SHIFT keys together switches all tracks to input monitor. Since the machine is intended for use with a mixer there are no internal mixing or level control options for track bouncing or monitoring.

The SETUP key gives access to the many initial

ON THE REAR PANEL unbalanced analogue I-O is a block of 8 phono connectors for inputs and a block of 16 outputs. The analogue inputs feed pairs of tracks 1 & 9, 2 & 10 and so on. Four TOSlink optical connectors cater for all 16 tracks of I-O at 44.1kHz or 48kHz sampling rates and are also used when backing up to ADAT or DAT. For 96kHz digital recording the optional 8 channel AES-EBU cards Model 8350 are required. Word clock I-O is on BNC sockets with a small termination switch and MIDI In, Out and Thru are DINs. The two RS-422 serial 9-pin sub-D connectors for remote input and through carry either Sony P2 protocol or Fostex System Exclusive data. A 100Ω termination switch on the input allows for terminating the last machine in a chain. The SCSI connection is a half pitch 50-pin item.

The blank panels, two small and one large, are for option boards. The big slot takes either the Model 5043 balanced analogue board that mirrors the phonos and gives 8 ins and 16 outs or two Model 8350 AES-EBU digital I-O boards. The Model 8345 Time Code and Sync board uses the smaller slots and adds video sync and time-code connectivity via XLRs for Time Code I-O and BNCs for Video In and Thru. Video has switchable 75Ω termination.

On the control panel the connection cable to the main unit is at the top left corner. A ¼-inch jack below connects a punch in/punch out footswitch.

The large, bright display is divided into a block of 16 bargraphs with 16, 35 dot alphanumeric characters across the top which display parameter name messages, time or bar-beat-click and

time remaining on disk. RECORD TRACK select buttons are below in two banks of four (plus TRACK SHIFT key) to access the 16 tracks. Below the track select keys three LEDs indicate All Input, (Disk) Access and TC Ready. To the right of the display are the dedicated status and mode indicators, sample rate, -1% (dropdown) COMPLETED, SURE?, and so on. To the right of the display the top row of keys begin with AUTOPLAY. This cycles through the autoplay, autoreturn and auto repeat options. Two associated LEDs indicate which, if any, of the functions is selected. Two keys store and Recall Clipboard In and Out times. The Autoreturn section has four keys for START, AUTOPUNCH IN, OUT and END. A separate Autopunch mode ON-OFF key sits above the RECORD key with two LEDs to indicate rehearse or take.

settings and submenus. These can be divided into three groups: Changing the initial settings, Check (for checking the number of events in a Program) and Execution (used for the more destructive options such as disk formatting and erasing Programs). The jog wheel scrolls through the circular list of possibilities.

Time can be displayed as absolute which is based on each new program starting at zero hours and continuing to a maximum of 24 hours, as MTC which is an offset added to ABS or MIDI bar-beat-clock which corresponds to the MIDI Clock-Song Position pointer. Adding the optional time-code board allows the machine to chase and generate LTC and adds 9-pin slave abilities. If 9-pin is required, it would be wise to check the D1624 works with your master controller. In fact this is a sensible precaution with any '9-pin' equipment.

There are a variety of ways of navigating through Programs. The 99 numbered locator memories are displayed in time or bar-beat-clock depending on mode. The CLIPBOARD IN-OUT, AUTO PUNCH IN-OUT, AUTO RTN START-END and LOCATE keys plus the STOP REWIND and FFWD keys in combination all store locations.

The editing functions are basic but I don't think editing is the primary purpose of this machine. As it is, editing is more than adequate for a tape recorder replacement. Practically, in and out points are defined and a function chosen. Copy and Paste makes a copy of the data and pastes it to another location via the clipboard, even between Programs. Move and Paste does almost the same thing but erases the source data including the clipboard and can only be used in the current Program. Using Copy and Paste multiple copies can be pasted in one operation to extend a sound. ERASE gives two methods of erasing data. Either erasing from a specified point to the end of program on a track or tracks, or erasing between two specified

points. All edits come with a fixed fade of 10mS or 5mS at 96kHz. To aid location of suitable edit points, the Envelope scrub function is available. This is invoked by pressing the SHIFT key and a RECORD TRACK select key with the transport stationary. The selected track is then 'scrubbed' with the jog dial. A rather coarse display of the waveform (envelope) is given on the bargraphs. Track Swap allows complete real or additional tracks to be exchanged singly or in 8 track groups.

Fostex has succeeded in producing a competent 8-track 96kHz 24-bit recorder with the bonus of 16-track capabilities at 44.1kHz or 48 kHz. The D-160 formula of providing a replacement for an analogue or digital multi-track with the added benefits of basic 'cut and paste' editing at an attractive price is maintained in the D1624. Connectivity has been enhanced with MMC, RS422 and numerous backup options. The user-interface has been tidied up somewhat.

Tape-style cut and paste editing is fairly painless and Fostex has made it relatively simple to move material requiring serious work to a DAW with WAV file compatibility. This adds a powerful argument to the case for the D1624. If the removable E-IDE drive could be read directly by a DAW the case would be stronger still. The cost per gigabyte of E-IDE drives has dropped dramatically in the last year to the point where you can now record 2496 for the same cost per minute as 1648 a year ago. This is now a viable option for original recordings regardless of the intended distribution format.

I could criticise this machine for its eccentric transport key arrangement, simple editing, repeating loop scrub and time spent housekeeping, but I think to do so would entirely miss the point. Truly professional kit concentrates on core purpose and does not rely on larding a product with unnecessary features in order to distinguish it from the herd. The D1624 is a useful step in the right direction and makes a lot of sense when considered as a tape recorder replacement. ∎

Minnetonka MX51

Aimed squarely at users of Yamaha's DSP Factory, MX51 is a virtual surround mixing package with a few surprises. **Rob James** goes into games mode

A COUPLE OF YEARS AGO I looked at Minnetonka's MxTrax, a multi-track DAW software package for PCs. At that time MxTrax worked with the DAL PC sound cards. Since then, the Minnesota-based software house has been busy —MX51 for DSP Factory is one result, building on the MxTrax interface but being specifically designed to take advantage of the massive capabilities and low cost of Yamaha's DSP Factory with, as its name implies, 5.1 surround capabilities. Minnetonka has also added support for the Microsoft Sidewinder Force Feedback Pro joystick and several other, more conventional, hardware controllers.

Software installation is straightforward but even if you are familiar with other PC audio packages, I would advise taking the time to work through the tutorial. It provides a relatively painless introduction to the Minnetonka way of doing things.

The default setup divides the screen into two main sections, track displays at the top and mixer at the bottom. The split point is moveable with the mouse to reveal more tracks or more mixer according to the current task, or either area may be zoomed full screen. Transport control is conventional with keyboard shortcuts supplementing the on-screen buttons. The track display is of the moving-cursor type so the screen redraws when you reach the edge. Much as

I like moving tracks with a static cursor, the processor overhead, even on fast machines, usually makes this a redundant option.

The real fun begins once you've found your way around. I have always liked 'build

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it yourself' mixers on software DAWs. Some manufacturers make a virtue of the absence of this facility, citing 'ease of use' in mitigation. Well, nobody is forcing you to use the mixer design functions, there are plenty of library mixers available but I think there is more satisfaction and utility in rolling your own for specific tasks. The great joy of this particular software-hardware combination is you don't really have to worry about running out of processing power. Providing your aspirations don't exceed the maxima imposed by the DSP factory card or cards there will be no problem. A mixer is 'built' by dragging and dropping components from a floating toolbar palette. Apart from blocks which control the kit of parts provided by Yamaha, Minnetonka has thoughtfully provided vertical and horizontal spacer blocks to help keep designs clear. To begin construction from scratch an input fader is dragged onto the blank sheet. Other channel components are added in whatever order suits you. Channels can be saved as objects, copied and pasted singly or multiply. Output buss controls are added in the same way and a complete 5.1 Output Monitor is added as a single component. Once defined, default connections are made to the physical studio monitoring outputs whenever a surround monitor is placed.

An easy way to patch is to left click on, say, the track control area of a track and drag down to the fader-meter area of the input channel you wish to connect it to. The cursor changes to a representation of a jack plug to show you what you are doing. The record, playback and editing functions are much as before which is to say workmanlike with most of the things you would expect to find. Editing is quick and easy to use without too many frills.

I would like to have seen some more stereo functions. Tracks or segments can be moved as stereo pairs by grouping them but it is too easy to forget and get things out of sync, although the 100 levels of undo come in handy here. It would also be nice to be able to link adjacent mixer strips for stereo. **>**

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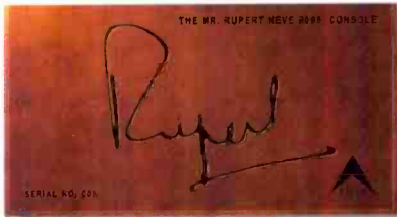
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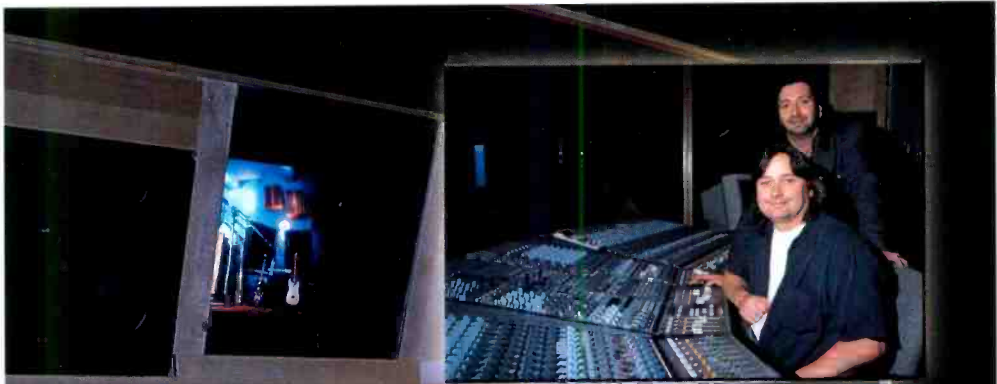
Reid Hyams (standing), President and
Operations Manager of Chicago Trax, with
Chris Steinmetz, Chief Engineer

9098i

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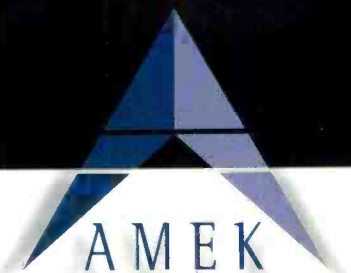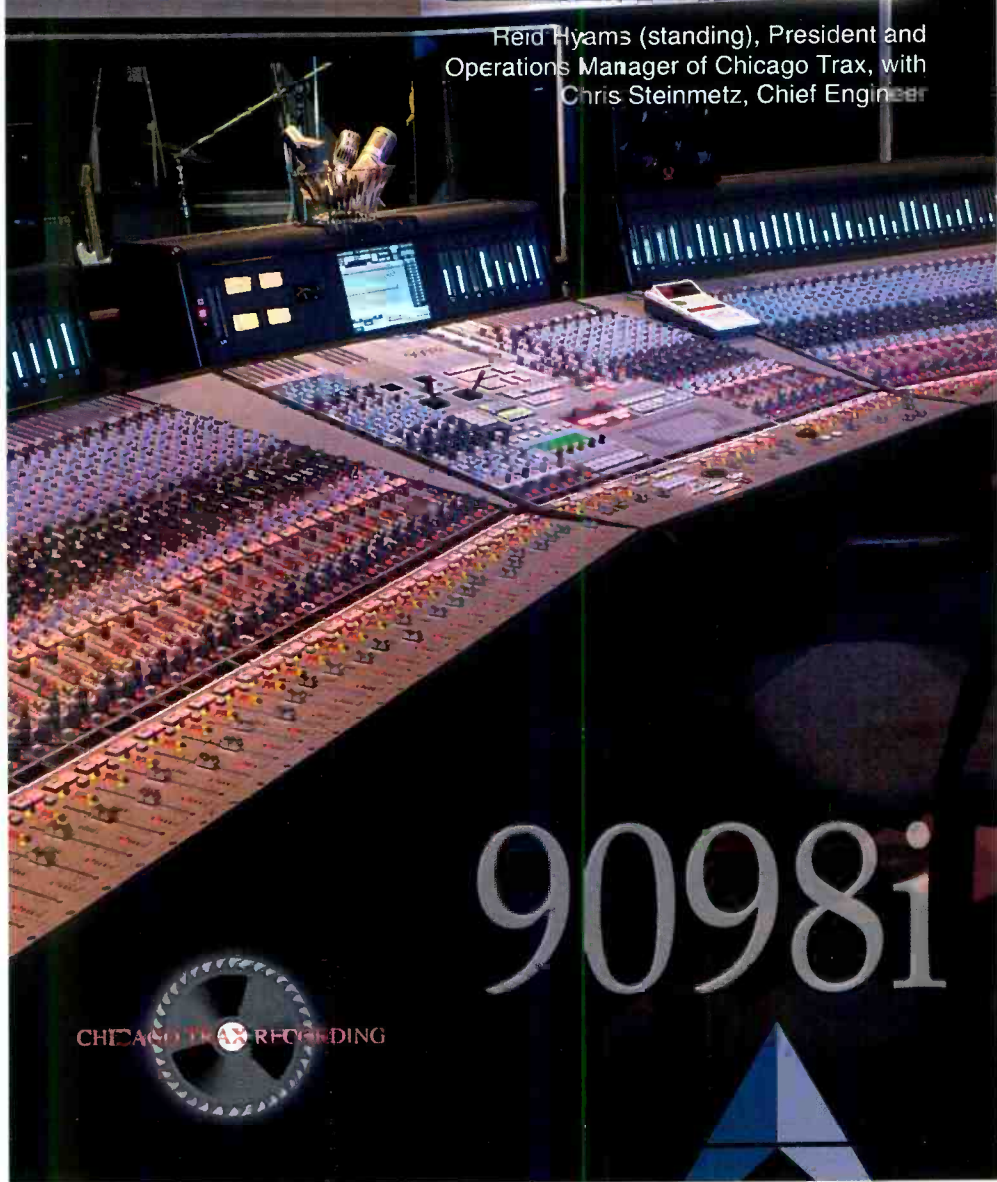
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< The mixer automation is simple and easy to use. All functions except solo are automated. The three possible modes, Read, Write, and Update are selected globally from a pull-down menu or on a per-channel basis using the A keys which show green, red or green-red to indicate the respective modes. There is no direct way of saving mix passes but a workaround is to save the project under different names. The other luxury I missed was the lack of a 'glide' function. Dropping out of automation record results in all the parameters currently in write immediately reverting to the previous value.

For surround work you simply drop a Surround Panner and SubBass filter into the channel strip. The panner graphic uses a red ball to indicate the current position and a light coloured V shape to show the divergence or spread of a centre source to left and right. The surround panner block can be zoomed to facilitate finer adjustment. To pan you simply grab the red ball and move it with the mouse, or you could use a force feedback joystick. If this seems a curious choice of controller for audio use then consider the cost of the alternatives and don't knock it until you try it. Apart from left right, X-Y axis, panning control this joystick has several but-

> To pan you move it with the mouse, or you could use a force feedback joystick. If this seems a curious choice of controller don't knock it until you try it

tons and can swivel (used for rudder control on flight simulators). On the top of the stick the big button on the left toggles transport stop-start and a mini 4-axis switch is used to select which mixer channel is controlled. Two more buttons toggle solo and zoom. The fire button is used to write automation to the selected channel. The channel automation mode is toggled by the A button at the base of the stick. The rudder control is used to position the playback cursor. Prior to pressing the fire button the force feedback is used to give an indication of the pan position at any given moment. Once the trigger is pulled to update the automation, the feedback can give you the location of the pre-existing panning information. This is accompanied by a grey ball in the panner display. The object of all this is to enable a seamless transition to the previous position. The amount

of force feedback is adjustable. In practice, I found it rather crude but fun. It also allows for fast working. If the force feedback joystick doesn't appeal there are other options. MX51 also supports Mackie's HUI, the Yamaha O1V, Peavey PC 1600x and CM Automation 'Motormix' hardware controllers via MIDI.

The Yamaha DSP factory has failed to attract the level of support among writers of non-sequencer-based software front ends I expected when it was launched. The merits of the approach should be obvious to anyone who has spent frustrating hours juggling scarce resources with software which uses native processing or even some of the DSP card alternatives which try to address the problem. The only real potential downside of the DSP factory is the timbre of the effects, dynamics and EQ are hard coded by Yamaha. In fact, they are the same as the algorithms used in the estimable 02R. For many purposes, this and the reliability and ease of use conferred, is well worth while and of course, if you are not a fan of the 'Yamaha Sound', there is nothing to prevent the use of software supporting other plug-ins alongside. Current versions of software such as MX51 and C-Mexx' C-Console with SEK'D Samplitude as the editor make the case for DSP factory even clearer.

By developing MX51, Minnetonka has provided a plausible alternative to some of the more obvious names. The surround features are a good demonstration of just how commonplace mixing in 5.1 is becoming in the US. Forthcoming attractions are software encoders for Dolby Digital and DTS. MX51 is also available for use with DAL hardware. On the Yamaha-PC platform the software appears robust and a good number of tracks are possible with PCs which usually struggle to keep up. If juggling resources is becoming tedious MX51 may provide the answer to a prayer. ∎

Wendt X2

Being small has its obvious and less apparent advantages, as in the case of Wendt's 2-channel location mixer **Neil Hillman** finds it perfectly formed

THERE IS A SCHOOL OF THOUGHT that believes that if it's big, it's good —big car, big guy, big hair, big fish. Big bill. Wendt have a product in the X2 2-channel mixer that's not big, but it is clever. Small size, small weight, small footprint, small outlay. Big performance.

All necessary adjustments and settings can be made easily, with accessibility obviously carefully thought out. The left-hand side of the mixer takes the microphone inputs on two electronically balanced female XLR sockets, with 12V Tonader, 48V phantom powering or dynamic settings selected via a 3-position slider switch located underneath and slightly to the left of each XLR input. Adjacent and to the right of each powering switch is a similar 3-position slider switch for amplifier gain, marked '10', '20' and '0'. By ear and comparison rather than instrument, I assessed '0' to correspond to a gain of 80dB, with the '10' position providing 10dB of attenuation to reduce amplifier gain to 70dB, and similarly the '20' position providing 20dB's worth of attenuation resulting in a net gain of 60dB. Below the inputs is the substantial battery compartment lid, indicative of the hugely robust feel of the Wendt X2, secured through a large-headed screw. The mixer takes six 1.5V AA batteries, which with care can turn the stated 12 hours endurance into a sensible two days location use.

Above the inputs on this left hand side is a ¼-inch stereo headphone socket with a level control knob alongside, with bags of gain available for my 250Ω DT150 headphones. Below the headphone socket is a 2-position slider, to switch the highly efficient limiter in or out. Only the very largest of input levels caused any discernible recovery-slump in its operation, and working with it 'in' and riding the gain to ensure only occasional peaks just pushed the '+6' vu meter-LED resulted in controlled use of the maximum available dynamic range. Alongside the headphone volume knob is a further 3-position slider switch to adjust the meter LED display intensity between Hi, Low and Off.

The meter LEDs are sited on the top of the front face above the SQN-esque rotary faders. Bright in the normal operating conditions of a Low setting, an intense laser-blue impression reminiscent of a trace oscilloscope is burnt into your retina should you ever be so foolhardy to consider the possibility of using the Hi setting. For a good 10 minutes after I tested on your behalf the meter's brightness, colleagues appeared to be walking around with a large '-3' in the middle of their foreheads. The vu meter scales are calibrated by four blue LEDs corresponding to '-9', -'6', '-3' and '0', with '+6' signified by a red LED. In the top left-hand corner of the front face, and alongside the row of LEDs is a toggle switch marked TAPE/MIX that enables the monitoring to change between the mixer output and the tape return signal. There is no level adjustment pot to trim the return signal on the mixer, so this needs to be adjusted on the headphone output of the recording device be it a DVCam

or MiniDisc. A second headphone output on a 3.5mm socket is sited beneath the return selector switch, to the left of channel one's fader, and this is a parallel feed of the ¼-inch socket on the left-hand side. The mixer has an in-built 400Hz, 0dB vu oscillator. It is engaged by means of a latching push-button sandwiched between the right-hand channel row of meter LED's and in the top right-hand corner, the green power monitor LED. This flashes on and off, in a ratio of about 1:2, to signify normal operating voltage and remains lit when the power is running low. When it doesn't come on at all is a sure-fire clue that either the batteries have completely expired, or the cameraman has walked off with your external NP-1 battery. The mixer is switched on through a 3-way slider beneath the power monitor LED, and is marked INT(internal), OFF and EXT(external). In between the two fader knobs is a hold-to-make, battery-test push-button that shows on the right hand row of meter LEDS, with full voltage corresponding to '0' on the meter scale. A slate-microphone, denoted 'MIC', is situated behind a grille in the bottom right-hand side of the front face and is operated by another white nylon hold-to-make push-button, identical to the battery-test switch, and located underneath the power switch. Beneath each fader is an associated pan-control that hard switches through a 3-way slider the channel output to either the 'L'(left), 'R'(right) or, when the centre 'PAN' position is selected, across both outputs. Also beneath the faders is their respective 3-way toggle-switch for the selection of low-cut filters, these being 12 dB/octave with turn-over frequencies of 140Hz, 20Hz and 100Hz.

The right-hand side of the mixer is the most uncluttered of the three control surfaces, carrying the two outputs on male XLR sockets. Above the two outputs are associated slider switches that set the output signal to Mic or Line levels. An unbalanced Mic Out on a 3.5mm socket provides a stereo feed at microphone level for unbalanced equipment, and is situated below the disappointingly flimsy external power connector. This accepts a 2.1mm coaxial power plug of the kind popular with manufacturers of transistor radios and really is a poor choice of connector given the otherwise unimpeachable build-quality of this mixer. The 3.5mm stereo socket for the Tape Return feed is closest to the operator on this right-hand panel, and sits alongside the preset pot for the gain adjustment of the slate microphone.

On location the X2 sat comfortably in the front pocket of the delightful Marantz PMD-650, and I found them to be ideally suited. Good things come in little packages. Small is the new big. ∎

TC Works Spark

Audio editing and mastering software with powerful plug-in effects
capability are a first for tc. **Simon Trask** goes in search of illumination...

COMING FROM PLUG-IN specialists TC Works, the Spark Realtime Digital Audio Editor program for the Mac has an impressive line in plug-in effects—four lines, to be exact. The program's FXMachine effects routeing matrix, accessed via the Master View window, supports up to 4 x 5 effects plug-ins; that is four parallel audio streams with up to five effect plug-ins on each stream—computer processing power and available RAM permitting. TC Works specifies a minimum 166MHz Power Mac 604 machine (Power Mac G3 recommended) running Mac OS 8.1 or higher with 64Mb RAM.

Spark comes in two versions: Spark (currently at v1.5) and Spark XL (v1.6). By the time you read this, Spark | Modular should also be available; this is an add-on component that brings modular synthesis into the FXMachine environment—or into any other VST environment. Spark supports VST native plug-ins, while XL adds support for TDM plug-ins and comes with two additional TC Works plug-ins for audio restoration work.

Spark XL running only in native mode (TDM support disabled) has the same system requirements as Spark. However, in order to run TDM plug-ins within Spark you need to have as minimum a Pro Tools 24 Mix system, and consequently a computer setup as specified by Digidesign. You'll also need DigiSystem INIT v5.0 or higher and the Direct Connect plug-in.

TDM plug-ins get their own Master View window, but in this case with a single serial arrangement of up to five plug-ins. However, you don't have to give up native VST processing to use TDM. Spark lets you use VST and TDM plug-ins at the same time, with the output of the VST FXMachine being routed into the TDM Master View chain.

The FXMachine effects matrix complements the Browser View and associated Cut Editor, which are where recording, editing and assembling is done. The mono or stereo audio file is fed into the FXMachine section, which at its simplest need use no effects at all but can provide a variety of effects routeing configurations.

FXMachine is not only an adjunct to the editing process, however. Select Realtime FX-Machine from the program's Options menu and it takes a feed from whatever audio source you've specified in its I-O Preferences.

A feature of Spark and Spark XL is that the FXmachine matrix is itself available as a VST plug-in. This means that it can be assigned as a plug-in effect within Record mode or to any of the plug-in slots in the FXMachine matrix in Realtime FXMachine and playback modes.

The FXMachine functionality can also be used outside of Spark as a VST or MAS plug-in, which means that it can be used in programs such as Cubase VST and Digital Performer. More accurately, FXMachine is a

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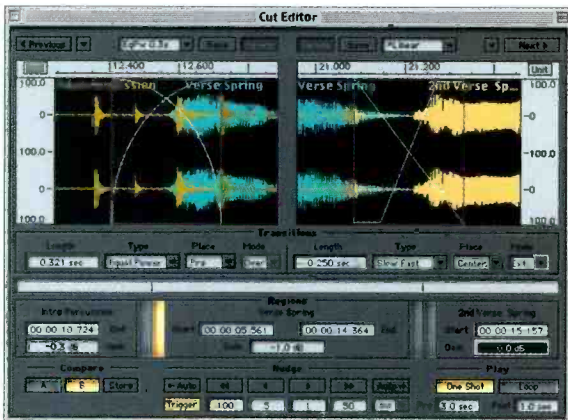
container for configurations of VST effects, so that within Digital Performer it acts as a wrapper for VST plug-ins.

Spark comes with 10 Spark VST plug-in effects (11 if you include the FXMachine in its plug-in form) and two additional TC plug-ins (Native CL and MaxIt), while Spark XL adds the TC DeClick and TC DeNoise plugins. What comes as standard, then, are bandpass, cutfilter, delay, expander, fuzzsat, grainalizer, oneband EQ, resfilter, reverb, and 3-band EQ Spark VST effects, plus the FXMachine plug-in and the two additional TC plugins mentioned.

To make the VST plug-ins available to the program you have to select the plug-ins folder in the Preferences window.Spark (all references include XL) supports the Steinberg-originated ASIO 2.0 audio streaming protocol for getting audio in and out, as well as the standard Soundmanager and Digidesign Direct I-O.

Spark's Browser View window (the Worksheet) contains the project file database (File View), audio editor (Wave Editor), and the file playlist (Playlist). Spark also has a Transport window that provides the usual controls plus a Jog Shuttle wheel for scrubbing (up to ±200%) or for setting realtime timestretching or varispeed amounts (up to ±25%). In addition, non-realtime (offline) timestretching and pitchshifting are available, via the Process menu, with associated parameters.

A Scroll Bar within the Transport window lets you move to any location in your audio file, and a CYCLE button lets you turn looping of the file or selected region on or off. You can

record from this window, or you can select the Record window from the Options menu, where you can select two effects to be added at the time of recording. The Record window is also where you select mono or stereo inputs, the record format (AIFF, SDII or WAV), and sample rate and resolution (11 options ranging from 8kHz–96kHz, and 8-bit, 16-bit or 24-bit).

In addition to recording via your audio card's inputs, you can also import audio from your Mac's CD-ROM drive, or import existing audio files off hard disk—including MP3 files, which are converted to AIFFs. Spark can also export to MP3 format.

TC Works has certainly maximised the flexibility of getting material in and out of Spark. The programlets you import samples from Akai sample CD-ROMs, and it supports import and export of samples from Akai S1000/1100, S2000 and S3000 ranges, E-mu ESI and E4 series, Roland S760 and Yamaha A3000.

With an audio file recorded or imported into the Browser View window you can set to work editing, creating regions and compiling playlists. The Wave Editor section of the window provides you with a familiar waveform display.

To quickly loop any section of the file, you just drag through it with the mouse in the edit view section of the Wave Editor; Spark automatically switches to looping this section. To adjust the loop start or end point in either direction, you just hold down the shift key and drag the left or right edge of the highlighted section accordingly. Extremely sim-

ple and quick. Regions appears as subfiles of the actual audio data file in the File View pane, where you can name them and save the audio data contained within a Region as an audio file in its own right. From this pane you can also add into the Playlist to create a new track or to create a playlist of whole tracks for a CD. Spark comes complete with Adaptec's Toast.

The program provides an array of processing functions for the audio data, with fade, change gain, normalise, reverse, invert, CD removal, resample, along with the aforementioned pitchshifting and timestretching, which work very well. You can also select a plug-in to process the data All the effects are editable via graphical 'front panel' displays.

I was very impressed by Spark's combination of power, versatility, depth, and yet ease of use. Some might say it goes above and beyond the call of duty as an editor with its plug-in effects capability, notably with the FXMachine matrix and the ability to use the FXMachine as a plug-in 'shell' in its own right—plus of course the XL version's ability to use VST and-or TDM effects in the same environment. However, these elements are an important part of what makes Spark special and creatively intriguing. The effects that TC Works have provided as standard are very capable, and provide a good basic set of effects that can be put to good creative use if you wish. It's also very easy to integrate into the Spark environment any plug-ins that you might already have. ■

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Lab Gruppen 300

Studio Sound's bench test amplifier reviews continue with the Lab Gruppen 300. **Paul Miller** reports

SMALL YET PERFECTLY formed, Lab Gruppen's 300 power amplifier is the baby of an 8-strong range that culminates in the 1.3kW Lab 4000 at £3,195 (UK ex-VAT). With one-tenth the power output of its bigger brother, the £695 Lab 300 still packs a considerable 110W punch from a very slim 1U-high rackmount case and is designed with 'safe operation' firmly in mind. Hook-up is painless with either balanced XLR or TRS jack inputs provided alongside a pair of Neutrik NL4FC Speakon sockets. Modes of operation include standard 2-channel stereo, tandem mono (2-channels driven from a single input), stereo reverse (phase inverted on channel B) and bridged-mono where some 300W/8Ω is cited.

Gain is switchable between 29dB and 23dB, the latter to optimise the usable fader range of modern mixing consoles. Two input attenuators are 'calibrated' over the pot's 40dB range, though I would not rely too heavily on the screen-printed values (see table).

Protection, meanwhile, is very comprehensive indeed and includes a clip-limiter that reduces the *input* signal level should the *output* exceed 1% THD. In practice, the limiter errs further on the side of caution and begins to bite above 0.4% or so, even under the dynamic conditions represented on Fig.1. With a 10ms cycle, the Lab 300 is capable of delivering 155W, 282W (8.4A), 423W (14.5A) and 323W (18.0A) into 8Ω, 4Ω, 2Ω and 1Ω loads, respectively. Good tol-

erance is demonstrated into 8Ω, 4Ω and 2Ω loads, but the impact of Gruppen's ALS (Adaptive Limiting System) is immediately evident from its behaviour into low, 1Ω loads (green trace).

DC input and offset protection will crowbar the output above 10V, though the inherent DC offset is no more than ±5mV. Thermal protection is included should the side-mounted heat-sinking rise above 90° though, short of a fault, it is difficult to see this occurring in any practical situation. Indeed, Gruppen suggests that several Lab 300's may be stacked without compromising their thermal stability and despite there being limited room for forced-air cooling.

In practice, the 300's reduced dissipation is achieved by its class-G operation, a combination of 25V and 55V supply rails that are switched according to signal and-or load conditions. This way, the two pairs of bipolar output devices (per channel) run rather cooler than they might in a typical class-A B or B design, although the switching itself is not entirely transparent. In all likelihood, the switchover will occur around 22-23V or 15-16Vrms, which corresponds to the anomalous increase in THD from 0.003% to 0.0125% on the right channel at 30W/8Ω. The extended pattern of odd-order harmonics is evident from the lower of the two combined plots that comprise Fig.2. Although this 'feature' is unintended, the result is still well within Gruppen's conservative 0.08% limit.

This same effect is de-

tected at higher frequencies (Fig.3) on both channels where HF distortion climbs quite rapidly above 30W/8Ω >
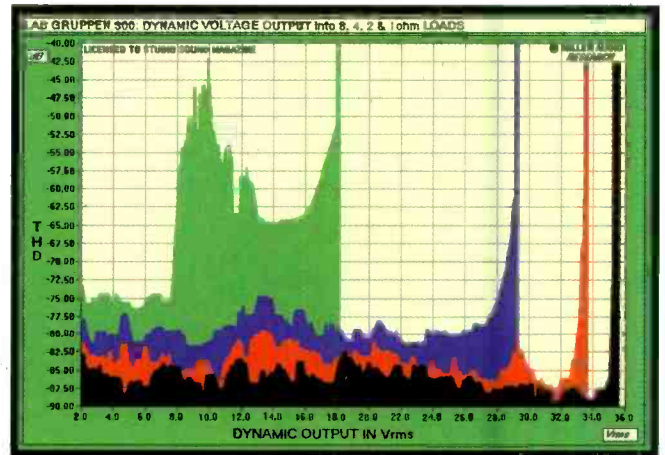
Fig.1
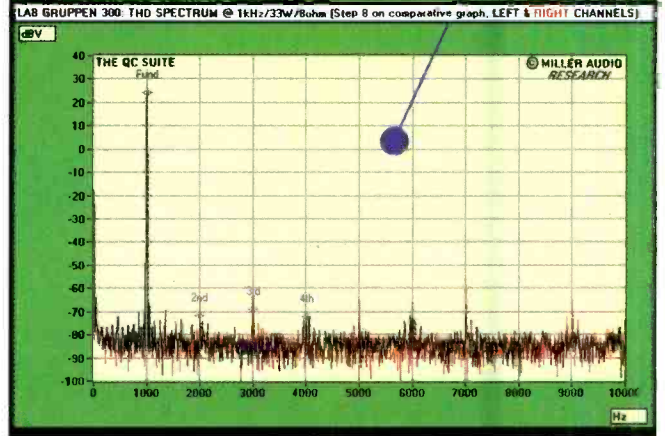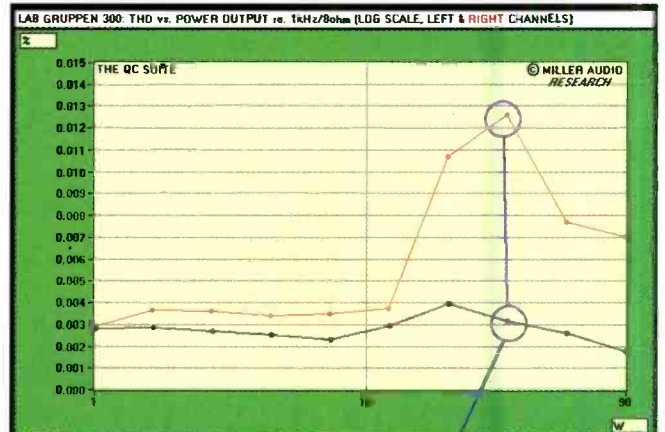
Fig.2

Attenuator Setting	'Real' Attenuation
-1dB	-0.15dB
-3dB	-1.8dB
-5dB	-3.7dB
-10dB	-7.8dB
-20dB	-17.5dB
-40dB	-31.5dB
-80dB	-85dB (Mute)

Precision with vision
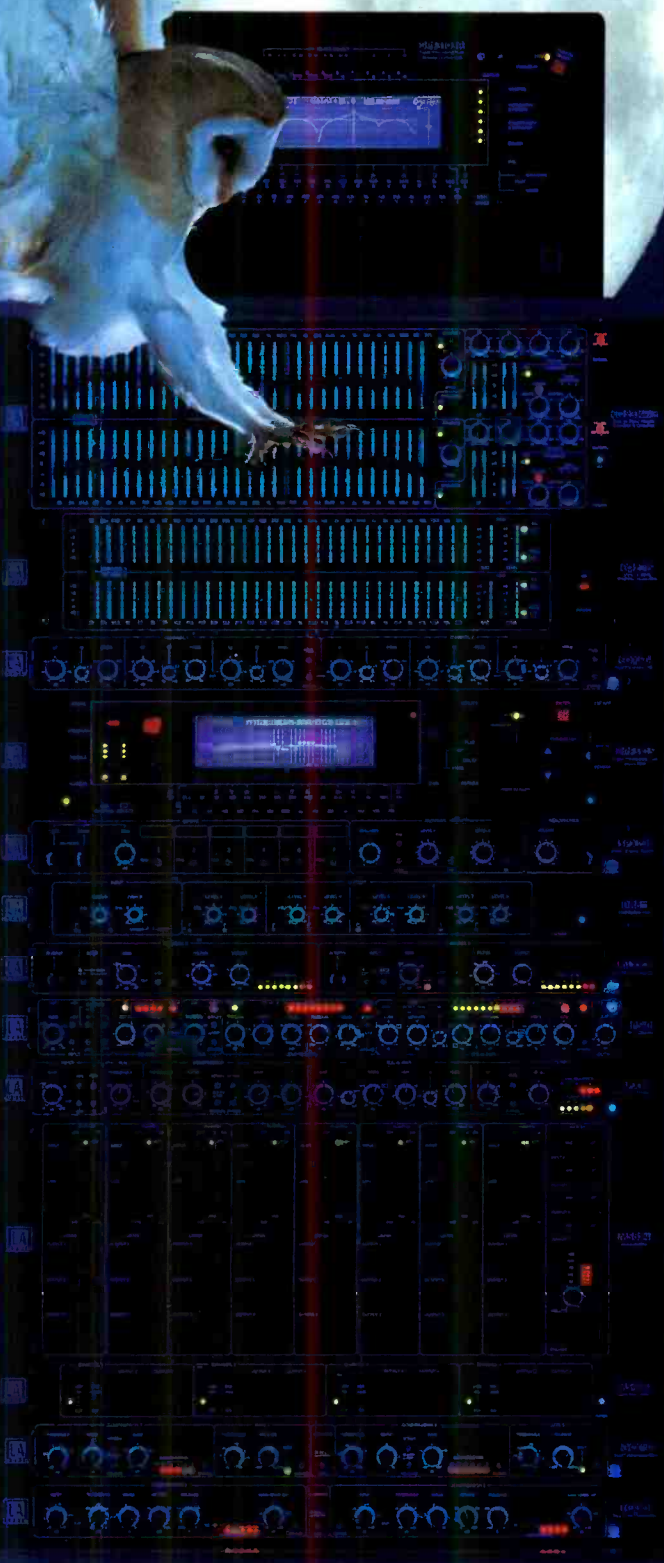
Rack viewed under UV illumination

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LAB GRUPPEN 300: THD vs. POWER OUTPUT re. 20kHz/8ohm (LOG SCALE, LEFT & RIGHT CHANNELS)

THE QC SUITE © MILLER AUDIO RESEARCH
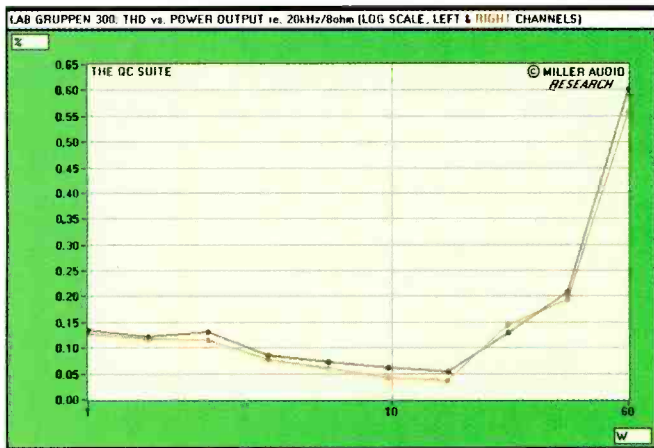
Fig.3

< from typically 0.05% to 0.6% by 60W/8Ω. This is compounded by Gruppen's VHF limiter that holds the single-channel output to 105W/8Ω at 15kHz, progressing to just 60W/8Ω at 20kHz (see Fig.4). Autograph Sales, the distribution and technical support arm of Lab Gruppen in the UK, was inclined to think that the VHF limit was somewhat higher than 20kHz which may indeed be the case in its costlier, higher power designs. It is difficult to escape the in-band limiting of the little Lab 300, however, which also shows a 0.5dB loss in *power* output at 20Hz compared to the healthy 110W available through the mid range.

The practical bandlimiting of this amplifier is equally evident from its frequency response (Fig.5) which, sensibly in my view, is attenuated beyond 30kHz. This may fly in the face of recent developments like SA-CD and DVD-A which offer the potential for extended, 100kHz bandwidths, but there seems little point in aggravating metal-domed HF units with the ultrasonic output of current CD sources. >

Power Amplifier: Lab Gruppen 300
(Rated Spec. in brackets where given):

	20Hz	1kHz	20kHz
Max Continuous Power Output,			
1% THD into 8ohm (one channel)	120W	130W	60W*
1% THD into 8ohm (two channels)	100W	110W (100W)	60W*
1% THD into 4ohm (two channels)	130W	160W (150W)	110W*
Frequency Response @ 0dBW	-0.3dB	0.0dB	-0.7dB
Dynamic Headroom (IHF)		+0.6dB (150W)	
Maximum Current (10msec, 1% THD)		18.0A	
Output Impedance		0.071ohm (0.03ohm)	
Damping Factor		112.7 (>250)	

Unbalanced Input	
Stereo Separation (1kHz)	98dB (90dB)
(20kHz)	70dB
Channel Balance, 1kHz (0dBW)	0.02dB
Total Harmonic Distortion	
(0dBW, 1kHz/20kHz)	-91dB / -57dB (-62dB)
(2/3 power, 1kHz/20kHz)	-83dB / -44dB (-62dB)
CCIR Intermodulation Distortion	
(0dBW)	-94dB
(2/3 power)	-90dB
Noise (A wtd, re. 0dBW)	85.5dB
(re. 2/3 power)	103.2dB
Residual noise (unwtd)	-76.0dBV
Input Sensitivity (for 0dBW)	65.2mV
(for full output)	749mV (775mV)
Input loading	20kohm
DC offset, left/right	-4mV / -1mV
Serial Number	991-136
Retail Price	£695 (ex-VAT)

< The full benefit of future digital sources will only be realised by amplifiers whose compensation extends at least two octaves beyond 20kHz and by loudspeakers free of sharp VHF break-up modes. Otherwise, VHF intermodulation will only cause potentially audible distortion to fold back into the audio range. If it were not for the intended bandlimiting, this fate might well have befallen the Lab 300 where, for example, at a steady 10W/8Ω (where the lower voltage rail is still firmly in command) there remains a ten-fold increase in THD over its final decade (see Fig.6).

The upshot is, Lab Gruppen's tiny 300 will never set the world alight for either its technical or subjective performance and yet, from the standpoint of both its economy and likely reliability, it's difficult to argue against the engineering compromises contained within. A pocket-sized powerhouse that runs cool in the hottest of close-field monitoring studios and the choice for any engineer ruled by his head rather than his heart. Seemingly bomb-proof and good value at just £695. ■
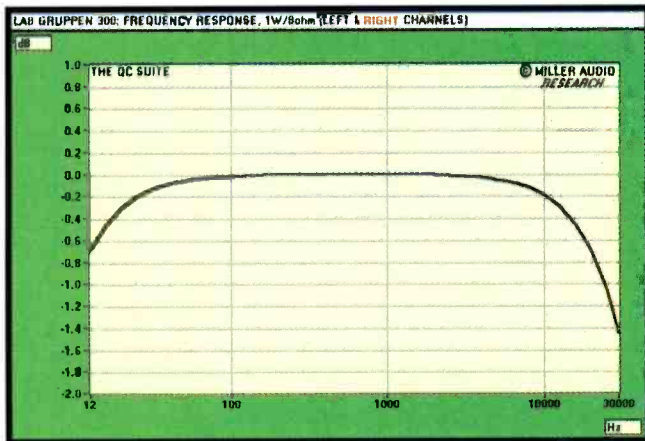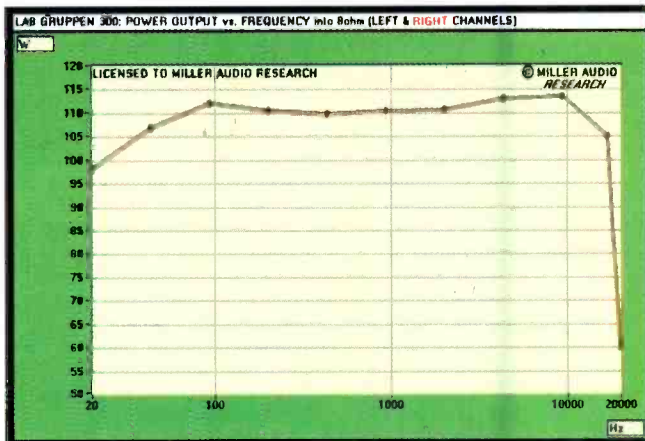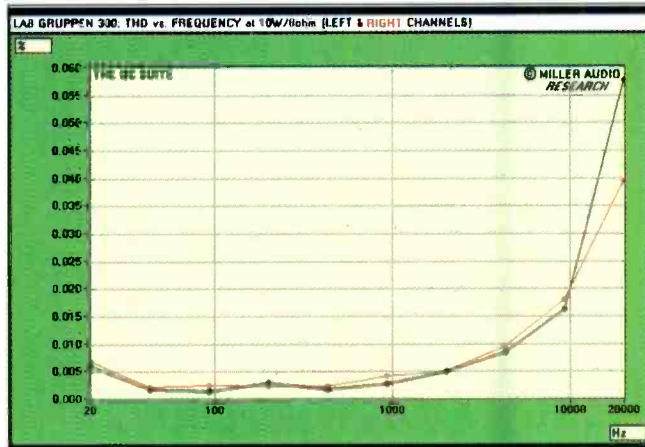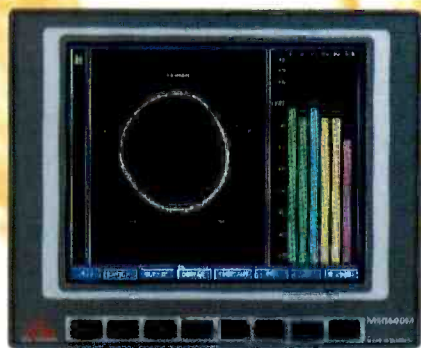

Fig.4

Fig.5

Fig.6

NEW TECHNOLOGIES

High power power

Two balanced power rack systems from Equi=Tech have been designed to power the needs of larger sound and video applications. The new Models ET7.5R and ET10R are large capacity systems designed to furnish continuous power for systems that require between 60A and 85A. Rack system features include a magnetically shielded toroid isolation transformer, surge

protection, overcurrent circuit protection, and GFCI protected power distribution. For 3-phase location power, the company offers Cube Series systems which employ 3 phases of balanced power for 120, 185 and 250A.
Net: www.equitech.com.

APT telecom codecs

APT has launched an apt-X-based audio codec family targeted at the telecoms market. Complementing advances, such as DSL, ISDN, E1, T1, Fractional E1 and Fractional T1, the codecs offer a new range of network audio solutions for broadcast circuits which claim savings in network capacity, improved network efficiencies and reduced network costs.
APT, UK. Tel: +44 28 9037 1110.

Maselec mic pres

The MMA4 is a 4-channel mic preamp in the Prism Sound Maselec Master Series which already includes the MEA2 stereo EQ and MLA2 stereo compressor. Used with the Prism Sound AD-2 24-96 A–D convertor the company claims it represents 'the ultimate microphone recording path.'
Prism Sound, UK. Tel: +44 1223 424988.

V multiformat upgrades

MAD Labs' CP8 and CP4 upgrades for the AMS Neve V series consoles are multiformat centre-section replacement packages that provide sonic and functional upgrades. The CP8 includes advanced signal routeing, switching and monitoring functions required for post, broadcast and music studios working on 5.1 projects. Features include bus and monitor mode buttons, status buttons, master meters and selector buttons, monitoring level controls, talkback facilities, studio loudspeaker controls, monitor select buttons, user-defined trims, paddles and tape control, fold-down mixer and a dialogue and EFX monitor path. The 4-bus CP4 is a stripped-down version. Depending on the desk-frame size and other features, the upgrade package starts at $65,000 US. Similar packages are expected for the SSL 4000 series.
MAD Labs, US. Tel: +1 818 353 6199.

LTO Ultrium cartridge

EMTEC Magnetics is developing a high-capacity Ultrium cartridge for use in the

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Helios EQ1 Lunchbox

Resurrected and patiently brought into line with modern expectation, the EQ1 is history in the remaking. **George Shilling** takes notes

A BRIEF HISTORY LESSON: the late Dick Swettenham was a maintenance engineer at Olympic Studios in its late-sixties heyday. In the early seventies he built a rather wonderful desk for Olympic and subsequently a number of Olympic clients' private studios. When Island Records wanted Swettenham to build a desk for Basing Street Studios legend has it that his Olympic employers were none too happy, so he left and went into business with the Helios name. As time went by, gear fanatic and music producer Tony Arnold started to buy old Helios consoles and acquired seven of them for export to the US. With a circa-1973 desk built for Eric Clapton as his reference, Arnold set about replicating Helios modules carefully including all the faults of the original to begin with. In the course of his research he found that the original desks' 20dB pad used a centre-tap from the Lustraphone input transformer. These sounded better than later Helios desks, which used Beyer transformers with resistors rather than a centre-tap for the pad. Arnold approached Sowter to copy the Lustraphone transformers. He then built a console for Elvis Costello and Chris Difford's Helioscentric Studios and put in 20 original Helios modules and 18 new modules. Over a period of over two years incremental modifications to correct problems with the original modules were carried out on the new ones. Each time these were carefully compared to the originals to make sure none of the modifications had any detrimental effects. Cyril Jones of Raindirk Fame helped with some aspects of problem solving. Having acquired the Helios name, Arnold set about perfecting the EQ1 modules we have here.

The originals' unreliability was always a bugbear for Arnold, so the new units are built to military specifications by CLI who among other things makes tyre-warmers for Formula One racing teams. The front panels are dipped in paint and the discrete circuitry is bulletproof. Modules from subsequent production runs will feature phantom power (which is already included in the rack version).

The Lunchbox comprises a flightcase with two vertically mounted Input-EQ modules alongside a panel with six bantam sockets for mic and line inputs and outputs—very convenient if you encounter bantam patchbays, a nuisance when you occasionally do not—and an IEC mains socket. There are no LEDs or lights anywhere. And with everything on the front, there is no need to open the back of the box.

The Input section features a 3-position toggle switch for mic-padded mic-line input and a stepped control for gain. I had a couple of the original modules in a Lunchbox for comparison: the old ones sometimes sounded slightly smoother. This is probably due to the new ones' extended high-frequency response, which if audible, is not necessarily

preferable to my ears. But any difference is extremely subtle. One vast improvement over the originals is in the build quality and reliability—crackly knobs are understandable on something almost 30 years old, but the originals were notoriously unreliable.

The stepped gain fixed 10kHz high frequency band is gentle and open, wonderfully enhancing clarity without introducing harshness. It goes in 2dB steps from -10dB to +10dB, although I have an inkling which way it will generally be turned. The Mid band comprises a selection of eight switched frequencies from 0.7Hz–6kHz. A toggle switches boost or cut, with an uncalibrated pot giving a roughly 10dB range. Gain for the Low frequency band is also controlled by uncalibrated rotary pot giving a maximum boost of approximately 10dB at 30dB, 80dB, 120dB or 240dB. These are 'boost only' frequencies. The frequency selector becomes an attenuation switch below zero, with cuts in 3dB steps down to -15dB at a fixed 75Hz shelf. I thought I might miss the ability to cut low-mid frequencies, being something of a fan of subtractive EQ, but with the Helios, boosting always sounds so good that I quickly adapted to this way of working. One Helios aficionado I spoke to testified to the glory of 'F-BAAF' (Full Boost At All Frequencies) on the Helios desk he had used. On the low frequency gain pot there is a click at the off-extreme. The originals had a low-frequency boost of 1dB even when the EQ was bypassed. For purists this has been retained, but the click position gives you the option of switching it off. A high-pass filter gives 15dB/octave cut at 40Hz or 80Hz, which is bypassed if the EQ is bypassed. A proper phase-switch circuit is incorporated, the original design for this being something of a bodge.

By modern standards, the features and control are limited. But here the sound is everything. You know you're in heaven when just about any setting sounds good, not just one compromise setting, which can be the case with many desk EQs. I recorded and processed all sorts of sources and the Helios always imparted a wonderful openness and clarity. If I were to own just one input module the EQ1 would certainly be near the top of the list.

A giant from a previous era returns in style, and in a portable package to take wherever you go. ■
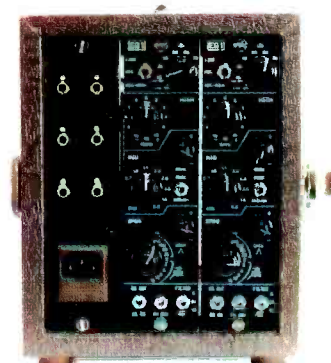
dbx 386

Adding its own mark of quality to the choice of combined mic pres and convertors, dbx 386 catches **Dave Foister** ear

THE MUST-HAVE OUTBOARD in the analogue studio is a pair of posh microphone preamps. The must-have outboard, increasingly, in the digital studio is a pair of posh microphone preamps with 24-96 digital output. There are few digital desks that can't be improved by the addition of a couple of high-grade input paths, and few projects that can't benefit from good mic

pres straight to digital multitrack. So obvious is this that every manufacturer has to have one, and moreover has to stamp its own personality on it. The latest is dbx with its 386.

It's two or three years since I first reviewed a valve microphone preamplifier with a digital output, and however bizarre a combination that might have appeared then, it's now almost commonplace. The valve thing has settled down, with its advantages in certain areas generally accepted, and two of those areas are microphones and the circuits to amplify their signals into the rest of the chain. Thus dbx, maestro of the solid-state analogue dynamics processor, brings us a pair of valve pres with a comprehensive range of digital

output formats, and very nice it is too.

The 386 is a 1U-high box that manages to pack in a tremendous amount of user-control thanks to clever use of indicators. Its front panel is not used quite as efficiently as it might have been, because even now, when we're all a bit blasé about the valve stuff, dbx has been unable to resist incorporating grilles to show off the glowing valves within. This scarcely matters, as the rest of the functions rely heavily and imaginatively on multi-coloured LEDs to show us what they're doing without hogging too much space.

The basic preamp functions are precisely what you would expect. Each channel has the expected pair of inputs, microphones on XLR and line on TRS jack, a TRS insert point, plus the now-familiar sight of a 2-pole jack on the front for direct connection of an instrument. Analogue outputs are available on XLR and TRS jack, and in between comes a useful and typically simple set of controls. The only rotary ones are for gains; the input gain is actually referred to as Drive as it allows for creative use of the valve distortion characteristic if

NEW TECHNOLOGIES

Bulk CDR80s

HHB has launched the CDR80 Silver Bulk. The 80-minute CD-R is unbranded and supplied in 50-disc 'cake packs' for easy handling and storage compared to

CDR80II BULK 50 PACK

conventional, shrink-wrapped discs. The discs comply with Orange Book specifications and cost the same as HHB's 74-minute CDR74 Silver Bulk discs. The unbranded surface allows for text and graphics.
HHB, UK.Tel +44 20 8962 5000.

CD-R Cliptray

An addition to its CD-R range, EMTEC's Cliptray is more compact and robust than the conventional jewel case. It is said to be useful for archiving CD-Rs as the black plastic case is hole-punched so it can be filed.
EMTEC, Germany. Tel: +49 621 59 20 406.

desired. There are two output level controls on each channel, allowing the gains to the analogue and digital output paths to be set independently. This is perhaps particularly useful in a preamp, allowing the digital signal to hit full scale without the slight risk that its analogue equivalent might overload subsequent equipment. This is still an occasional problem with some kit, particularly where the calibration has been set such that 0vu corresponds to -18dBFS. This means that digital full scale delivers +22dBm, which might be pushing it a bit.

Everything else is on push-button switches, and the clever part is the illumination that shows how they are set. Each switch is of the small rectangular backlit type, and use is made of bicolour and tricolour LEDs behind them to indicate the various settings. For the preamp sections themselves, the functions are simply toggled on and off, shown by the switch being lit or not. There are the expected four switches, controlling phantom power, a 20dB pad, phase invert and a low-cut filter, fixed at 75Hz and 12dB per octave. A separate switch selects the Line input. Each channel has its own horizontal LED meter, with a button to decide whether it will show analogue or digital output levels.

The A–D convertor section also holds few surprises in terms of function. Conversion is 24-bit, and dithering down to 20 or 16 bits is provided. Two dither types are available—TPDF and SNR2—and noise shaping can be set to one of two curves, or turned off. The two shapes are essentially similar, following the

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The valve thing has settled down now, with its advantages in certain areas generally accepted

expected psychoacoustic lines, with the second being a more pronounced version of the first. The sample rate can be 44.1kHz, 48kHz, or double these, and this is where a tricolour LED shows which is chosen. The only disadvantage of the indication method is the necessity for very small type on the front panel to show what the various colours mean. The output format can be SPDIF or AES-EBU, and the chosen format appears on both output connectors—it can't do both simultaneously. There are word clock in and out BNCs on the back, and the sample rate LED shows what frequency it's locked to.

The conversion uses dbx's proprietary Type-IV processing to maximise the use of the convertors' dynamic range. In essence this is gentle soft-knee compression of the top few dB of the analogue input signal, but the fact that half the manual is devoted to a white paper on it suggests it's not quite that simple.

The preamps themselves are good examples of modern valve circuits—very clean, very quiet and smoothly musical. Coupled with dbx's now-established digital expertise this makes the 386 a powerful and useful combination of technologies, with the flexibility to slot into a wide range of applications. ■

Stage Tec Reference Mastering A–D

High end, high quality, **Dave Foister** has high hopes for Stage Tec's flagship conversion package

WHATEVER BLACK ARTS APPLY in the design of some types of equipment, there will always remain areas where good science and engineering are the only things that will get you what you want. Perhaps the best example is conversion between analogue and digital domains, where the technology belongs in a

laboratory and there is no room for fairy dust. This can make convertors quite dull objects, but no doubt the designers would take that as a compliment.

But the Stage Tec Reference Mastering A–D manages to rise above the obvious, principally because of its scope. Here we have a big modular rack system that when full provides 24 channels of A–D conversion, and when filled with the appropriate options has microphone preamps on the front of all 24. This gives it a hugely impressive front panel, with rugged practical and cosmetic touches and a carefully designed, simple control interface. The review rack was full of the complete modules, giving me 24 mic amps delivering their signals solely on 12 AES-EBU outputs.

The complement of controls is surprisingly small: one rotary encoder on each channel, turning out to be a push-button as well for function selection. Along with it in a small panel for each channel is an alphanumeric display and an LED meter; no ordinary meter this, but an arc-shaped array of small lights offering unusually high precision. If the input modules have only line inputs, there is nothing to adjust, with this meter showing the level present in the channel. With the more elaborate microphone modules, there are the expected functions to adjust individually on each.

Although each channel has its own control, the entire front panel is globally switched to each function, which makes for much greater ease of use than having them all separate. The neat trick is that the current function is selected from any one of the 24 controls; pushing any knob will cycle through the parameters after which any control is live to adjust that parameter for its channel. In line with this, all the associated displays show the currently selected parameter value for their channels.

Gain is digitally adjusted in 1dB increments that are entirely click free, with a maximum

gain of 70dB. The LF filter is labelled EQU, which suggests more than it actually does; in fact it offers a useful selection of cut-off frequencies up to 80Hz but no equalisation beyond that. Phantom power and phase are simple toggles.

That's it for the preamp controls; and, as it turns out, for any controls at all. The very high-end specification of the analogue path is matched by an advanced convertor stage using Stage Tec's proprietary True-Match sigma-delta system. The specs claim 28-bit resolution at 64x oversampling, which will no doubt raise a few eyebrows; this signal can be transmitted intact via AES-EBU or dithered to 24 bits on a channel by channel basis. After that it's up to you what you do with it, as there is no facility for reducing the wordlength below 24 bits, elegantly or otherwise.

Conversely, and surprisingly in this day and age, the sample rates only cover the range from 32kHz to 48kHz. Perhaps current mastering requirements are met by this selection, but I doubt if that will be the case for long. I was also left scratching my head for quite some time as to how to change the sample rate. The unit fired up at 48kHz on its internal clock, and my initial impression that there was no onboard way of changing this turned out to be correct. In fact the unit locks to incoming clocks on its rear BNC, and when this is removed its internal clock stays locked to that frequency until something else gets connected. House sync would mean this would not be an issue.

Output formats are determined by the cards fitted, and encompass AES-EBU on XLRs and SPDIF on either phonos or optical terminals. A fourth option gives eight channels on a 15-pin D, but there's no facility for delivering ADAT or TDIF signals.

On the same card as the BNC word clock in and out is a 9-pin D for hooking up an external PC. This gives almost no extra facilities, other than a more comprehensive graphic display of system status and control of everything from the computer; there are however one or two extra bits like an on-board analogue signal generator that can only be run from the PC.

To go back to the beginning, the Stage Tec system looks, behaves and is specified like a laboratory instrument with attitude. There can't be many more elegant ways of getting microphone signals on to a 24-bit recording medium or into a digital mixer, although there are few systems that can interface directly without adding optional cards. This is a small additional price to pay for access to this much class; barring the 96kHz issue, you'd be hard pressed to find this quality in either analogue or digital domains in anything like this little space. Look for more output cards and this is the way forward. ∎

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ServerSound modules
mSoft's ServerSound is a cross-platform hard-disk-based audio retrieval system for use with multiple types of DAW. The system consists of an NT server with hard-disk array and local network, and can convert between multiple file formats while downloading to a client workstation. Its browser technology makes it accessible via a local network or remotely over the Internet or T1. Pro/Spotter is an automated spotting software module available in versions for those working in short or long format productions. Pro/Master module provides a fast method of automating the process of digitising raw audio from CD or DAT, splitting it into individual track and index and finally linking each bit of audio to a database. Not limited to audio, ServerSound can also be used as a centralised library for video, animation, or stills.
mSoft, US. Tel: +1 818 716 7081.

PreSonus stereo mic pres
The PreSonus Blue Tube stereo microphone preamplifier with instrument inputs is now shipping. Designed to be a companion to the Blue Max stereo smart compressor the Blue Tube features two high performance mic preamps with a Sovtek

12AX7 tube and dual servo gain stages (no capacitors). Front-panel controls consist of tube drive, gain, 48V phantom power, 20dB pad and polarity reversal. Each channel of the Blue Tube features a Neutrik combo connector on the front panel that accepts microphones or instrument direct inputs. Output connectors are XLR and TRS with two 8-segment LED meters.
PreSonus, US. Tel: +1 504 344 7887.

All valve mastering EQ
Fairman's all-tube TMEQ tube master equaliser is a 6-band stereo device in a 6-U high rackmount. It employs 22 valves in 12 separate frequency controlled amp sections for phase linearity across all frequencies. It provides overlapping sections for low cut and low boost, and high cut and high boost. The low-mid and high mid offer boost or cut with Q switchable for 18dB, 12dB or 6dB per octave. Frequency and attenuators are detented switches and each of the 12 separate sections can be bypassed individually. Input and output gain switches are adjustable from -6dB to +6dB in 0.75dB increments while a green LED for each channel indicates true internal 0 VU.
ATR, US. Tel: +1 717 852 7700.

Portable recordable
The RCX220 portable digital audio recorder also doubles as the engine for a PC-based audio workstation. Designed jointly by Nagra and Digigram the stereo

Beyerdynamic DT150

Attempting to improve on a well-established and popular headphone model is a tall order. **Zenon Schoepe** admits it has come to pass

FEW ITEMS OF AUDIO EQUIPMENT can make a stronger claim to the tag of industry standard than Beyerdynamic's long lived DT100 headphone. The appeal can be attributed to a peculiar combination of durability, dependability, serviceability and flexibility, although the sonic performance, while solid and consistent, has never been the major selling point. People love DT100s because they immediately know where they are with them, because they know they can fix them because they were made to be fixed, because they come in a wide variety of configurations that include mic boom options and because no other headphone can be clamped, as only a DT100 can clamp, to the head of a drummer and feed him all the level he wants before his eyeballs explode. I still remember an instance where an engineer was holding a pair (DT100s not drummer's eyeballs) when the playback was rolled and he dropped them in fright as they sprang in to life. Bad stuff in these SPL conscious days, but then the customer is always right.

You'll see the unmistakable items on the heads of location recordists, on FOH desks, draped on the arms of chairs in film mix theatres, instructing camera men on the studio floor and in the field, on the table in the voice over booth and somewhere in every studio you visit.

These enduring qualities have made the DT100 a strong seller for decades but it's been a little known fact that Beyerdynamic addressed the model's single weakness of lacklustre sonic performance with the introduction of the DT150 some time ago. Here is a headphone that carries over the original's features list and appeal, but betters it and adds a far more distinguished reproduction circuit.

The range encompasses the standard DT150 headphone with multiconductor cable in straight or curly leads; DT180/DT190 with boom DM190 dynamic hypercardioid; DT181 /DT191 with EM191 electret condenser omni (can be used for wireless applications); and the DT182/DT192 that can also be run wireless, but with the EM192 electret condenser for high noise environments. The microphone equipped models can additionally be supplied as V.11 versions with integrated carbon equivalent matching preamps for use with TV cameras.

Visually the thing that differentiates the DT150 is the use of dark grey plastic on the ear piece outers—the moulds and other components look identical, although the newer model does have a less bulky headband cushion. The feel is identical, but the specs start to tell a slightly different story with a claimed frequency response stated as 5Hz–30kHz compared to the DT100's 30Hz–20kHz with nominal SPL of 97dB compared to 94dB on the older design. Impedance is 250Ω compared to the DT100's 400Ω.

The reality is the the DT150 is a far more pleasant listening experience with a marked extended bass response and a smoothness to the remainder of the range with none of the stridency, and, if we're honest about it, lumpiness of the DT100 which while it aids intelligibility doesn't amount to a discerning occasion. It's unfatiguing performance that is a light-year and several decades removed from that of the original. A truer and better all round performer.

I do prefer to have the cord running to the left lug-hole for historical reasons rather than Beyer's arrangement, however the signal comes in on a cable of heavy-duty-guitar-lead proportions and one of the advantages of the multipin connector is that if you do inadvertently step on the lead while you stand up then the whole assembly pulls out quite cleanly without ripping the ear-piece off or your ear, or bending the connector if you haven't employed a grub-screw permanent fix. Reconnection in these circumstances is fast and positive.

I've never been a fan of the vice-like grip that the venerable DT100 exerts on the cranium, but I've now ascertained that the reason for this is that I have never actually managed to wear a set for any length of time —it's always been a case of being helped in to a pair, realising that they still sound a tad basic and pulling them off before the pressure builds up and the nose starts to run. Because the DT150 sounds so much better I persisted and found that the clamping actually doesn't feel that bad for extended periods thanks mostly to the plushness of the ear cushions.

There's an involving and engrossing quality to these headphones that open back models never achieve and, while they do get hot, isolation is exceptional which makes them ideal for voice-overs and vocals. They remain the definitive article for applications where operators need a high degree of cocooning from their surroundings. However, positioning is critical as the grip is strong enough to pull up the eyebrows in a permanently surprised look or serve as a cheaper alternative to a face-lift.

So there you have it, an industry standard has been improved on in every respect. Beyerdynamic continues to manufacture the DT100 for all the traditionalists, but for the record, and for their information, a better can does exist. I'm converted. I just hope that nobody thinks I'm using DT100s. They've taken ten years off me. ■

Contact
Beyerdynamic, Germany.
Tel: +49 71 31 6170.
UK: Tel: +44 1444 258258.
US: Tel: +1 516 293 3200.

digital recorder is rugged and immune to environmental extremes, but when connected to a PC by USB link it provides the DSP power of a Digigram PCX sound card allowing Digigram-powered operations to be performed without the need to purchase additional hardware. The device is based on Nagra's ARES-P technology and sports stereo balanced mic inputs, a headphones circuit, removable PC Flash card for up to 3¼ hours of stereo recording time with MPEG, automatic or manual level control, and play, forward and rewind controls. Operation centres around an LCD screen and power comes from batteries.
Digigram, France. Tel: +33 04 7652 4747.

MOTU goes 24/96

MOTU 1296 is a 2-U high computer-based hard-disk recording system for Mac and Windows that offers 12 simultaneous channels of 24-bit, 96kHz input and output on balanced XLR connectors. As with the company's current range of audio interfaces, the 1296 core system includes a PCI-324 audio card with three 'Audio Wire' connectors, allowing users to expand the system to three

1296 interfaces for up to 36 simultaneous channels of 24/96 input and output. The 1296 audio interface will also be available as an expansion I-O that can be mixed and matched with MOTU's range of audio interfaces, including the 2408 mkII, 1224, 24i and 308. In addition to analogue I-O, the 1296 includes stereo AES-EBU I-O, word clock I-O, and an extra 'AES' word input that allows the unit's AES-EBU section to independently resolve to an external word clock source.
MOTU, UK. Tel: +44 1767 313447.

Schoeps tubes

Schoeps is now offering its Colette tubes as a mechanical accessory for the CCM compact condenser microphone series. With the Colette tube an exchangeable microphone capsule was screwed onto the front end whereas now a CCM microphone takes its place. At the lower end there only a miniature Lemo connector is left instead of the CMC microphone amplifier that has to be screwed on here with the Colette version. There is also a version available that carries two microphones one above the other for MS recordings. The VMS 5-U high mic preamp with MS matrix is a successor to the VMS 02 IB and VMS 52 UB and features phantom powering and an input impedance of 10kΩ.
Schoeps, Germany. Tel: +49 721 943 200.

Tascam by TimeLine sync

TL-Sync is a stand-alone synchroniser

AKG C2000B

The latest variant on AKG's landmark SolidTube-fronted range is the C2000 B. **Dave Foister** finds it suits all tastes

ALL THE BIG MICROPHONE PEOPLE have had to respond quickly and positively to the influx of cheaper alternatives from various points around the globe. They've all done a good job, maintaining the status of their flagship models while bringing us scaled-down versions that compete well with the alternatives. Fundamentally the attributes that put them where they are are rarely compromised, making sure that when AKG brings us an entry-level version of an established model, it will be well-built and it will behave itself.

The C3000, with its obvious links to the 414, became quite a hit when launched for these very reasons. Now there's a new AKG style, begun by the SolidTube, and the new C2000B follows those lines. Just over a year ago I looked at the Solid-Tube's first relation, the C4000B, and as the name would suggest the 2000 is in many respects a scaled down 4000.

The first and most obvious respect is its size. The curved cylindrical shape is very similar, with its protruding connector shaft that mounts the microphone in its holder, and the champagne gold colour and close mesh grilles are the same, but the whole thing is about half the size. Where the SolidTube and 4000 appear to be designed to catch the eye and make a statement, the 2000 lends itself more to general use in the thick of it, allowing easier access and reduced visibility.

Its flexibility is also scaled down in that it is a simple single-diaphragm cardioid microphone where the 4000 has twin diaphragms and three polar patterns. It is not totally devoid of controls, however, as there is both a 10dB pad and a low cut filter. The filter is very unusual; where most turn over quite low and drop off steeply, this is 6dB per octave below 500Hz. This may in fact be better suited to dealing with proximity effect than the more familiar characteristic, although for removing rumble it may be a bit intrusive on the wanted spectrum.

Although it is small and relatively light, it is no lightweight in terms of its construction. The body material is ambiguous to the touch, and could almost be plastic; in fact it is all die-cast metal to ensure adequate screening as well as robustness. The diaphragm is plastic foil, gold sputtered on the front only to reduce the risk of internal shorts to the back electrode when subjected to very high SPLs. It claims to be able to handle 150dB with the pad in, almost dynamic territory. The capsule is shock mounted internally, and there is a built-in pop screen in addition to the optional foam windshield, which is the same as the one supplied with the C4000B. The accompanying literature makes it clear that AKG see a role for the C2000B on stage as well as in the studio.

The stand mount is also the same as that supplied with the C4000B, and as I recall thinking that was a little over the top, it's certainly more than adequate for the 2000. It is a full-blown suspension mount, with a clever rotating clamp that locks around the connector shaft at the microphone's base and engages with a small lip for added safety. A slight slackening of the clamp allows the microphone to be rotated, and the vertical position is locked with a typical AKG handle. So big is the mount that this little microphone almost disappears when nestling in it, and in fact it becomes a bit awkward to get at the switches.

This is more than compensated for by its manoeuvrability; poking a 4000 in amongst a section is tricky, but the 2000 is much easier even with its huge mount. And this is the kind of situation the microphone is going to be happiest in. It doesn't have that big fat present sound of the classic vocal mic, but it lacks nothing in clarity and ability to pick up the real sound of the instrument. At the same time there's nothing small or thin about the sound, which in many ways is closer to that of a big diaphragm than a small one—it looks to be around ³/₄ of an inch. It's an all-rounder in the best sense of the word; not the one that's left over when you've stuck your favourites up on the important things, but the one you know you can use anywhere with confidence. It's clean, quiet and natural, and its SPL and transient handling capabilities were well demonstrated on a percussion session I put it up on pretty close. It got quite a hammering (sonically, not physically, AKG please note) and never flinched.

AKG has always had a knack for covering all the bases; whatever type of microphone you want, studio or live, expensive or affordable, specialised or general-purpose, there's an AKG that will stand comparison with anybody else's offering. The C2000B continues that trend, and falls nicely into the slot that will get it used in the big studio as much as the back bedroom. It's a good response to the competition and should find wide acceptance. ■

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module that can work with DTRS, ADAT sync, SMPTE LTC, MTC, 9-pin, TimeLine's Lynx bus and MMC. The device also includes TimeLine's TL-Bus for sample accurate sync to the new Tascam by Time-Line MX24242 digital disk recorder both of which can be controlled by the RC24242 remote that by connection will also then be able to control the aforementioned transports. Clock sources include digital wordclock, superclock, video sync, AES-EBU and SPDIF and 4 low jitter digital clock outputs are provided.
Tascam, US. Tel: +1 323 726 0303.

SPL adds affordable Atmos
Area 5.1 is a miking and mixing system for 5.1 recordings and represents a streamlined Atmos 5.1 that accommodates the primary features in an affordable system. The Area 5.1 also allows the simultaneous recording of 5.1 and stereo. Area 5.1 includes five high precision mic preamps which can be coupled with motorised controls. Each preamp

delivers up to 70dB of gain with pad, phase reverse and phantom power together with a stereo downmix section. The Sub/LFE channel creates the sub signal from a selection of front, surround and centre microphone signals. Accessories offered include an INA5 mic array. The Stereo-Q is a 5-band parametric stereo equaliser based on Qure filters in proportional-Q design. The unit is equipped with three full parametric stereo-bands, a Sub-EQ, an Air-Band-EQ and a new 3D-Enhancer.
SPL, Germany. Tel: +49 2163 9834 14.

RTW's 5.1 metering
RTW's Surround Monitor 10800 is designed for use in surround sound and multichannel recording, postproduction and mastering. Based on a dual-screen design, with one screen being used as an 8-channel peak programme meter and the other as a 2-channel, 3-channel, or 5-channel audio vectorscope, the 10800 provides the information to achieve the optimum level and signal control. The 10800 is fully equipped with analogue and digital inputs and is able to display different domains on one screen. The eight PPM's are multi-standard types that can also calculate and display levels derived from the LCRSL/SR signals, as well as loudness. RTW claims that the audio vectorscope offers selectable grids for different surround modes such as 2/0, 3/1, 3/2 (5.1) and four phase meters for the LC, *CR, LR and SL/SR channel pairs. These additional phase meters are said to be especially helpful during surround sound recording in order to check the setting of multi microphone arrays or to check phase relations between the channels. Additional features of the 10800 are a real-time third-band

Soundscape EDL v2

Away from the entertaining aspects of workstations is a host of equally essential functions. **Rob James** checks the list

JUST AS THERE ARE GLAMOUROUS pieces of hardware and less exciting but essential black boxes, so it is with software. In the arena of sound-for-picture, no matter how well a system performs in the headline tasks of recording, editing and mixing, it will be useless for many applications if it cannot perform certain fundamental tasks. Some of these are immediately obvious such as synchronisation to picture and the requirement for a variety of

frame rates, others may be less so. Autoconforming is just such a function.

The need for autoconforming arises from the frequent use of NLV (nonlinear video) editing systems with 'guide' audio. Autoconforming, then, is taken to mean the entire process of semi-automated rerecording of the source audio into a workstation and arranging the resulting cues on the correct tracks and time code positions in conformity to the picture edit.

Soundscape has carved a secure niche in sound-for-picture and have offered an autoconforming package for some years.

There are several ways of presenting autoconforming to the operator. Some manufacturers integrate it into the main user-interface. Others use third-party packages for the job. Soundscape has chosen to separate autoconform into a stand-alone software application that addresses its standard hardware. The workstation interface does not become more cluttered and fewer compromises are necessary.

At the simple level autoconform has to convert an imported EDL into the native EDL format of the workstation and control the source machine to fill in the blanks. The essential information contained within an EDL is source and destination in and out times, source and destination tracks and the source reel number.

Machine control may be simple time-code chase or '9-pin' RS422 serial. This works well enough with straightforward projects where the source material has nice tidy continuous code. The fun begins where there is a requirement to merge lists or the source code is discontinuous. An alternative is to simply play entire source reels feeding time code into the EDL processor which records only the required sections.

Soundscape has obviously learned a great deal from experience with the first version of the Soundscape system and the new package contains a wealth of features which will save

a great deal of time and trouble to those who need them. Rushes recording is now supported—including automatic recording and identification of audio clips with broken code or time-of-day. Also, handles (extensions of audio beyond what the EDL specifies) and crossfades can be independently set for each reel. Multiple EDLs and Audio Projects can be contained in one EDL processor project which allows both guide and source audio to be conformed. The Scenes feature enables a project to be split into several Soundscape arrangements to allow several workstations to work on parts of the project. Conforming may be undertaken from existing sound files even across a network. Additionally, I would have liked to be able to use reel order lists and to be able to directly edit EDLs but neither of these are essential.

EDL source files must be 'A' mode—that is, in finished product order. As the processor loads the EDL it is sorted into source reel order. The original EDL is displayed on the right of the main screen. The left-hand side shows the EDL audio sources (reels) and or source files. Clicking on a source reel brings up a list of Takes to Record, in blue from the selected reel. Entries with errors are shown in red. Tracks and source reel mapping allows for lists containing mono events to be conformed as stereo or for where the source reel numbers have changed. Video edits can be converted into markers for swift location of video events.

As recording proceeds, takes will disappear from the Takes to Record list to be replaced by entries in green on the Recorded Takes list. When all the material has been recorded the final stage is to create the Soundscape arrangement. This will usually be created 'from source reels' however it is also possible to 'create from destination reels' where the audio has been recorded from the NLV workstation.

Manufacturers are always pulled in certain directions in part by client feedback, but also by alliances formed with manufacturers of other kit. To a casual observer of the NLV market, a couple of names may seem all pervasive, but there are several other serious players. So I am delighted to find support for Tektronix Lightworks and Discreet Logic/ D-Vision formats in addition to the 'standard' variations on CMX, GVG, Sony, Avid and OMF. (OMF composition import-export and audio import is also available for the Soundscape editor.)

With v2, Soundscape has rounded out the autoconform capabilities of its system to rival anything on offer from other manufacturers. All the expected functionality is present together with a generous helping of useful extras. Autoconforming may not be the most exciting of applications but this example makes the process as painless as possible if not actually pleasurable. ■

Contacts

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NEW TECHNOLOGIES

analyser and the AES-EBU status monitor. The latter is a helpful tool when digital interface problems occur. It allows checking of all bits and bytes of channel, user and audio data. The status monitor detects the real word length of faulty bits on a digital signal.
RTW, Germany. Tel: +49 221 7091 333.

Eela Audio's Anniversary

Eela Audio, celebrating its 25th anniversary this year, has the IDMX Integrated Digital Mixer and Xpoint system components. The system offers integration of audio, communication and machine control. The mixer can easily be configured for a broad variety of applications. The IDMX system components comprise the On Air console, the ICom panel, the Producer panel and the Presenter panel. Targeted at broadcast journalists and producers, Eela's Logos is a flexible audio mixer with incorporated matrix functionality. It provides extensive control of source machines and DAW editing software. More complex facilities such as telephone hybrids and ISDN codecs can be addressed from a phonebook or manually by the dial keypad. Via the Logos Windows Menu, complicated settings can be pre-configured. Eela reports interest in its Reportable S30 DSP PCMCIA recorder for ENG applications due to reduced prices for the flashcards, file-compatibility with PC automation systems and less maintenance than tape media according to the company.
Eela, The Netherlands. Tel: +31 40 251 0484.

CR-10S update

The popular BPM CR-10 condenser has been updated to CR-10S status with the addition of switches for PAD and Bass roll-off. The 1-inch gold sputtered 2µm diaphragm mic comes with a suspension, cable, windscreen and aluminum flight case for a retail price of US$299.
Net: www.bpm-studiotechnik.com

SPL launches channel strip

The new SPL Channel One is a complete channel strip based largely on processing elements already developed and proven by SPL within its existing range of separate processors. It incorporates a tube mic-line preamplifier, de-esser, compressor-limiter based on SPL's Double-VCA-Drive circuitry, noise gate and 3-band EQ section. The incorporated headphone monitor can be used to pass an individual mix directly to the singer-speaker without latency. To enable immediate digital recording, a 24-96 A–D/D–A convertor can be provided optionally. Further options are sound finishing Lundahl transformers for the inputs and outputs.
SPL, Germany. Tel: +49 21 63 8761.

Mackie HR824

Studio Sound's 'bench test' loudspeaker reviews continue with the HR824. **Keith Holland** reports

THE MACKIE HR284 is a 2-way, active loudspeaker comprising a 220mm mineral-filled polypropylene cone woofer and a 25mm aluminium dome tweeter mounted on a shallow, die-cast axisymmetric horn. The electronic crossover, protection circuitry and power amplifiers are all built-in. The cabinet has overall dimensions of 254mm wide by 400mm high by 310mm deep, and is fitted with a large, elliptical, rear-firing passive radiator with an aluminium honeycomb piston. The front edges of the cabinet are finished with a radius to reduce diffraction effects. Mackie specify the power amplifier ratings as 150W for the woofer and 100W for the tweeter giving a maximum peak SPL of 120dB and a maximum short-term SPL of 111dB at 1m for a loudspeaker pair. The crossover is described as a modified Linkwitz-Riley having 4th-order slopes and a crossover frequency of 2kHz. The rear panel has controls for input sensitivity, acoustic space, low-frequency cut-off and high-frequency shelving. The input sensitivity is continuously variable, the acoustic space control is switchable between quarter-, half- and whole-space giving -4dB, -2dB and 0dB low-frequency shelving respectively, the low-frequency cut-off is switchable between 80Hz, 47Hz and 37Hz and the high-frequency shelving is switchable between -2dB, 0dB and +2dB. This review was conducted with the switches set to whole-space, 37Hz and 0dB. The mains and input signal sockets exit downwards at the bottom of the

rear panel allowing the loudspeaker to be placed with its back against a wall if desired. The cabinet assembly has an overall weight of 15.2kg.

Fig.1 shows the on-axis frequency response and harmonic distortion for the Mackie HR824. The response is kept within ±2.5dB from 38Hz to 20kHz; a remarkable result, and among the best of the loudspeakers tested to date. The low-frequency roll-off is 6th-order with -10dB at about 32Hz which is very low for a loudspeaker of this size. Harmonic distortion is also very low, having a maximum in-band level of -46dB (0.5%) at 50Hz. There is, however, evidence of a sharp peak in distortion at 300Hz indicative of a rattle. The horizontal and vertical off-axis responses are shown in Fig.5 and Fig.6 respectively. The horizontal directivity exhibits a gradual narrowing with increasing frequency from about 500Hz upwards demonstrating good matching between the drivers. The interference notch at the crossover frequency of 2kHz, due to the spacing of the drivers, can clearly be seen in the vertical off-axis results. Time domain performance is presented in Fig.1, Fig.2, Fig.7 and Fig.6 which show the acoustic source position, power cepstrum, step response (Fig.3) and waterfall plot respectively. The movement of the acoustic source position to about 3m behind the loudspeaker at low frequencies is due to the 6th-order roll-off, and is typical for loudspeakers of this type. The power cepstrum plot (Fig.4) shows very little evidence of reflections indicating good control of diffraction and horn termination. The step response is fairly accurate with the high frequencies leading the mid frequencies by about 0.5ms; the woofer and tweeter are in phase with each other. The waterfall plot (Fig.7) shows little evidence of ringing, but the effect of the rapid low-frequency cut-off is clear. Overall, the Mackie HR824 is a remarkable loud-

speaker. The measured performance is the equal of any loudspeaker tested to date and it demonstrates no particular weaknesses. The on-axis frequency response is commendably flat, the off-axis response is very well controlled and the harmonic distortion is very low. The rapid low-frequency roll-off may offer some cause for concern, but this is perhaps a worthwhile compromise given the impressive low frequency performance for a loudspeaker of this size. The provision of acoustic space selection means that this loudspeaker should work well under a wide variety of acoustic conditions. ■

Fig.5: Horizontal Directivity
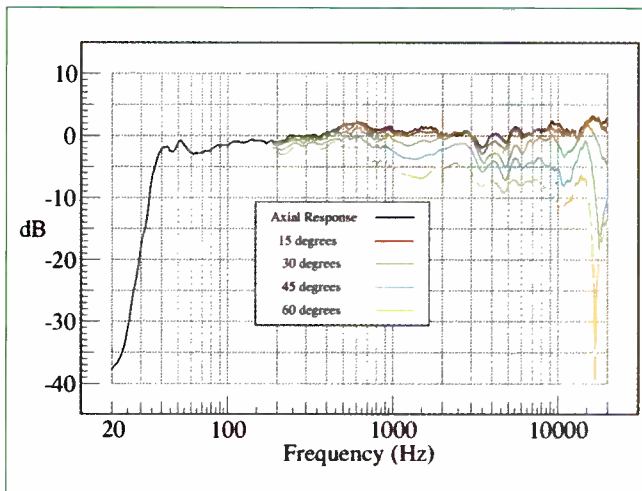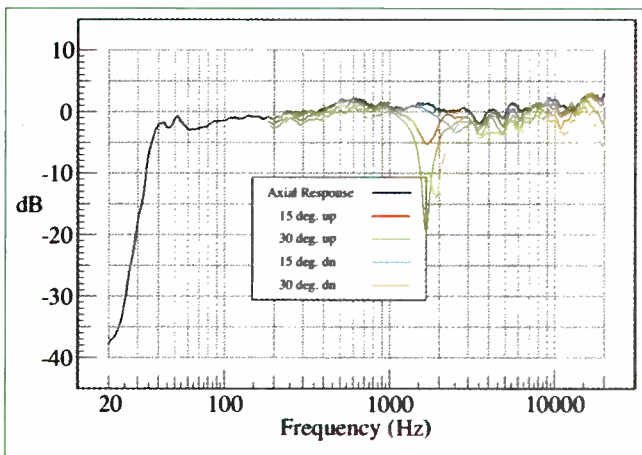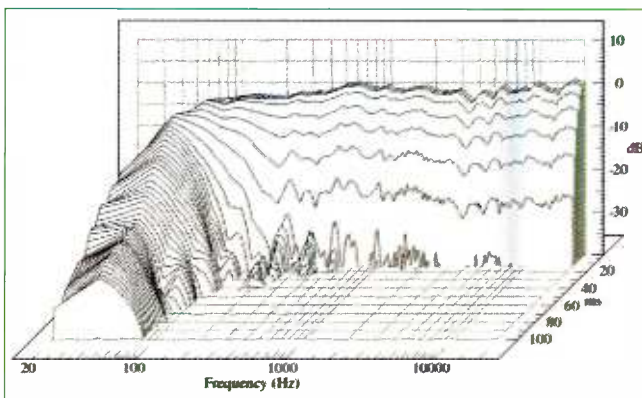

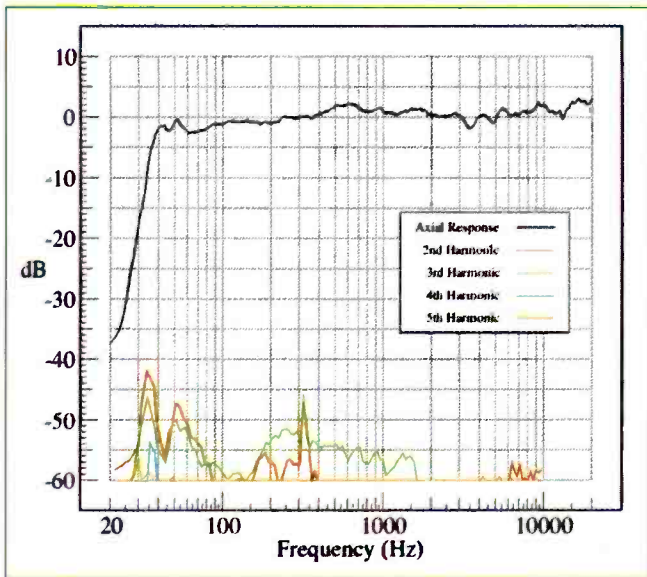**Fig.6: Vertical Directivity**

Fig.7: Waterfall

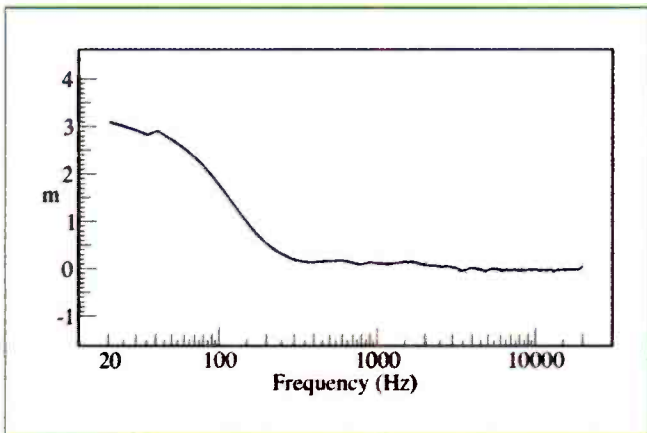Fig 1: On-axis Frequency Response and Distortion
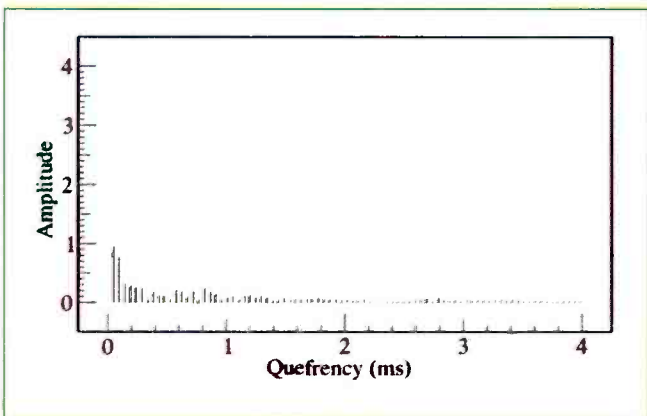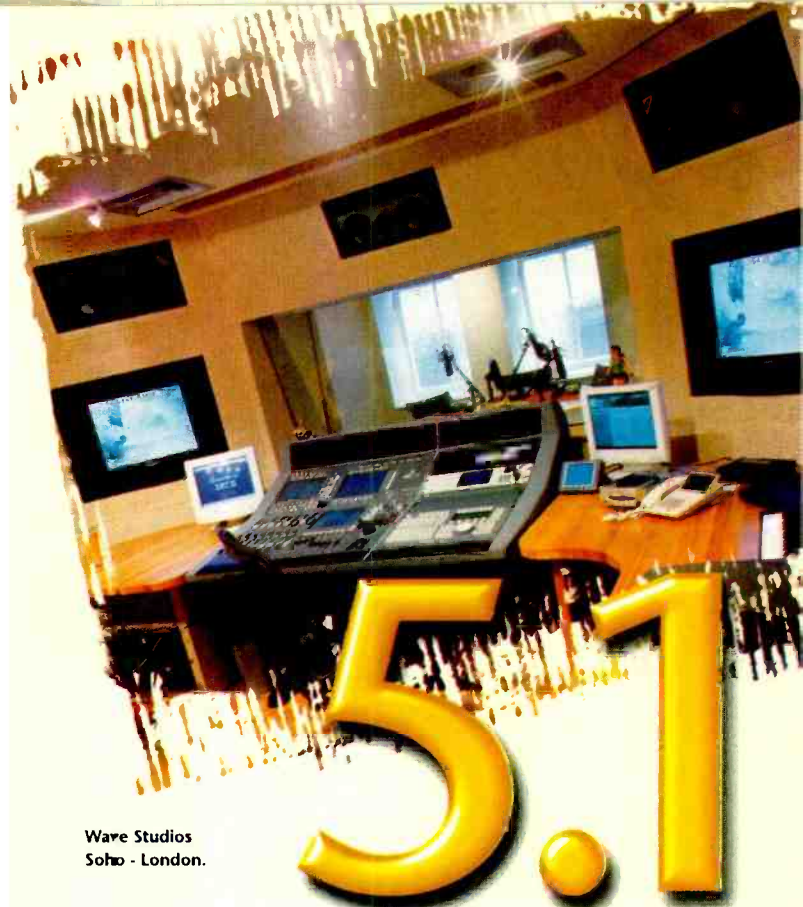

Fig.2: Acoustic Source
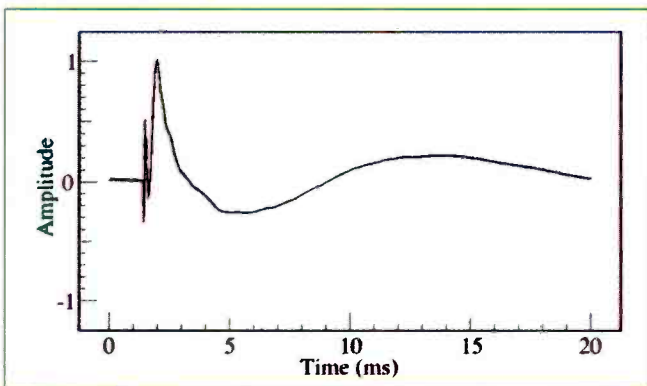
Fig.3: Step Response

Fig.4: Power Cepstrum

Studio Sound   **www.prostudio.com/studiosound**

Reelin' in the years

Producer Gary Katz worked on all seven Steely Dan albums but these days he's looking out for the hottest hip hop, as he explains to **Simon Croft**

THE PHONE RANG. 'You like Steely Dan, don't you?' Oh, the power of the rhetorical question. Like them? Like them! No, I bought all those albums to hide an embarrassing gap in my record collection somewhere between the letters R and T. Actually, all I said was 'Yeah, why?' and so I found myself heading for north London to meet producer Gary Katz.

Since the last Steely Dan album was recorded in 1980, I was curious to know what Katz had been doing in the last two decades. His later credits include album tracks with Diana Ross, *Civilised Man* with Joe Cocker, *Zazu* with Rosie Vela and Laura Nyro's final album *Walk the Dog and Light the Night*—as well as numerous 'Dan-related' projects, including Donald Fagen's *The Nightfly* album. Among more recent Katz credits were the 1996 *Legs & Arms* album with Lode—otherwise known as Inara George, daughter of Little Feat's late Lowell. Katz' work of late has also included artists I hadn't come across; Raw Stylus (1995), Repercussions (1995) and Digital Underground (1998). As I was soon to find out, Katz is a man of quite eclectic tastes.

Although the early Steely Dan albums were recorded in California, Katz is actually from New York, where he still lives. It was a fortuitous start for a would-be producer. 'I had some friends called Jay and the Americans when I was growing up and they were having a lot of hits in the States,' he recalls. 'They were in the studio, I was about 18. I was always listening to music, it was a passion of mine. So when they went to the studio I went along with them.

'They were being produced by Leiber & Stoller, who were my favourite producers at the time and made all my favourite records with The Drifters and The Coasters. I watched them produce and I saw that one of them was really good and musically trained. And one of them really knew nothing about music—and basically worked with his ear. I said, "I can do that". That's really what kicked me off,' says Katz.

Luckily for the aspiring Katz, Leiber & Stoller were happy enough to have him around. 'They befriended me and let me come to the studio to watch them work,' Katz acknowledges. 'That was a great thrill.'

Music is clearly very important to Katz and he later describes himself as Steely Dan's 'biggest fan'. He certainly shows little interest in the technicalities of recording. 'I'm not an engineer-producer. I don't want to be involved with that,' he affirms. 'I work with people who are really good and I try to concentrate on the music with the artist. I try and bring the most out of them performance-wise.

'Because most of the artists I work with write their own material, I know a good deal about that insulated feeling. I can take a step back and have my suggestions, help them to structure the tunes and realise their full potential.'

But isn't he to some extent a project manager, contracted by the record company to deliver on budget? 'I don't look at it that way,' Katz responds. 'No—I'm working for the artist, always. I never consider myself as working for the company. I mean, it's not like I don't have a sense of responsibility with regard to budgets and whatever, but in the actual process, I'm always working for the artist.'

Although Katz feels that this philosophy shouldn't create 'any sense of contradiction' for the record company, he becomes uncharacteristically animated if you ask him to what extent to accommodate the artist's way of working.

'You're at the mercy of the artist,' says Katz. 'What are you going to do, say to the artist, "What do you mean you don't want to sing at 2 o'clock? We're gonna sing at 2 o'clock." So you work at the mercy of the artist's likes and dislikes.'

For the interview, I brought along eight Steely Dan albums, which included one I doubted Katz had even seen before. Entitled *Sun Mountain*, it features Donald Fagen and Walter Becker ploughing through some of their early songs. No guitar solos, no brass, no drums to speak of—in fact I'd hesitate to say it was recorded in a studio as such. God knows how it ever got released. It certainly doesn't do much for the duo's reputation as perfectionists.

'These are definitely demos,' says Katz. 'Some of these are songs that never appeared. 'Android Warehouse', 'Sun Mountain' and 'Ida Lee' are tunes that we had worked with prior to it being Steely Dan. They were just two guys writing songs—there was no Steely Dan.'

The album also appears to have been packaged by someone who has no idea what they have on their hands. Hence 'Barry Town' is rendered as 'Berry Town', although there are more startling contrasts between this and the version that ended up on the third Dan album, *Pretzel Logic*.

'There's no relationship whatsoever,' opines Katz on the two versions. 'When we went to the studio, the song was in the form it was going to be and there was never any reference to the one that had been done previously.'

It is instructive to discover that *Sun Mountain* also includes 'Caves of Altamera', a song which was not to surface until the sixth album, *The Royal Scam*. So the team were obviously not short of material in the early days—but how prepared were they when they came into the studio?

'Donald and Walter came to the studio with what they thought were finished arrangements,' Katz attests. 'I don't know if I can count on my hand the number of times that they got changed once we got to the studio, so far as structure goes. As a matter of fact, I can't think of one. They were very prepared.'

Katz met Fagen and Becker through a mutual friend and was struck by the duo's talent. Although they were all living in New York, Katz secured a job as producer for ABC in California. Once he had his feet under the desk, he soon put in a word for his friends.

'I told them that I was working with some people in New York that I thought were just terrific and please let me bring them out. You know, unheard,' Katz recalls. 'They had a lot of faith in me and they let me bring them out, and they reaped the rewards.

At first, the pair were working purely as songwriters, but Katz saw more potential for them. 'They were writing songs for a little bit for other artists and we realised that was a waste of their talent. So we decided to put a band together. The drummer was in a band that I had produced earlier and Jeff Baxter was a guitar player that I had worked with earlier. They knew Denny Dias, so we flew the three of them out.'

The first fruit of their labours was the seminal *Can't Buy a Thrill*, which was loved by critics, radio stations and the listening public on both sides of the Atlantic when it was released in 1972. Although its success would be eclipsed by later Steely Dan albums, its popularity caught almost everyone by surprise.

'There wasn't anything like that music,' Katz explains. Therefore there was nothing against which they could judge its potential popularity, apart from Katz' gut instinct. 'We didn't expect it to be popular, we just liked it. Long solos, long songs, sophisticated chords... we made the record and liked it a lot, but no-one thought it would be popular. Except for me.

'I always thought it was great, I couldn't think how anyone wouldn't like it. But Donald and Walter didn't think it was commercial and it wasn't by virtue of what was on the radio in those days. So I got proven to be more right than them, which wasn't often believe me!'

While Steely Dan arrangements were sophisticated and the performances little short of immaculate, there was one thing that distanced their music from that of most of their contemporaries. It had so much groove, it was almost impossible to sit still when it was on. 'We all listened to R&B music, jazz and black music,' Katz says. 'The grooves are always sort of laid back and that's the kind of music we liked. Maybe it wasn't the most fashionable at the time, but we couldn't record records that didn't have that type of groove. We didn't like it. These we basically blues songs turned around a bit. I can't work without tapping my foot. That was the essence of what we were, good grooves.'

Can't Buy a Thrill was recorded at the Village Recorder in LA, with Roger Nichols engineering. Nichols retained his involvement for many successive albums, but the increasing sophistication of the recording process was to see him promoted to executive engineer. Elliot Scheiner was another engineer with a long-running association with the band. It is a testimony to their pivotal roles that when I ask Katz the secret of a successful mix, he grunts: 'Get yourself a good mix engineer.'

Steely Dan's second album was released a year later. The big change on *Countdown to Ecstasy* was Donald Fagen's elevation to lead vocals, replacing David Palmer. Perhaps not the most technically accomplished singer, Fagen none the less could fill the lyrics with a kind of dark significance. Katz does not dispute Fagen's limitations, but explains why the move was unavoidable.

'He became the singer by virtue of the fact that no-one could sing the songs like he could. We tried on the first album and I personally didn't like it at all. So we had this conversation where I said —that's it, nobody's going to sing your songs but you. We all agreed that we were prepared to take however long it took to get what we wanted,' says Katz. 'I was more than willing to do that.'

Did Katz have any special technique for drawing the best out of his new vocalist? 'There's no such thing,' he counters emphatically. 'Donald got up every day. Some days he was really good and some days he wasn't. When he wasn't, we stopped and did other things.'

Pretzel Logic followed in 1974, the third album in as many years. Once again, it contained compelling songs, complex structures and great grooves. It also contained a development that was to come to the fore with the 1975 album *Katy Lied*. Walter Becker was starting to contribute some of the stranger and more memorable guitar solos.

'Walter was the bass player and he's a great bass player and very facile with it,' Katz explains. 'He wasn't as facile at that time, playing guitar, but we loved what he played. Donald and I were just huge fans of Walter and we were willing to take the time that it took Walter to accomplish what he wanted to.'

By the time album four, *Katy Lied* came round, Steely Dan's musical pool had grown to more than 16 musicians, who were carefully selected for their suitability on a certain track. For instance, Becker's contributions on bass were supplemented by Chuck Rainey and Wilton Felder, while the roster of guitarists now included session legend Larry Carlton and one-time Johnny Winter sideman Rick Derringer.

The eminence of the guest artists alone would account for the increasingly slick, jazzy sound Steely Dan were making. The sleeve notes to *Katy Lied* also reveal a singular approach to the recording process. 'This is a high fidelity recording. Steely Dan uses a specially constructed 24-channel tape recorder, a "state of the art" 36-input computerised mixdown console and some very expensive German microphones. Individual equalisation is frowned upon. The sound created by musicians and singers is reproduced as faithfully as possible...' This little treatise goes on to wax lyrical about the Neumann >

> 'We didn't expect it to be popular, we just liked it. Long solos, long songs, sophisticated chords... we made the record and liked it a lot, but no-one thought it would be popular. Except for me'

< VMS70 lathe. Quite a purist point of view in the world of pop and rock.

It's not a purist view,' Katz counters. 'It's a sonic view and purity really doesn't have a lot to do with this. Remember we were 90 percent with live recordings, so these weren't machine-orientated tracks. We were always in the school of good miking and trying to get it to sound good before EQ. It sounded good to us.'

For the record, *Katy Lied* is Katz' favourite Steely Dan album. He says he likes them all—and restates his claim to be there biggest fan—but he likes *Katy Lied* best of all.

Steely Dan albums continued to get bigger and more polished through the successive releases *Aja, The Royal Scam* and the final album, *Gaucho*. There seemed to be a constant process of refinement taking place.

'I just think it was an evolution of their music,' Katz observes. 'There was no —I don't think—preconceived notion to say, let's make this more technical or less. We approached every track from the first one on *Can't Buy a Thrill* the same way. We learned more and there were more processes around.'

One of those technological developments was digital recording, which was used for part of *Gaucho*. (Donald Fagen's first solo outing *The Nightfly* in 1982 was recorded 3M digital from start to finish.) Did they like the sound quality of digital?

'Yeah I did.' affirms Katz. 'For the kind of music we made it was really state-of-the-art sound quality. We liked it a lot.'

One of the benefits Katz will acknowledge is freedom from gradual deterioration of the multitrack as successive passes of the tape wear off the oxide, but it's not a debate he cares to enter too deeply. 'It's just a matter of taste,' he says. 'I never have listened to music in those terms. Maybe because I'm not technically inclined. If I can tap my foot, I'm liking it.'

In order to increase the continuity in sound from track to track, *Gaucho* also saw the use of triggered drum sounds, a real innovation at the time.

'Roger Nichols built a piece of equipment called the Wendel,' Katz recalls. 'And it was the first high-tech, fast trigger music machine to work with drum tracks.' By the mid eighties, replacing the 'live' drums with samples was epidemic in modern recording and bands would spend hours agonising over snare sounds. Or you could use a sequencer and cut out the middle man altogether. How did Katz feel about these techniques?

'I personally like live drums, but that's not to say that the technology these days has gotten to the point where you can't create good drum tracks without them. I still prefer in some fashion to use real drums, even if you record a drum track, capture two bars and loop them, or whatever. But the actual feel would be live.'

Katz carries a complex set of preferences when it comes to recording. He prefers live bands and to capture the band in one room, even if he subsequently only uses the drum tracks. Guitarists, he prefers in the control room where he can talk to them. Sequencers have a place he acknowledges, but he's not the kind of producer who is going to cut up someone's performance in Pro Tools. If he has to pick favourite equipment and studios, it's Neve desks and Bearsville New York, for the quality of the room.

So, Steely Dan apart, who are the artists he admires? 'Van Morrison. I just love his music,' offers Katz. 'I've been in the studio with him and I love him and the band. I guess I like more of the old stuff than the newer stuff these days. I used to love Police records. They had a good drummer. It helps a lot. I'm not a particular fan of Sting records, but I love Police records. Jimi Hendrix, I listen to Jimi Hendrix still.

'Today? Do I like anything today?' Katz muses. 'I like hip hop mainly today. I like Dr Dre records and I like Snoop records—West Coast hip hop I like a lot, it's my favourite.'

There is an unwritten, but generally understood, rule that it is bad manners for an interviewer to adopt a look of astonishment, or repeat odd parts of the interviewee's last sentence in an incredulous voice. Too late. 'Hip hop?'

'I know, it's the antithesis of what I used to do, but I like it,' Katz responds evenly. Digital Underground, with

whom Katz recorded the *Who Got The Gravy* album in 1998 are indeed a rap act, I am informed.

Katz says he would also like to work with Snoop Dog or Dr Dre: 'I like what they wind up with a lot. I like the grooves and I think it's the only form of music these days where you are listening to things that are inventive. Not all the time, of course, but when it's good it's good.

'I'm a listener like everyone else and when you hear a record that works well, you like it,' Katz explains. There is another role Katz plays as well—talent scout. 'It's part of the process. I find artists through word of mouth or people find me. But I look for artists all the time,' he affirms.

It turns out that Katz was running a record company for a while and one of the reasons he is in London is to check out new people he might be able to work with. 'That's what I do,' says Katz. 'I'm looking for artists. I'm looking for talent, which is what I really like to do: find talent and make records.'

Later on, back at the office, I check out Digital Underground on the Internet and find this entry: 'Digital Underground came out of Oakland with a homage to Parliament-Funkadelic. Shock-G formed Digital Underground in 1987 with Chopmaster J. They released a single "Underwater Rimes" which went to number one in the Netherlands. In 1989 they signed with Tommy Boy and released the single "Doowutchyalike". By 1990 Digital Underground had new members: DJ Fuse, Money-B, and Schmoovy-Schmoov. Their debut album had the hit single "The Humpty Dance". Rapped by Shock-G's alter ego Humpty Hump…' Nice research, just a little late.

But thinking about it, hip hop is American black music. Its lineage—however convoluted—can be traced back to those Drifters and Coasters which led the young Katz to a career in music, and to the R&B records which influenced Becker and Fagen. It's just, well, if you want to flip a disc onto the turntable and create a sound that goes something like, 'RikkiRikkiRikki don't lose that number', please use someone else's copy of *Pretzel Logic*. Mine's still in use. ∎

> 'Today? Do I like anything today?' Katz muses. 'I like hip hop mainly today. I like Dr Dre records and I like Snoop records—West Coast hip hop I like a lot, it's my favourite'

Digits and Mics

Continuing from last month's oeuvre, **Chris Woolf** completes his study of digital radio microphone systems

S O FAR WE HAVE NOT touched on the third area, power. No matter how a receiver is powered it is axiomatic that the transmitter runs from a battery and usually a very small one. Given that an operating period of over five hours is required the power consumption needs to be of the order of <1W–1.5W much of which goes to running the audio processing. The RF output of analogue radio microphones is rated at anywhere from 2mW–250mW. At the extreme low-end range is poor, at the higher end the problem of intermodulation between multiple channels becomes very serious and the range improvement is far from pro rata so most professional devices fall into the 30mW–50mW category.

The power rating is rather nominal because absorption by the body is highly significant and increases with frequency. At 1800MHz there is no doubt that a great deal of the available power will 'disappear'. Raising the output is to some extent self-defeating since the battery consumption increases and the intermodulation problems become more severe. Worries about excessive exposure to RF also mitigate against this policy. For most manufacturers improving reception, perhaps with increased diversity is more attractive.

Space diversity—spreading at least two aerials apart by $>\frac{1}{2}$ wavelength and using the strongest signal at any one moment—greatly reduces the fading caused by multipath signals which create deep nulls where the radio waves interfere destructively and hence improves usable range. Digital systems may ease their troubles by using multiple diversity techniques with three, four or more aerials to give good coverage and, with the extremely short wavelength 1800MHz band, to ameliorate the problems of severe shadowing. As a transmitter moves further from its receiver so the carrier-to-noise ratio becomes poorer until at <30dB (for analogue units) the noise performance of the channel starts to deteriorate. The practical range is usually governed by the point at which a multipath null is produced that is significantly worse than this—the familiar drop-out.

As with so many analogue systems, this degradation is relatively benign. A good many users will have allowed

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the odd brief 'zzzt' to pass if the signal is otherwise reliable. Manufacturers are still debating how best to deal with digital range problems. The use of bit-rate reduction necessitates some form of framing structure to the audio so any drop-out will damage the entire frame rather than simply be a momentary splat. If the frame is short it may be possible to discard it but short frames cannot be used with heavy bit-rate reduction. Error correction of the damage is possible if there is sufficient overhead to carry some redundant data but impressive error correction on a signal crippled by excessive bit rate reduction may be a poor trade off.

The acceptable data-to-noise ratio for a digital system may be as low as 10dB —it only has to recognise the difference between a 1 and a 0—but eventually the point will be reached where any error correction can no longer resolve what the signal should be. In common with other digital streams the only safe failure mode is muting, but this is far from graceful. It will prove impossible to gauge how close to failure the signal is just by listening to it. Of course a manufacturer could provide an output display of the error-correction activity that should be a clue to the proximity of disaster, but it might go against the grain for them to show how much repair work was going on during normal operation.

Another aspect of power is how the spectrum is used, digital modulation can take many forms—frequency, phase and amplitude, sometimes in combination and often in more than single steps. The advantages of simple modulation schemes are that they take relatively low RF power and are easy to detect. The disadvantage is that they cannot code very much data onto an RF carrier. Increasing the modulation complexity allows more bits to be coded but the signal becomes increasingly like an analogue one where several different levels or phases must be distinguished and this requires a higher data-to-noise ratio. That means more power or a shorter range.

The energy in these dense signals is also no longer concentrated in a narrow centre band, but tends to spread out and fill the entire allocated bandwidth. In a 'mixed economy' digital and analogue systems may make poor bedfellows and there is some resistance to letting them share RF spectrum. Digital Video Broadcasting—DVB—using COFDM (the European terrestrial system soon to be used in the US too) is proving to be even less friendly, so the accepted, even if sometimes unofficial, practice of borrowing broadcast TV channel space for radio microphones may become impossible as well as illegal.

So far we have concentrated on the problems of digital radio microphones, but there are undoubtedly advantages. Although FM signals are resistant to AM interference they still suffer when it is severe. Electrical switch 'splats' cause impulsive noise and hum fields from electrical equipment may cause audible effects. It should be possible to gain immunity with digital techniques, and, within the defined range, get greater reliability. The compander faults that beset cheap analogue systems and occasionally trip up the better ones should disappear completely too. Overall quality could also improve. We have only discussed 16-bit systems since the data rates threaten to stretch current techniques to the limit. The practical dynamic range of these is only about 90dB (10dB lower than good analogue) but the limitation is not fundamental as it is with analogue. Given clever modulation schemes or improved bit-rate reduction a 20-bit system could be developed that would give a satisfactory mastering margin.

Many manufacturers have prototype digital radio microphones on their drafting screens and workshop benches. During this year we can expect to see a few being delivered for beta test even though the RF spectrum regulators have yet to sort out a long-term home for them. This indecision is certainly unhelpful to development. For users it is undoubtedly time to think very hard about how a digital radio microphone —which is unlikely to be a simple, painless plug-in swap for its analogue brother—will affect the way that they work. They might use the short time available to evaluate and voice their needs clearly. ■

The Green Mile

Taking the hot seat, **Richard Buskin** talks with the production sound mixer, dialogue, sound effects rerecording and music rerecording mixers about their work on the Oscar-nominated story of Death Row

THE TIME IS 1935 and America is in the depths of the Great Depression. The place is Cold Mountain Penitentiary, a prison in the Deep South, and as for the inmates… Well, they're on Death Row.

Superintendent Paul Edgecomb (Tom Hanks) is a decent, peaceable guy who takes a sympathetic approach towards his work. The governor's nephew, on the other hand, is his polar opposite; a sick, sadistic guard named Percy Wetmore (Doug Hutchison) who gets his kicks by watching the sentenced men being fried in the electric chair. Among the new arrivals in this cemetery-away-from-home are both a white-trash murderer and a seven-foot black man named John Coffey (Michael Clarke Duncan), who has been charged with the rape and murder of two little girls. Stir Crazy this is not. When Coffey cures Edgecomb of a painful bladder infection by wringing it out with his hands and releasing the venom through his own mouth in a stream of mini-locusts, the kindly guard starts to perceive him in a different light. Indeed, the child-like hulk with the symbolic initials and miraculous powers just might not be guilty of the crime he's been convicted of after all. So begins the real drama.

Based on a Stephen King novel, *The Green Mile* is written and directed by Frank Darabont who previously directed King's *The Shawshank Redemption*. The title refers to the pristine green floors of the penitentiary itself, and it was this environment which presented the film's dialogue mixer Robert Litt with something of a challenge when he had to determine whether or not reverberation should be applied to the characters' voices.

'We were looking at a 20-reel picture and asking just how much a person could listen to in terms of reverberation,' says Litt who, along with his fellow dubbing crew members, had first befriended Frank Darabont when they all worked together on *The Shawshank Redemption*. 'We wrestled with that problem for a while, and what we ended up using is a very soft, subtle reverb. If you were to listen to the picture—or a reel or a scene—without the reverb in there you would definitely miss it.

'In a true jail cell you would have hard surfaces and plenty of reverb, whereas on a set it doesn't really exist. You've got a ceiling that's 40-feet high and so all of that sound gets sucked up. It felt very dry and not at all like the atmosphere which we were looking at, but then on the other hand there was a real fear that we would put on some reverb and, many reels down the line, we would screen the picture and say, "Oh my God, it's too much". Then we would be trapped and faced with how to get rid of it. So, we actually did a couple of reels without Frank being very involved, and when we asked him how he felt about the reverb he said, "Well, it feels okay to me".'

Indeed, what is refreshing about *The Green Mile*'s Oscar nomination in the Best Sound category is the fact that this isn't a special effects extravaganza that attempts to bombard the audience with dazzling sounds. Rather, it uses an atmospheric score, crisp dialogue and subtle background effects to support the story line.

Willie D Burton was the production sound mixer on the movie, recording the dialogue and any simultaneous effects, such as the sparks that were actually flying when scenes were being filmed in the execution chamber, the creaking of the guards' shoes and leather belts, and the sound of the Tennessee winds. Burton was involved in all of the main location work that took place in Nashville over the course of three months, although not that in North Carolina which was utilised for Reels 1 and 20—the film's beginning and ending—and which were shot a year later in just a few days.

'I use a Fostex D2 digital recorder,' he explains, 'and for the dialogue most of the time we try to mic overhead with conventional microphones. However, from time to time we do have to put the wireless radio mics on, so on this film I would say it was a combination of 85 to 90 per cent miking overhead and the rest of the time using wireless mics.

'The biggest problem that we ran into on location in Nashville was the 30 to 40 miles-per-hour wind, and so in that situation we would have to decide on whether we were going to use the wireless or the conventional microphones. Finally we went with the overheads because we were able to get in fairly close, cover the microphones with wind-socks and keep them out of the camera shot, whereas with radio mics we would be able to bury them out of sight and get a good sound, but it would be muffled.'

Audio postproduction work on *The Green Mile* took place at the Warner Hollywood Studios in West Hollywood, using a 96-channel Harrison PP1 analogue console equipped with Flying Faders, Fairlight to record sound effects and dialogue, and mag tape for the music department which elected to sit on the digital fence.

'The console itself is probably 12 years old,' says effects rerecording mixer Elliot Tyson, 'so I had to bring in three additional automated outboard consoles because I didn't have enough faders. They were all actually Harrison PP's which we had chopped up from bigger consoles, and so, thankfully, the automation all matched. As a result I had 96 faders just to myself, with everything wrapped around me—in fact, the way that the room was configured, there was a back bar behind us, and then elevated above that was where the director and producers sat, so I had to climb up on the back bar to get to the back console. Still, as it was all automated, once you made the move hopefully you were kind of done with it.'

In the light of the aforementioned in-house technology being a throwback to the time when a dialogue mixer didn't necessarily require more than 15 faders, and given the fact that these days said engineer would usually adhere to the adage that 'you can never have enough', both Robert Litt and music mixer Michael Herbick also had outboard consoles to help expedite their work.

'We all sat and worked together on the same stage, but while [sound designer-supervisor] Mark Mangini tended to work closely with Elliot, I worked closer with Julie Evershade,' says Litt. 'She cut all of the dialogue and all of the ADR.'

Meanwhile, receiving the music material on 24-bit Pro Tools, Michael Herbick was using anywhere from 14 to 28 channels on the Harrison console. To a director like Frank Darabont the little things mean a lot, and for the postproduction crew this translates into spending plenty of time making minute adjustments. In Herbick's case, this often amounted to playing down certain big orchestral pieces according to the director's wishes, yet the trick was to achieve this without losing any of the intended flavour. Sample the scene in which John Coffey is

discovered with the two dead girls and people start rushing towards him.

'There's a big orchestral piece that swells and builds, and I wanted to go with that, but Frank also wanted to hear the agony of the father screaming "I'll kill you" at the guy,' Herbick explains. 'So I couldn't go as big as I might have wanted to, but it worked out okay, because we filled in those gaps with the people yelling and screaming.'

For their part, as all of the predubs done by Bob Litt and Elliot Tyson were recorded to the Fairlight, they were working with a completely digital path. This included the sound effects which Mark Mangini brought in on Pro Tools, meaning that there was no mag tape involved in any of the sound effects or dialogue work.

'Mark went out and recorded specific backgrounds in prisons around Los Angeles, and then did some semi-mixing of those and brought me about 48 tracks of what we call "air candy",' recalls Tyson. 'Ninety per cent of the film takes place on *The Green Mile*, and [the film-makers] didn't want to have the standard off-stage "wallah" of other prisoners. They wanted it to be fairly confined even though it wasn't that far away, and so, at least as far as backgrounds go, we had to make the soundtrack be more than just air sitting there. It couldn't be dull. There were air tracks that had pipes breathing and the corrugated roof breathing—metallic things that came and went—and, even though you might not notice them, there were also voices way, way off in the background. All of these sounds provided texture, like a little pepper sprinkled into the air so that the whole thing wouldn't be too monotonous.

'Mark and I were picking things out of these air tracks and going, "That voice is great, but maybe it's in the wrong spot. Let's leave this one out and move it down about 20 feet into that little hole where it will work better". We were doing that all day long, even during the final mix, and we'd sometimes find things that we hadn't seen in the picture. For instance, the exterior picnic scene featuring everybody—Mark and I had watched this like a thousand times, and suddenly I said, "Hey Mark, there are two flies that go by. I've never seen them until just now". So, we put in these two ever-so-subtle

fly buzzes, and we did a lot of stuff like that which doesn't go over the top; it's right on the edge and makes it real. Of course, there's always the temptation to get too busy, but that's exactly what Mark and I don't want to do. We want the film to speak for itself instead of saying, "Watch me, listen to me, listen to me". There are a lot of other pictures that do that.'

Of course, one way of unsubtly attracting attention is by way of the surround channels, with supplementary background voices, noises and musical pieces cropping up from behind and, in some cases, panning all over the place. However, as noted before, *The Green Mile* was not the type of film to overtly go down that road.

'If it isn't already there, I usually just try to put some sort of delay on the orchestral score so that it'll fill the room,' says Michael Herbick. 'It will be anywhere from 30 to 90ms on the front channels and in the surrounds, so you'll hear most of it coming off the screen but there will also be a short delay to the back of the room and when it's all playing together you don't notice that it's coming out at different times.
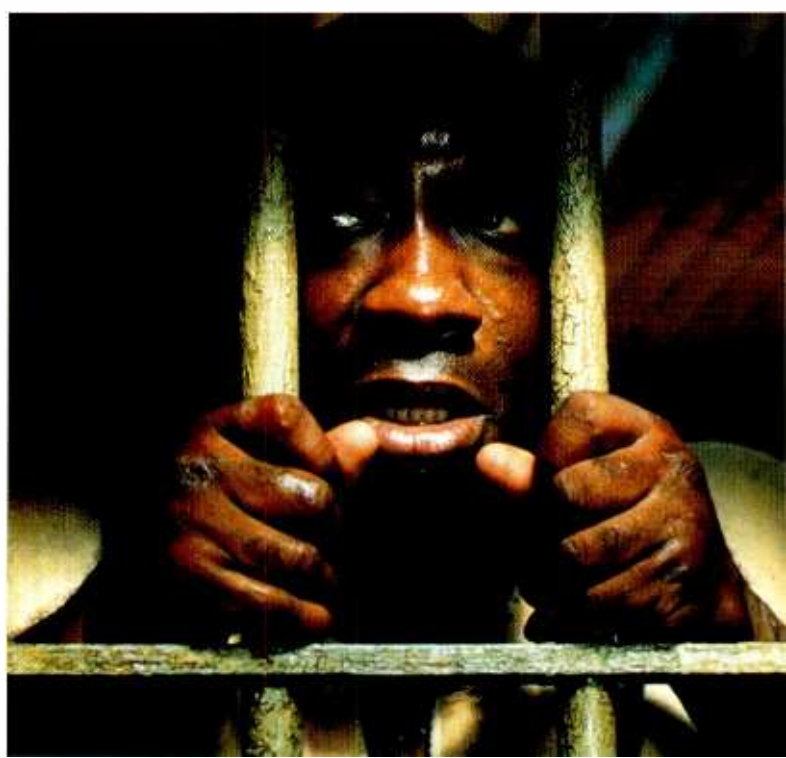
'In this film there's a scene where they put this bad guy to sleep with drugs, and the score was recorded so that there were guitars flying around the room, left to right. Well, I took those and put a bigger delay on them and put them in the surrounds, so that, as his head was going around, the guitars were spinning around the room as well. This sort of thing is down to a little trial and error. I've done this before, but how much delay you put on depends on how fast the instrument is being played, so that it doesn't sound like a repeat.' **>**

< 'Surrounds are not to be taken lightly,' adds Elliot Tyson. 'They are there to create the mood, but they're not there to distract people from what's really happening on the screen. Still, on *The Green Mile* we actually did a lot of surround work with the insects. Mark and I had the idea to fly them up over our heads, so that the audience feels them coming out of the screen. You know, they're definitely moving. We did that a little bit with the air and temperature accents, but then sometimes those things could also be distracting, and Frank made a specific point with regard to a scene where the corrugated roof was sweating and stretching and contracting. We had to excise a lot of those noises on one particular reel, and they were in the surrounds.'

Tyson and Mangini did, in fact, premix the insect scenes without Frank Darabont even hearing them, in the hope that there would be enough separation to allow for adjustments later on if necessary.

'We made about five predubs each on those scenes,' Tyson recalls, 'so we could raise, lower, take out or add anything that Frank liked or disliked. Mark would bring me 25 or 30 tracks of each different food group, as it were, and we would premix those together and then, of course, premix everything in context against those. Consequently, when we got to the final, the balances were basically there the way that Mark and I liked them, and then we could show them to Frank.

'We worked the same way on the execution scenes. They were made up of lots and lots of different elements, and for some of those scenes there were as many as nine predubs. We wanted to give the film some edge without making it super-gory, because that's one of the things we were terrified about; there were three executions and one of them was fairly long, so it could prove to be a little much. You know, it got tough to watch over and over again, so we had to strike that fine balance where we kind

of put ourselves in the movie-goers' seats and said, "What would they want to hear and see without just banging them over the head?"

'In the case of the last execution —John Coffey's execution—we elected to go with more music and less effects, and, although we predubbed it to make it real, we kind of knew that was the direction in which we would be going. However, at least it gave us the opportunity to present our vision and then pull it all back, rather than play it all down and not know in which direction the music was going, because they didn't score that scene until the last minute.'

Trying to avoid too many characteristic Foley sounds, at the inception of his work Mark Mangini decided to record all of the Foley on location. Then, because the film's sets were still available, he and the sound department formulated an idea whereby he would use the actual *The Green Mile* set with fibre optics, and send the results back to the Foley stage where they would be laid down on tracks.

'They did some experiments and the stuff sounded incredible,' says Elliot Tyson. 'When we took away the production track while we were listening on the stage, it sounded like these guys were actually walking on The Mile. It was almost scary. You played the production track and then you took it out, and it was like these guys continued walking, but where did their words go? They were just talking a second ago. It was so seamless. One thing that we had been afraid of was noise build-up, because basically you're shooting one track at a time on the set. However, when we put the tracks up, with a little bit of filtering they just melded seamlessly.

'With that, we added the squeak of the footsteps. Percy's squeak was added in Foley—or in Mangini, as we now call it. All of the web belt sounds, all of their Mangini movements, were added afterwards, and it gave this kind of heightened realism that these guys were walking on *The Mile*. After all, they did-

n't make that much sound. We had a separate key track for everybody, and we had a separate web belt, because Frank wanted that real leathery kind of guard sound. He wanted these guys to be a little larger than life whenever they moved, and the same applied to the mouse.'

That's right, the mouse; Mr Jingles, John Coffey's sidekick, and a veritable scene-stealer if ever there was one.

'Mark did mouse vocals separately and those were then processed,' explains Tyson. 'There was Foley for the mouse footsteps and anything that he does within the box with the bobbin and the little spool. Practically everything was added.'

Elliot Tyson, Robert Litt and Michael Herbick all worked for three weeks on the predubs and then followed up with four weeks of final mixing before returning after the beginning and end of the film (the book-ends) had finally been shot. In total, the post schedule ran to a total of about 12 weeks.

'We had the nicest schedule on this darn picture,' says Litt. 'For many weeks it was like nine in the morning to seven in the evening and then we'd go home, although when those first and last reels became available we were already onto *Magnolia* and *The Three Kings*. Still, we really didn't want to farm *The Green Mile* project out to anybody else, and neither did Frank, so we managed to squeeze in a couple of days here and there and finish it up.
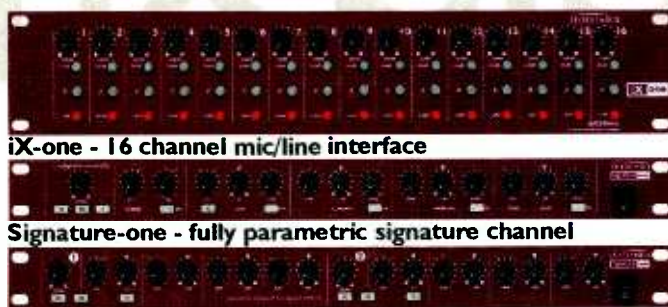
'It's not unusual to go from one project directly onto another, but it is extremely unusual to only complete a portion of a picture, totally jump ship on that, get completely immersed in another project, and then jump from that one and get re-immersed in the original film... before jumping back into yet another one. Within an hour you're back into the project and you can do it—You just have to cancel one out of your head and go into the other, and I have to admit that it does allow you to see it in another frame of light. Then again, at times it can also get pretty crazy.' ■

Starting Over

Reports of the impending death of the digital audio workstation have been greatly exaggerated. **Rob James** plays devil's advocate and files on the evolution of the DAW

WE REGRET to announce the death of Digital Audio Workstation at the tragically early age of 15. DAW was born in 1985 and survived to see the new millennium. Although a sickly infant, and remembered by many for fits of sulking and absentmindedness as a child, DAW grew into a robust, and trustworthy teenager. Always an individualist there was a period of experimentation with rock'n'roll and speed enhancement. The rise to fame was nonetheless meteoric and DAW energetically hounded elder relations into early retirement. Many of us were initially lost in wonder and admiration at the achievements. Unfortunately, this later gave way to a rather blasé and cynical acceptance of virtues and vices. Recently DAW had begun to show distinct signs of middle-age spread and loss of direction and focus. Too late these were recognised as symptoms of a terminal decline. Despite this, DAW remained useful and productive to the end.

If this report seems a little premature, please bear with me. I believe the DAW has almost had its day as an identifiable single product category. At its best as a finely honed tool for a specific job it changed the world, and perhaps in doing so, signed its own death warrant.

Consider the areas where the DAW is found in large numbers: Broadcast television was an early adopter. Here it enabled downsizing of whole departments and improved quality. To put it rather more bluntly, it enabled broadcasters to make substantial numbers of operators and engineers redundant and saved a fortune in rent on the real

estate occupied by the people, and the bulky machinery they operated and maintained.

Storage capacity and processing speed have arrived at the point where real-time video workstations are available at a fraction of the cost of an early DAW.

Highly significant for several reasons. Along with the video a number of audio tracks are thrown in along with mixing and effects unheard of on dedicated DAWs just a few years ago. Even if individual specialists continue to perform picture editing and sound-editing mixing there will soon be little justification for using different kit. There may be specific user-interfaces for sound and picture, but all will address the same underlying hardware. Networking is an integral part of this. While the basic craft competencies will remain almost the same as they always were, the user-interfaces will be immediately familiar to the computer generation and require only limited training.

At the zenith of hardware complexity and scope, the movie business, change is also afoot. Picture is now almost universally edited on NLV (Non Linear Video) workstations and sound on DAWs. Mixing takes place on huge (and hugely expensive) stages with massive automated consoles. However, the industry is seeking to reduce costs and the focus is shifting from the shooting end to post and one way of improving efficiency is to use highly sophisticated networking. In editorial, potentially this can provide tools for tightly managing projects and reduce or eliminate whole stages of copying.

A generation has grown up with com-

puters. They also understand enough of the rudiments of sound and picture grammar from constant exposure to enable them to be rapidly trained. More importantly, they are available far more cheaply than the established operators.

This may be all very well for track-laying and sound design, but surely a large sophisticated console is still required for the final mix? For any of the elite dubbing (rerecording) mixers who still think they are safe, think again. At least one respected Hollywood sound facility is using the same arguments seen in television ten years ago. These basically boil down to: 'If the same or comparable results are achievable on cheaper kit with fewer, and-or more cost-effective operators, then why not do it?' If this principle is established similar arguments will apply as in television and we will see networks carrying video, audio and metadata with remote processing addressed by various control surfaces. What we won't see are individual DAWs.

What, then, of music? The dedicated DAW has long been vastly outnumbered in this area by MIDI sequencers with audio (and often video) bolted on. Recently, the instruments themselves have begun to be seen as extensions of the package.

The best of these are highly effective and creative tools. At worst they can be jacks of all trades and masters of none. One of the most ubiquitous of the computer front ended systems, Digidesign's Pro Tools, is rapidly evolving from pure DAW into a combined sequencer, audio generator, editor and mixer with pictures attached.

So the last bastion of the true DAW is radio, right? Well wrong actually. Radio broadcasters are just as keen on finding viable networking answers to problems such as how to arrange multiple access to material and word processor-style editing of audio. Multiple access allows source material to be used in many different products—like news bulletins and in-depth analysis. Text editing of audio enables journalists to edit without technical knowledge. The logical end-point is source material physically recorded in only one place. Edit decision lists then determine how this sounds in the final products. In the bi, tri and quad media-world of film, TV, radio and the net, a single source might be manipulated by operators on different continents before contributing to all these destinations.

The reports of its demise may be exaggerated, but the essence of the DAW is certainly transubstantiating into a far more amorphous creature. Pure bred examples may survive a while longer in a few habitats, but I confidently expect the rest to be part of a remarkable evolution. Expect key manufacturers to be announcing the conception of their new children in short order. ■

Copycat Killing

Beyond the familiar and laborious use of cassette duplication systems, there are now short-run CD copiers that can revolutionise the operation of a variety of facilities. **Beth Arnold** investigates Virgin Records' Nashville operation

AS AN A&R Coordinator, at Virgin Records' Nashville outpost, an important part of Jason Krupek's job is creating compilation albums. This usually involves culling large numbers of songs to give the 8 to 15 songs that the artist, manager, producer and the A&R department feel best represent a lifetime's work. In order to ensure that the selection is all it should be, every week or two the selection is refined with the weakest songs swapped out for newer, stronger ones until the compilation is complete. At each stage a working compilation is produced so that everyone can evaluate not only the selection, every aspect of the finished record.

In his previous position with another record company, Krupek created those comps using an overworked, interlinked system of 24 cassette decks—two banks of six and one set of 12—which required him to make separate modifications and adjustments for each job on each cassette deck. The setup made the process tedious, time consuming and prone to error, which resulted in recordings—done in real time—of varying and inconsistent quality.

'Someone could accidentally bump the controls, causing decks to record hotter and possibly distort the sound, and others to be recorded at a lower level making songs sounding much quieter on the cassette,' he recalls. 'And if one cassette decided to click off in the middle of the song, it was very time consuming to stop all cassette decks, rewind each one individually and rerecord the song. Basically, you had to tend to each deck one at a time.'

Clearly, there was a better way. They called it short-run CD copying.

In Krupek's case, MediaFORM's CD-5900 CD2CD stand-alone CD duplicator offered everything he needed. And when one was installed at Virgin Records' new offices in the Fall of 1998, Krupek cured one of the worst headaches of his working life.

Capable of burning multiple CDs in a single pass, the CD-5900 tower houses eight recording bays and a reader, and will deliver eight copies with a minimum of attention. In full flight, the system will dupe **>**

MediaForm CD-5900 CD2CD

StartRec 400 also offers audio editing

Other benefits of having an in-house CD duplicator include not only the time and money the company saves by freeing up Krupek's time for other A&R responsibilities, but also the resources Virgin Records Nashville would spend on materials. It is cheaper to stock CDs at $1 per blank CD, than cassettes, which can cost roughly $3 for top-quality audio-cassettes

< around 48 full CDs in an hour—a significant improvement over the more-than-a-day turnaround traditionally associated with outsourcing CD duplications. In this, the MediaFORM tower is representative of the rapidly increasing number of CD copying systems now appearing on the market from the likes of MediaFORM, Microboards, Rimage, Copy Pro, Trax Data, Magellan and CD Cyclone. Common to all is also a dramatic space saving.

If Virgin Records Nashville were to outsource each CD duplication order, it would spend time and money on writing, submitting, delivering and retrieving each order—without taking into account the costs of materials (CDs, labels, and so on) or the time demanded of an operator. If it has to submit two or three of these orders a day, the costs to a facility obviously increase dramatically. Whether it's CD-to-CD, a CD already copied to the computer's internal hard-drive or imported from an external DAT link, Krupek consistently finds that he burns anywhere from 10 to 150 CDs a day, depending on the needs of the departments he serves.

'For promotion, it might be 30 or 40 which go to various radio stations,' he explains. 'For A&R, I'll burn 8 to 10 for the artist and the others in the department.'

Krupek has also used the CD-5900 to duplicate photo and art files created by the art department for those occasions when visual promotional material need to be forwarded to the media.

Other benefits of having an in-house CD duplicator include not only the time and money the company saves by freeing up Krupek's time for other A&R responsibilities, but also the resources Virgin Records Nashville would spend on materials. It is cheaper to stock CDs at $1 per blank CD, than cassettes, which can cost roughly $3 for top-qual-

ity audio-cassettes.

Technology in CD-R duplication equipment is constantly developing making in-house duplication an even more cost-efficient and sensible alternative to outsourcing duplication needs. In fact, newer models of CD copier are capable of duplicating up to 130 CDs (300Mb) per hour with 12x technology. Antipiracy is another persuasive argument for keeping CD duplication operations in-house, regardless of the kind of facility being operated. With the advanced antipiracy features inherent in systems such as MediaFORM's SmartDRIVE2—like watermarking and copy protection—studios can maintain greater control of content and distribution. The MediaFORM SmartSTAMP codes each CD with a unique serial code, allowing CDs to be tracked to the original source of duplication.

Krupek admits to outsourcing when he needs need 500 copies in two days, but for the most part, a short-run machine has met all the CD duplication challenges thrown at it.

The bottom line is that, for short runs, in-house CD duplication invariable saves time and money on both materials and human resources. ▨
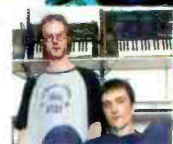

Microboards Copywriter 121

www.americanradiohistory.com

One of the defining designers in the early recording studio, Dick Swettenham's work will outlast him.

Keith Spencer-Allen

tells his story

Richard Swettenham 1927-2000

THE DEATH OF Dick Swettenham, in early April, will have caused many in the widest domains of the recording industry to stop and reflect. He was one of the 'names' of the UK studio business, and through that, both important and influential on a far wider scale. Although most widely known by the reputation he gained in the seventies, when his company, Helios, was one of the major forces in mixing console development, his is a far more diverse story.

To understand, you have to look back, perhaps 40 years, to a far, far smaller industry, where the few studios around built their own equipment and the 'rules' of the business had yet to be laid down. Multitrack was years away and even the small consoles then needed for recording were largely a technology under development. Dick was one of those who helped make that happen, pushing at the available technology without forgetting that a human being had to use the console, and that the purpose behind the whole exercise was to capture music.

After leaving the RAF in 1948, Dick spent three years studying Physics and Telecommunications Engineering at the London Northern Polytechnic before joining the maintenance department of Abbey Road Studios in 1951. Five years later he moved to Argo Records, a small independent record label where he learnt classical recording and disc mastering, as well as spending a period at La Scala in 1956. There was also an enduring parallel interest in folk music, being a director of Topic Records, and even performing himself.

He joined Olympic Sound Studios on their formation in 1958, as technical director. This was a position he held for 11 years, during which time Olympic became one of London's leading independent studios. In 1961 he built and designed what is widely considered to be the first professional transistorised mixing console—a 12-channel desk that went on to make hits for Dusty Springfield, Troggs, Rolling Stones and Bee Gees.

Following the studio's relocation to Barnes in 1966, a new generation of consoles was required. Dick was able to integrate the operational expertise of Olympic's chief engineer, Keith Grant, into the design and build of the new studios following Grant's detailed analysis of the recording process. It is here that many of the basic ideas that characterised Dick's later console designs were formed—a fusion of his technical ability, and skill in realising the operational ideas of Grant, his high profile staff, and producers passing through the studios. Certainly the early equalisers were the result of this collaborative process, while the ergonomic approach to console layout also had its origins here.

In 1969, Dick was, in his words, 'headhunted' by Chris Blackwell's fledgling Island Records to design consoles for their new studios. This opportunity led to the formation of Helios Electronics to provide consoles for Island and in turn, a long succession of further clients. Helios actively marketed itself as a bespoke console company—'We start with a blank sheet of paper…' Virtually every single order was custom designed and built. Only Island Records who ordered four consoles had the same modules, and even those were in only three of them. In total he personally designed over 50 custom consoles.

Consoles went to such UK studios as The Manor, The Townhouse, the Beatles' Apple Studios, Maison Rouge, Strawberry, the Who's Ramport, and all Island Studios. The custom building and lightweight construction techniques found favour in mobiles with consoles going to the Stones Mobile, Manor Mobiles, Maison Rouge and Island, while international sales grew steadily —to Europe and the US. The seventies also saw the beginnings of personal studios and Helios' flexibility in design gave them an enviable client list—a virtual who's who of rock—an area he liked to handle himself. An observer of Helios of the time noted 'Studios bought Neve, musician's buy Helios', which had more than an element of truth to it.

While Helios consoles were notable for frequently being of a wrap-around style, they had more conventional approaches and the company successfully moved into broadcast consoles through the seventies. One notable console was for Mainos TV which had perhaps the first integrated video monitor for level display and the introduction of 'free grouping' of faders. However, a combination of factors led the company to cease trading in 1979, the publicly quoted reason being increasing competition in the form of standard off-the-shelf products.

Dick had always run Richard Swettenham Associates, a specialised consultancy company, in parallel with Helios, and from this time onwards it was here that he focussed all his activities. Aside from studio design, planning, manufacturer consultancy and upgrading clients' consoles, he continued developing ideas on console design and digital control, particularly in relation to mobile recording. His unmistakable form was frequently to be seen at AES Conventions world-wide, usually deep in discussion with fellow designers.

He wasn't actually that interested in old ideas but had long wished to return to manufacturing products. The growing demand for his classic equaliser designs presented an opportunity. He was about to become a director and shareholder of Helios Professional Audio, a company established by close colleagues to develop both his old and future audio designs. Unfortunately this was not to happen.

Dick Swettenham died of cancer on 9th April. He was a major influence on mixing console design, one of an elite band of 'names'. He should be remembered for introducing the idea that a console should be an active combination of audio quality and ergonomics.

Or in his words, 'It is the control surface under the hands of creative engineers which enables them to achieve wonders, and the quality of the resulting sound that is their, and our, primary objective'. ■

Gimme Some Truth

1. imagine
2. crippled inside
3. oh yoko
4. jealous guy
5. it's so hard
6. i don't wanna be a soldier mama

more
main menu

Already a pop milestone, John Lennon's *Imagine* album has been remixed for surround and is the subject of a DVD documentary. **Kevin Hilton** explores its history and rerelease

IT DID NOT OCCUR TO ME that there was any particular reason why so many Beatles songs were being played on the radio that morning. I was late for school, so it was all a blurry soundtrack to my haste. Besides, the Beatles were having something of resurgence in 1980; there was nothing strange about hearing them anymore. Many people appeared subdued in class; when a friend said 'Isn't it terrible?', the shocking reality was revealed to me.

That evening the television was logjammed with tributes. The news, magazine shows and quickly assembled programmes interspersed the thoughts of friends and contemporaries with reruns of old interviews with John Lennon in his various incarnations: chirpy, wise-cracking Moptop, spaced out high priest of psychedelia, serious spokesman of a generation, drink-fuelled rocker, domesticated house husband. The abiding image was the white room where a white-clad John played and sang at a white piano while a similarly attired Yoko opened white window shutters.

Despite now being criticised for its art college philosophising and latent hypocrisy, 'Imagine' is undoubtedly one of the great songs of the late 20th Century. Last year it got to No.3 in the UK singles charts after being voted the British public's favourite lyric. The same titled album, recorded in 1971, is arguably more important: it demonstrated Lennon's development as a solo performer and writer, spawning songs that veered between melancholy and anger, 'Jealous Guy' and the vitriolic 'How Do You Sleep?' being the obvious cornerstones. Its importance as a piece of production should not be underestimated: melding what he had learned from George Martin and Abbey Road, Lennon worked with Yoko Ono, Phil Spector, a couple of other ex-Beatles and a group of leading session players to create a sparsely beautiful collection of songs.

In the twentieth year since John Lennon's murder, both album and man can be examined and reappraised through the release of a remastered, remixed audio CD and the DVD of a documentary mixing 16mm home movies and footage shot by independent film-maker and film historian Jonas Mekas. *Gimme Some Truth—The Making of John Lennon's Imagine album* is a logical progression from *Let It Be*, with every aspect of a recording, and the time-off that helped drive it along, being documented to give an insight into the creative process. John and Yoko filmed themselves and musos including George Harrison, Ringo Starr and Nicky Hopkins, while Mekas recorded visits by the enigmatic Jack Nicholson, the even more enigmatic Miles Davis and the completely enigmatic Andy Warhol.

The BBC screened a shortened version of the film earlier this year to much acclaim. This DVD version is a 63-minute cut and, as is now expected from all releases on the format, features added extras. In addition to 40 minutes of interviews filmed at the time, the release offers a stereo soundtrack, a 5.1 surround mix and a full Lennon discography with album-track listings, cover artwork and clips from the songs. Yoko Ono has been criticised, along with the surviving Beatles, for grave-robbing the works Lennon deemed not good enough for public consumption—'Free as a Bird' being the obvious, heinous example—but 'Gimme Some Truth' goes some way to justify her policy of issuing previously unreleased material. As one newspaper critic put in: 'Why we haven't seen most of this before is a mystery…[It is] a fascinating and historic rock document.'

Ono is a constant presence in the film, not just in the private moments with John but during the recording sessions as well. There is fun to be had watching her making suggestions to the supposedly dictatorial Phil Spector, who produced *Imagine* with Lennon. For this new version she visited London last autumn to oversee the album's digital remix and to executive produce the movie. Remixing and production of the DVD took place at Abbey Road Studios, where most of the Beatles' recordings were made. Now, as Abbey Road Interactive, the studio has become a leader in DVD authoring.

The stereo and 5.1 soundtracks for this new DVD release of *Gimme Some Truth* were engineered by Peter Cobbin, who has been a senior recording engineer at Abbey Road since 1995 and was instrumental in the remixing of the *Yellow Submarine* soundtrack for 5.1 release last year. Trained at EMI in Sydney, he worked as an engineer and producer in Australia before coming to Britain. Cobbin's credits include Texas, Air and Janet Jackson; at the moment he is specialising in 5.1 and has created discrete surround mixes for Bruce Johnson of the Beach Boys.

'Gimme Some Truth' is about a moment in time: how an iconic musician, at the height of his creative powers, came together with a group of like-minded artists in a specially created environment to record what is, 29 years on, regarded as a classic album. Lennon is dead, Phil Spector mad, by all accounts, Yoko difficult to get hold of at short notice; but behind the star names were the engineers who worked on the sessions. They may only be bit players in the movie who sometimes get the rough edge of Lennon's acerbic nature, but they can still claim to be part of the story.

Eddie Veale is one. Now a studio designer fronting Veale Associates, he was surprised to be asked for his reminiscences, but agreed that it might be fun. Veale's main role at the time was engineering support. He had been a staffer at Advision, involved in the studio's move from Bond Street to Great Portland Street. 'After that I was looking around for new things, for new projects for people that would help me progress,' he recalls. Veale had met the Beatles by dint of the fact that he was looking after Moog synthesisers in the UK; he had then worked at Abbey Road while George was recording 'Here Comes the Sun'. Through that the Quiet One asked Veale to help out at his private studio in Esher, Surrey; the Beatles connection continued when he became involved with Apple Studios in Saville Row.

In the wake of the Beatles' break-up >

< and speculation over who would do what, John Lennon released his acclaimed eponymous solo album. In 1971, Eddie Veale was invited down to Lennon's country home, Tittenhurst Park just outside the town of Ascot. 'He commissioned me to build a studio for him there,' Veale says. 'The intention was for what became *Imagine* to be the first thing to be recorded there. After I finished the design and build, I sat in on the sessions, mainly providing engineering support but also doing some tape op'ing.'

Veale designed the acoustics and specified the equipment for the studio, which was constructed in a single storey wing of the manor house. The console was specially designed for Lennon's needs and custom built using mostly Cadac modules; the 8-track was a 3M machine. 'The equipment and the studio embodied various ideas John took

from Abbey Road and Apple studios,' Veale explains. 'It was always intended to be a full-blown studio, not just somewhere to record demos.'

During the build, Veale and his engineering colleagues did not appreciate that Lennon had specific projects in mind for the studio. 'John became keen for us to finish the work so that he could start recording,' Veale remembers. The pace was forced when Lennon brought Phil Spector over from the States, partly with the intention of him working on the new album, partly to get his input on what the studio needed. 'Phil came in, made comments about what was already in and requested equipment and features that he wanted,' says Veale.

John and Yoko are the focus of 'Gimme Some Truth', while *Imagine* the album is centred on John, his angst, his love for Yoko and his anger towards Paul

McCartney. But one-time wunderkind and certified mad genius Phil Spector is the archetypal special guest star. For all his reputation as a task master, Spector appears vaguely bewildered by proceedings through the whole film. 'That was just Phil's way,' Veale says in the producer's defence. 'He was very laid back but when it was required, he took a strong lead. Most of the time he was quiet but things always got sorted out when he was around. He was very positive—an exciting guy to be around and work with.'

Lennon too, it appears, took a strong lead, setting a brisk pace for the sessions. 'John had very clear ideas about how he wanted everything to proceed,' Veale recollects. 'They recorded every day of the week, but there was no work at weekends, apart from some maintenance on Saturdays.' With Lennon and Spector setting the pace, the album was recorded in around two months.

Veale attributes much of this to the experience of Spector and the discipline that had been instilled in Lennon, Harrison and Starr during their earliest recordings at Abbey Road. 'Most bands today are fairly leisurely in how they

N THESE EARLY DAYS OF DVD postproduction there is the tendancy for many 5.1 soundtrack mixes to be merely fold-downs of the audio album stereo mix. *Gimme Some Truth* is regarded by those who worked on it as something of a first: a completely different approach that creates a wrap-around-sound picture from the music and the movie dialogue and effects. Remixing and remastering of the album took place during October and November 1999. Senior Abbey Road recording engineer Peter Cobbin worked on his own for this new version of the *Imagine* LP, with Yoko Ono, one of the three original producers, sitting in for many of the sessions. When it came to the DVD it was more of a team effort: Cobbin again dealt with the music while engineers Paul Hicks and Guy Massey worked on the dialogue and effects for the rehearsal and documentary sequences. The whole process was supervised by Alan Rouse, with Ms Ono offering advice.

There is always the danger on these projects that time and carelessness will have made many of the elements difficult to find, but in this case just about everything, down to the original mono location soundtrack, was available. 'All the film recordist's material was mono,' confirms Peter Cobbin, 'but the scenes in the studio and the control room were also recorded onto the 8-track, so Paul and Guy matched that as much as they could to the mono track. This gave us the scope to work in surround more often than in just one channel.'

Cobbin says he was able to 'create a sense of space' for the scenes where Lennon, Spector and Ono are listening to playbacks or discussing what to do next. He adds that he was able to do the same for the songs. 'Having done the stereo mixes, I had a good feel for what would work in surround,' Cobbin says. The starting point was, fittingly, the track 'Imagine'. 'The track was recorded was just with the rhythm section live to the 8-track, then Phil Spector overdubbed the orchestra and bounced that to another 8-track to free up the tracks. It seemed logical to put the strings in the rear channels and have the band in the front. My philosophy was

issue**25**

Solid
State
Logic

5.1 surround
mix for kd lang

The basics

Although sound delivery methods have evolved dramatically over the years, musicians still perform in front of microphones and our ears are still the final interface between the delivered audio and the human brain.

The preservation of sonic integrity within modern working methods has never been more important. No matter what the final format may be, the challenge for SSL is to capture, mix, enhance, re-mix, and route the performance to whatever delivery system is desired. SSL is unique amongst console manufacturers in offering a choice of large format multitrack consoles in either analogue or digital technology at the highest possible sonic integrity with reliability, repeatability and a world wide back-up which is second to none.

The SL 9000 J SuperAnalogue™ console is the analogue choice for the world's great studios and scoring stages. The Axiom-MT digital multitrack console has amazed engineers and producers with its accurate, warm and musical sound, and has further enhanced SSL's reputation as the leader in integrated console automation. The new MixTrack 96-channel hard disk option is an additional development in this total integration.

With Avant for Film and Post and Aysis Air for Broadcast, Solid State Logic provides consoles at the highest specification levels for installation around the world wherever serious sound is the issue – whatever the delivery system.

No matter whether your choice is analogue or digital – the future is SSL.

John Andrews, Marketing Director

*Front cover: kd lang and Damian le Gassick at O'Henry Sound Studios
Photo credit: David Goggin*

contents

Solid State Logic

The MT in Hit
Factory's Studio 4
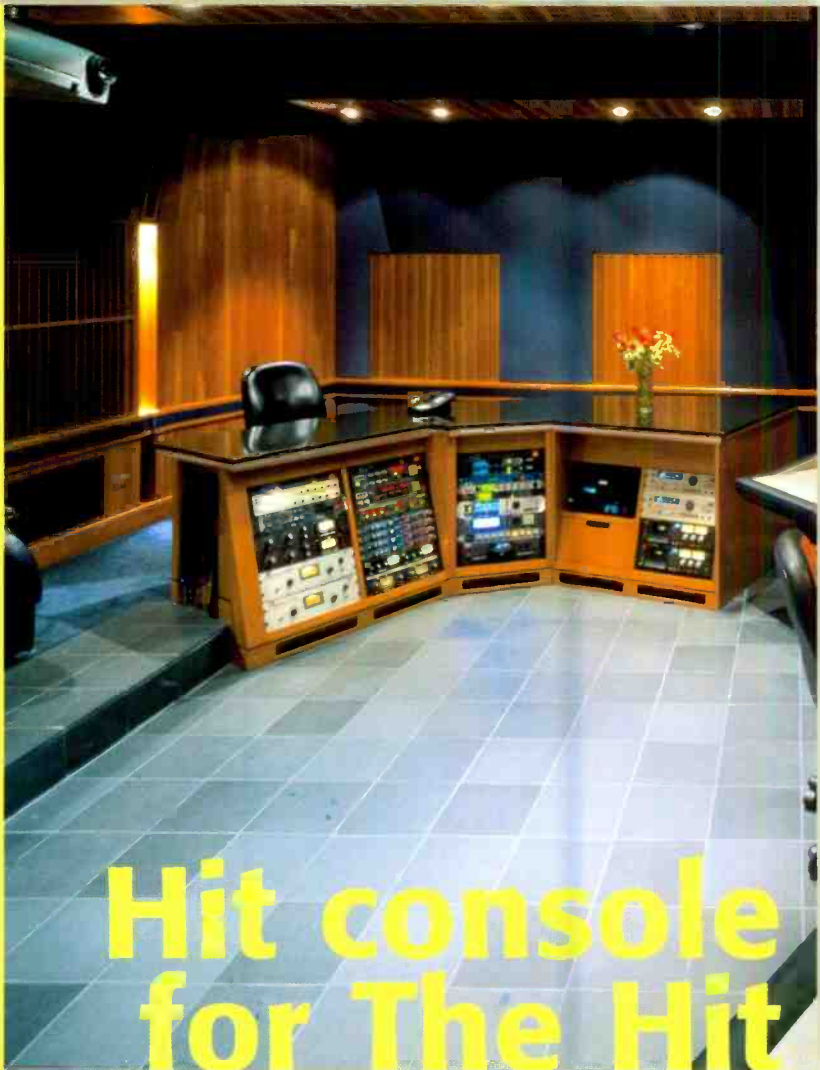
Hit console for The Hit Factory

New York's The Hit Factory, one of the world's premier recording facilities, recently installed a 48-fader, 96-channel Solid State Logic Axiom-MT digital multitrack console in Studio 4. The MT joins the seven SL 9000 J Series consoles already in service in both The Hit Factory, New York and The Hit Factory Criteria, Miami.

"Given the overwhelming success we have with the 9000, the move into the MT was a natural extension of our ongoing relationship with SSL," says Troy Germano, vice president of The Hit Factory. "The best engineers in the music business are already producing world-class results on our 9000 consoles. The MT shares the same ergonomic design as the 9K, so the learning curve for our clients and staff has been very short, allowing us to produce the same world class results immediately."

The MT, with its superb automation system and extensive channel dynamics processing, offers all the power of the 9000 with the advantages of an all-digital workflow. This combination of features serves The Hit Factory well, considering its stellar and diverse client base, which includes such artists as Billy Joel, Whitney Houston, Wyclef Jean, Paul Simon, Aerosmith, U2, Tony Bennett, Jennifer Lopez, Nas, Marc Anthony and Jim Steinman, among a host of others.

Avant moves ahead in Paris

The versatility of the Avant digital console for both film and post-production is clearly demonstrated in Paris – home to the most buoyant national film industry in Europe, including established Avant users SIS, and leading international post-production company Dubbing Brothers, who use two Avant consoles, the second in their Italian facility.

Two recent installations confirm the Avant's continued success. The first, for Jackson, is the second Avant console to be installed by this leading facility in St. Ouen, Paris. Jackson were one of the first film studios in France to equip for digital, and the latest Avant now positions them as the only film studio in Europe with four digital consoles at the same location.

The 80-channel Avant is designed to accommodate two operators for film mixing but the versatility of the console will enable the facility to mix TV dramas with a single operator. Jackson's film output is prodigious with over 35 feature films mixed each year – in addition to TV drama production.

The second Avant, also an 80-channel console, is to be installed at Copra of Boulogne, Billancourt, Paris — a company that has grown steadily throughout its long association with SSL. Copra's business has evolved through video editing and long form TV work and the decision has now been taken to expand with a fully equipped 5.1 surround theatre — with the new Avant as its centrepiece.

Studio 3 at Sarm West – home to the third SL 9000 SuperAnalogue™ console to have been installed by this leading studio group.

Invincible kd lang in 5.1

The latest project from kd lang, 'Invincible Summer,' was mixed on an SL 9000 J Series console with the SL 956 surround monitoring section at O'Henry Sound Studios in Burbank, Calif.

Produced by Damian le Gassick, the record was mixed entirely in stereo and then in 5.1 surround sound. The 5.1 version will be released on the Warner Brothers record label as part of a collection of releases produced in the DVD audio format.

"I'm extremely happy with the 5.1 mixes that I've done on the 9000," says le Gassick, who likes the console for its sonic integrity and familiarity. "This is the first mix I've done in 5.1 surround sound and working on a board that was already familiar to me was a definite plus. I didn't have to learn anything from scratch. The SL 9000 is the only board I could imagine doing this on." le Gassick worked with engineer John Smith on the production for lang's 'Invincible Summer.'

At home in Media City
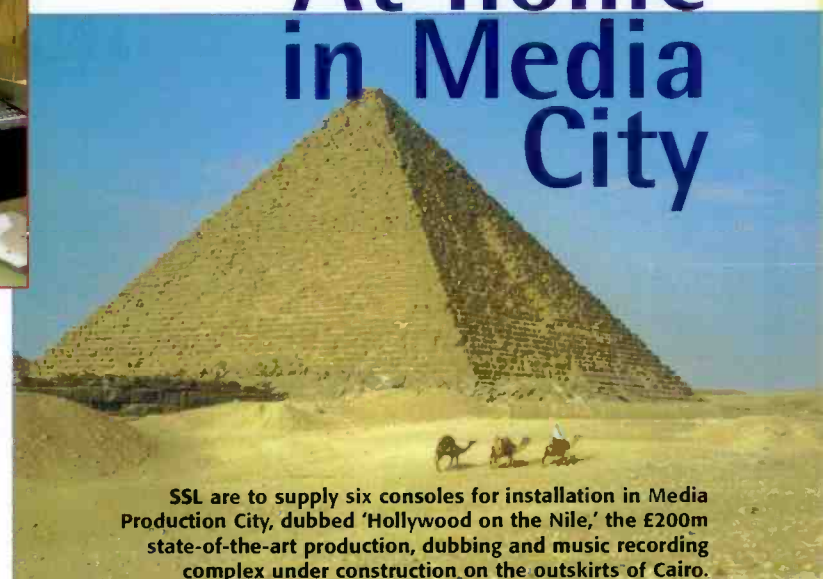
SSL are to supply six consoles for installation in Media Production City, dubbed 'Hollywood on the Nile,' the £200m state-of-the-art production, dubbing and music recording complex under construction on the outskirts of Cairo.

The 420,000m² studio complex is being built for the Egyptian State broadcaster (ERTU) and will be the focal point for future television and film production in the region.

Four SL 8032 GB analogue multitrack consoles with stereo channels will be deployed for broadcast production and a 64-channel Avant digital console will be used in a post-production role. A 96-channel Axiom-MT digital multitrack console for music recording and mixing completes the initial order.

SSL consoles were selected for the project for a combination of audio quality, ease of use and marketability — as the centre plans to attract clients from around the world.

4

SUN STUDIO

Sun shines in Copenhagen

Sun Studio of Copenhagen has an enviable record and an outstanding international reputation. With 120 staff, 35 studios, and facilities in Norway, Sweden and The Netherlands in addition to the main studio in Denmark, they are arguably the most successful audio post production company in the region.

A long-established SSL customer with numerous ScreenSounds and two OmniMix rooms, Sun Studio's latest phase of expansion incorporates a 96-channel Avant digital console which takes pride of place in Stage 1 – a new all-digital 60-seat dubbing theatre.

The new room is fully THX™ approved and is fully equipped for surround with 5.1 for Dolby SRD™, DTS™ and the newer Surround EX™ format, plus eight-channel SDDS™ – one of the few studios in the world to have such comprehensive capabilities.

Sun Studio has a long and successful history in feature film foreign language dubbing for the Hollywood majors including Disney, Warner Brothers, DreamWorks, 20th Century Fox, and Columbia TriStar. A considerable amount of work is also provided from Scandinavian TV stations and, more recently, the company has utilised its in-house project management expertise and become involved with original local film production in both sound design and mixing capacities.

Benni Christiansen, who mixes at Sun with his brother Brian and father Svend – Sun's founder – concludes, "We tried other consoles but preferred the Avant's layout and knob-per-function operation. The automation is a very powerful tool."

Aysis Air Mobile for Turner Studios

Turner Studios in Atlanta, Georgia has purchased an Aysis Air Mobile for installation in its new 53-foot production truck. The trailer, scheduled to go on-line in August, will be used primarily for live sports production.

Bob McGee, director of technical operations at Turner Studios Field Operations, sees the design philosophy SSL employs with Aysis Air Mobile as critical for the move into digital.

"The Aysis Air Mobile console control design ensures a smooth and comfortable transition from analogue to digital for our audio engineers," McGee explains. "This 'Mobile' configuration of the Aysis Air affords greater flexibility in room design due to the reduced console footprint. The SSL 6000 we are retiring was the first 6000 in a network truck 18 years ago and has outlasted two trailers. Our second 6000 purchased 14 years ago is now in its second trailer. That's a combined 32 years of continuous remote service and we have never had an on-air console failure. We believe replacing our first 6000 with the first Aysis Air Mobile is in keeping with Turner Studios' commitment to provide the highest level of dependability possible to our clients. With the Aysis Air Mobile, SSL has designed a console with the needs of the mobile broadcaster in mind."

First SL 9000 J in Berlin

Rainer Oleak's large private studio in East Berlin now boasts an SL 9056 J series SuperAnalogue™ console which will be used in conjunction with the owner's impressive collection of vintage outboard gear collected over many years.

Oleak is a well-known composer and producer from former East Germany who describes himself as an 'audio-junkie.' Trained in electronics and acoustics, Oleak became a sound engineer in 1978 before turning to professional composition.

The main body of his work is music for German TV programmes and films – he's scored over 250 in a 20 year career – and his compositions from pop to rap have reached a wider audience since reunification, with artists such as Ute Lemper and German band Puhdys.

The choice of the SL 9000 J followed lengthy evaluation but eventually the decision was made. "My budget didn't originally stretch to a 9K," Oleak candidly admits, "but as sound quality was my absolute priority, the choice became inevitable. Also I was looking for a console as an investment over the next 15 years. The 9K provides that and gives me the flexibility and features I need – and it gives me a lot of fun!"

MT: Focus on France

SSL's MT digital multitrack console has broken all sales records for large format consoles — analogue or digital — and the host nation of this year's AES European Convention has more than its fair share.

One of the first MT consoles in Europe went to Thierry Rogen's internationally renowned Mega Studio in Paris where its enthusiastic owner soon concluded, "It sounds just great — you simply wouldn't know it was digital." Rogen's enthusiasm was clearly infectious as there already now a further five MT consoles in France.

It was announced at AES Europe that Blue Sound Studio of Levallois is to install their 64-channel console in April. According to owner Carlos Duarte, Blue Sound decided on the MT "So we could take advantage of the digital productivity benefits like instant reset without surrendering the sound quality that we love."

Hot Line Studios, who relocated from Courbevoie to Levallois last summer, wanted to create a larger and more prestigious facility — adding a second studio and editing suite. They also wanted to offer their clients more performance using the latest technology and consequently examined the range of digital consoles on offer before opting for the MT.

A further MT console is soon to find a home with Roland Romanelli, the highly acclaimed French composer, who plans to install it in his private studio, Space Music, on the outskirts of Paris.

Le Voyageur's MT — installed in one of the world's most advanced mobiles — was the subject of considerable admiration at the AES Convention in Paris.

The most recent MT order in France comes from Euromedia who will install thei new 96-channel console in a digital video mobile in April.

Philippe Guerinet, Director of SSL France, accounts for the success of the MT in his country by asserting. "As soon as you listen to the MT, concerns about quality evaporate.
The next issue, especially in a country with a strong freelance tradition, is ease of use. With an intuitive, tried and tested control surface, MT addresses this concern completely and creates a clear path where analogue studios can see a logical migration to digital — and a way to develop their business."

Major TV series gets film-style mix

Randall and Hopkirk (Deceased), produced by Working Title TV for BBC1, is a million-pounds-per episode remake of a classic 60's series which now stars comedians Bob Mortimer and Vic Reeves.

With high production values incorporating a considerable amount of computer-generated effects, Working Title also invested substantially in the audio quality of the series which was mixed on one of the two SSL Avant digital consoles at Anvil Post Production in Denham, England.

Anvil's Alan Snelling explains," The new series makes considerable use of complex sound effects and music to complement the visual trickery. We mixed in Dolby Surround™ in a style more commonly used for feature films. The Avant greatly simplified that task."

(left to right) Anvil's Sound Mixer Alan Snelling with Charlie Higson, Producer and Colin Chapman, Supervising Sound Editor on Randall and Hopkirk (Deceased)

techno*file*

The Numbers Game

The characteristics of analogue consoles can be compared at a basic level by measuring various parameters and setting out tables of input noise, gain, overload margins and the like. Comparisons between digital consoles are not so easy, as digits out should equal digits in, and any differences are usually attributable to the performance of the converters.

Some pundits enjoy quoting processing speed, and figures are bandied about in much the same way as enthusiasts discuss the 0-60 acceleration times for cars and motor bikes, and whether the engine has four or five valves per cylinder. Most digital audio processing uses floating point arithmetic, and speeds are quoted in millions of operations per second — MFLOPS. The latest SHARC (Super Harvard Architecture) chips are quoted by their manufacturers as offering hundreds of MFLOPS, but this is only half the story. The ASICs (Application Specific Integrated Circuits) used in SSL's digital consoles can perform up to 7 FLOPS per clock cycle, and are very efficient at parallel processing. A further advantage of the ASIC philosophy is that ASICS do not use software and therefore, unlike SHARC devices, can't crash when a software bug bites.

SHARCs and ASICs each have their advantages, but comparing consoles is best done, as with motor vehicles, by asking what they do, not how they do it.

Matthew Willis, Technical Director
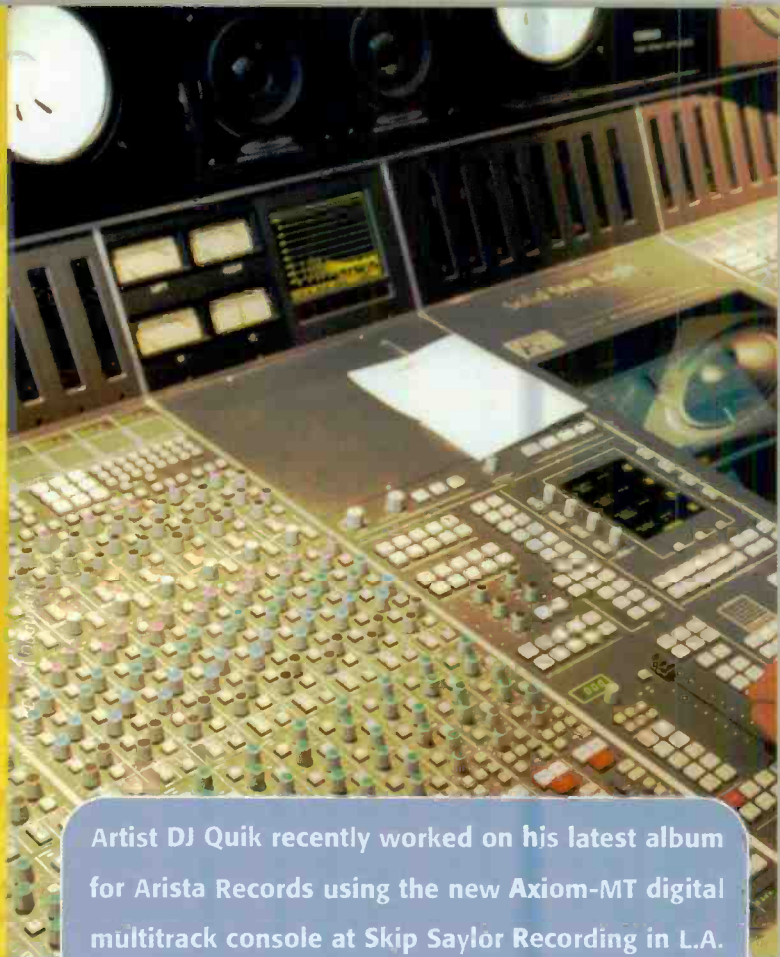
Artist DJ Quik recently worked on his latest album for Arista Records using the new Axiom-MT digital multitrack console at Skip Saylor Recording in L.A. Pictured here at the console, he has also produced artists Tupac, Snoop Dogg, R Kelly, Tony! Toni! Tone! and Ice Cube. *Photo credit: David Goggin*

Aysis Air digital broadcast console

KABC opts for Aysis Air

When KABC, the ABC owned-and-operated television station in Los Angeles, broadcasts the news from its state-of-the-art Glendale studios in September, it will be with the assistance of an Aysis Air digital broadcast console. Designed for on-air studios, production studios and OB trucks, Aysis Air will be used by KABC primarily for live news production at its new facility, currently under construction.

"We wanted an all-digital system that had a proven, on-air track record," explained Michael Englehaupt, KABC's Director of Engineering. "The Aysis Air has been on the air successfully in other television markets and we expect to continue that success in Los Angeles."

"The console provides an easy-to-learn user interface for operators," Englehaupt continued. "SSL really puts the power of the multiplexed environment into routing capabilities with an interface that is workable and makes sense."

"When you've been in a situation where you've had to pick up the phone and reach somebody, SSL's reputation is put into action. They have a very strong track record of delivering excellent customer service."

Sheffield Remote ➤ Recordings

Solid State Logic's Axiom MT is the digital console of choice...

...for Phoenix, Maryland — based Sheffield Remote Recordings, a leader in remote recording, broadcast and production. Sheffield has already found great success with the console in the nation's capital, where it made its debut at the CBS broadcast of America's Millennium Celebration.

The 96-channel MT was used to mix and broadcast an 80-piece orchestra, which was part of the live show. The broadcast was produced by Quincy Jones productions and mixed by Tom Vicari. "The MT performed beautifully," says Tom Kline, engineer-in-charge for Sheffield Remote Recordings. "The MT is perfect for complex shows such as the America's Millennium Celebration. It allows for virtually instant reset of settings, which is vital when broadcasting a live show with numerous groups or sections performing. The MT's flexibility and ease of use made it unbeatable for an event of this magnitude."

Five SL 9000 Js at Visual Arts will provide source for new talent

Following a recent order, Visual Arts, Japan's premier music engineering school, will have a total of five SL 9000 J series SuperAnalogue™ consoles installed in colleges throughout the country.

The original SL 9000, delivered to Visual Arts in Kyusyu in 1997 will now be joined by four other consoles for their engineering schools in Osaka, Tokyo, Nagoya and Sapporo.

As Takeo Asano, Managing Director of SSL KK, Japan, explains, "The adoption of the console by leading music engineering schools such as Visual Arts will help to ensure an important source of new talent for the recording industry in our country."

▼ Didier Lord, VP Ubi Soft

Northern exposure

Both the SL 9000 J SuperAnalogue™ console and the MT digital multitrack console are gaining plenty of northern exposure with new Canadian installations announced in Ontario and Montreal.

Metalworks Recording Studios, which has been host to superstars including David Bowie, The Cranberries, Anne Murray, Tom Cochrane and DEFJAM hip-hop star Ja Rule, is celebrating its twentieth year in business by building Studio 6, a new 5.1 surround mix room featuring an 80-channel 9000 J Series console. The console, which includes the SL 959 eight-track multi-format surround monitor system option, joins two other SL 4000 consoles in Studios 2 and 3.

When Montreal-based video games designers Ubi Soft sought to enhance its audio capacity, the company decided to create a specialized multimedia studio based around the MT. The new studio ensures improved quality control and optimises audio production, allowing the company to achieve a level of sound quality unparalleled in the field of computer games.

In an industry that values both technological and artistic achievements, Ubi Soft manages all stages of sound production of its products. "I strongly believe 5.1 is the immediate future of video game sound," says Didier Lord, vice president of Ubi Soft. "In order to compete in the fast-changing industry of video games, one needs quick flexibility, upgrade ability and pristine sonic quality. The MT met these tough requirements and became the obvious choice. The MT will help us lead the way in video game sound."

news bytes

Second Avant for Videolondon

Videolondon has now completed the installation of its second Avant digital console as the latest stage in the major refurbishment and digital upgrade of Studio 1.

Videolondon recently celebrated its 18th birthday and the wide variety of projects passing through its five busy studios includes longer form drama, documentary programmes and foreign language dubbing. Boasting the largest sound effects library in the West End, Videolondon also has another important asset in BAFTA "Best Sound" award-winning co-owner and Head of Sound Colin Martin.

Two years ago, Videolondon became the first in London to purchase an Avant when a 64-channel mixing console was installed in a Munro Associates designed Studio 4. According to Videolondon's, Colin Martin, "Quite simply, we selected a second Avant as we were so pleased with the first."

MT 'Made me a fan of digital'

When top record producer Tony Brown booked time with Chuck Ainlay at Backstage Studio at Sound Stage in Nashville, the experience he had on the MT convinced him of the benefits of digital mixing. The MT's flexibility, automation and exceptional sound made all the difference for Brown.

"I used the MT to mix a ballad for Wynonna's new album, and it may be the best sounding mix I have ever done," says Brown. "Chuck is always the first person in town to get the latest gear, but, because I am so busy, I didn't go into the session thinking I was going to analyse the console. I just listened to a mix and immediately was very impressed with the clarity."

The 96-channel MT at Sound Stage was also recently employed by rock legend Peter Frampton for a major concert project set for multi-format release including HDTV - with multi-channel sound, audio CD and DVD.

New German Office

Solid State Logic has established a Regional Sales Office in Germany, based in Dusseldorf.

The founding of SSL's German Office follows a successful pattern of European regional offices already established in France and Italy — areas where the company has experienced substantial growth in recent years. The German market is similarly expanding, particularly in the broadcast sector where national North German broadcaster NDR, for example, has already taken delivery of six SSL digital consoles. SSL's German Office may be contacted on 0211-737 7888-1.

MixTrack - new hard disk recording option for Axiom-MT

MixTrack is a fully integrated hard disk recording option for the Axiom-MT digital multi-track console providing a number of unique operational advantages. MixTrack's 96 track capability eliminates the need for a second 48-track DASH machine and significantly reduces costly tape head maintenance.

Robust and transparent in operation, MixTrack operates under direct Ethernet control from the console, eliminating delays in response which are typically encountered when console and recorder are operating under different software regimes.

The integrated design of the system, with MixTrack connected directly to the console processor, makes instant locate and cycle functions a given, and ensures that SSL's familiar track arming facility, a standard feature on MT, functions seamlessly. Additionally, the integration extends to the console's project management system.

Arnold Mühren migrates to MT

Studio Arnold Mühren, the leading Dutch residential recording studio on the outskirts of Amsterdam, has an established reputation of recording hits for home grown and international artists. A family-run concern, whose success has been founded on the popular SL 4000 series analogue console, the studio has now moved successfully into digital with the MT.

As Arnold Mühren explains, "We wanted to move into digital and were looking for a console that would prove as good a long term investment as our original 4K. We also needed at least 80 channels but we were tight on space in the control room. The MT provided the ideal solution here as the 48-fader frame gives us 96 channels with over 200 inputs to mix. Furthermore, the similarity of operation to the SL 9000 series console naturally makes the MT popular with freelance engineers so it will help us consolidate our leading position."

Niall Feldman appointed Director of Product Management Group

Niall Feldman will head up SSL's Product Department following his recent appointment as Director of Product Management. Niall has an outstanding twelve year record in senior roles at SSL and has served in the Pro-Audio industry for his entire career.

record,' he says, 'but the Beatles were pretty well organised in all respects. That was reflected in the pace of their recordings and performances. Their style was to get things done.'

Despite the best plans and intentions, plus engineers from Abbey Road, Veale admits that not everything went as smoothly as it would have at a longer established studio. He remembers one occasion when nobody was sure how to achieve a double-tracking effect Phil Spector wanted. The problem was solved by people travelling up to Abbey Road to see how it was done, then returning to Tittenhurst Park to do it for themselves.

Veale observes that the facilities and support industry almost taken for granted today did not exist to the same extent in the seventies. 'John was coming up with new ideas and they required a bit of thought because it was a new studio,' he says. 'It was inevitable that we would make errors because of the pressure. John obviously got frustrated and let off steam, but it was never acrimonious.'

It can be fatuous to ask someone who worked on such a significant project whether they knew at the time that they were working on something special, but the temptation is too strong. 'During the formative stages of the album, I didn't form the view that it was going to be as special as it turned out,' Eddie Veale replies honestly. 'But it started to become evident in the later stages of recording.'

The fact that, nearly 30 years on, Imagine has been remastered and a home movie of its making transferred to DVD tells you everything you need to know about how special this recording became. ■

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to be true to the style of the production and the music rather than do anything too gimmicky.'

While many in recording and postproduction are still deciding how to approach multichannel audio-only sessions, Peter Cobbin has already built up a pedigree. Most significantly he engineered the Yellow Submarine 5.1 remixes, where his work first came to the attention of Yoko Ono. 'Gimme Some Truth and Imagine started about 18 months ago, just after I finished Yellow Submarine,' he observes. 'Yoko was already putting the film together and asked me to do an experimental mix of "Imagine". She loved that, which was the catalyst for the whole project.'

A high-profile project like this, with the backing of a major record company and involvement of a strong personality like Yoko Ono, will have time and money lavished on it, explaining why it is more 5.1 than many other current releases. Peter Cobbin is of the view that it should be the norm as the format establishes itself. 'When you pick up a DVD you can never tell exactly what it is. It may say surround or 5.1, but a lot of titles are pseudo 5.1 mixes. This is a separate mix from the stereo, not just a fold-down. There was a lot of attention to detail on this; we cleaned up the original dialogue tracks and tried to eliminate as much noise as possible, all of which added to the ambience. This creates a sense of space, but is true to the original.'

Cobbin worked in Abbey Road's Studio 3, while Hicks and Massey worked on some of the dialogue in Studio 2. The bulk of the work was done on Studio 3's SSL G Series console; the tracks were laid back to a 3348 HR at 24-bit. Editing was a combination Pro Tools and Sonic Solutions (the latter device housed in the complex's production room), with the CEDAR used system for restoration and declicking. New technology was tempered with a range of classic outboard gear to match that used on the original sessions, including valve compressors and tape delays.

Yoko Ono has always endured criticism for her involvement with John Lennon. This continues today, but Peter Cobbin pays tribute to her commitment to her late husband's music. "'Jealous Guy' is Yoko's favourite; it is a very important song to her and we spent longer on that than any other song. We worked on the strings in particular because they were added later; we put those in the rear channels more than other elements. She felt very comfortable with what we were doing.'

Whereas Eddie Veale only recognised the potential of Imagine as the sessions progressed, Peter Cobbin knew full well the importance of this project and so says he was not nervous about being able to do the job, it was more a case of touching a piece of history. 'The album is very important to people,' he says, 'and there have been remixes of other things that have added inappropriate elements. I was very aware of all that, but I wanted people to listen to what I'd done with an open ear. The task was to make it sound like the original but better at the same time.'

Multichannel mayhem

Surround sound's additional delivery channels challenge accepted microphone techniques. **Terry Nelson** takes in a multichannel mic workshop

THE ADVENT of multichannel audio for music—whether Dolby Surround, DTS, DVD-Audio or SACD—is prompting experimentation in microphone techniques. It could be argued that this is not particularly new and that the seventies' B-format Ambisonics recordings have already paved the way. However, these recordings were matrixed over four channels and the current requirement is for 5.1 channels.

The Paris AES Convention presented a workshop entitled *The Comparative Study of 8 Multichannel Sound Recording Arrays of the 40 Vocal Motet by Tallis at the Paris National Conservatory of Music*, chaired by Christian Hugonnet of the Conseil Supérieure Technique, a prominent exponent of multichannel techniques. The aim of the workshop was to present recordings of the same piece using various mic techniques. Two pieces of music were chosen, the motet and a jazz ensemble using percussion instruments. The workshop featured presentations of the different arrays and techniques used, together with debate among the panel and audience.

The aim of the experiment was laudable in that it set out to provide comparative studies of different systems recording the same work under identical acoustical conditions in order to provide a realistic basis for reference. The playback system was laid out according to the ITU recommendation 775-1 and used Cabasse loudspeakers.

The hall at the Paris Conservatory where the recordings were made has a total surface of 400m² with a usable surface of 311m² The room volume is 3500m³ and the RT is 1.5s. The critical radius is 2m for an omnidirectional source. All of the various microphone systems were mounted in the same vertical plane with the exception of the array from Radio France and the system from the Conservatory itself.

The motet, *Spem in alium* by Thomas Tallis (16th century) was performed by the Madrigal de Paris under the direction of Pierre Calmelet. The piece is exceptional in that it was written for 40 voices, divided into eight 'sub-choirs' of five voices. Even more interesting is that at its original performance, the composer was thinking in terms of surround sound. The premier of the motet was in the chapel of Arundel castle, home of the Duke of Norfolk, who commissioned the work. The chapel is octagonal in shape and this explains the eight internal choirs, which were placed around the perimeter of the room with the audience in the centre. This allowed the polychoral effects to come from the different parts of the room. The recording saw the 40-piece choir placed in a circle around the microphone arrays at a radius of 4.77m. The second recording consisted of five percussion instruments with the musicians improvising on a theme by Al Jarreau.

Before opening the workshop Christian Hugonnet posed various points to ponder by the panel and the audience: are we looking for a virtual or coherent signal? How important is spatialisation? What coherence do we expect from the LCR speakers and what

phantom imaging is possible with the rear or surround speakers? Is the 'sweet spot' a vital ingredient?

The ASM 5 system is an array of five Brauner VM1 microphones with double membranes and controlled by the SPL Atmos processor. The microphones are arranged in a star with the three front microphones placed on arms of 17.5cm at 0° and ±90° and the two rear microphones placed on arms of 59.6cm and angled at ±150°. The position of the microphones can be adjusted on the arms and the capsules can be pivoted over ±45° should this be desired. The polar characteristics of the individual capsules are continuously variable from omnidirectional through to Figure-of-8, which also provides panning functions for LCR and front-to-back. In addition, the distance between the front and rear arrays can be 'enlarged' electronically.

The array proposed by Mike Williams and Guillaume Le Dû places the microphone array in relation to the loudspeaker array, which means that the location of the microphones can be optimised. The other idea is that the pickup pattern represents segments to be captured and reproduced by the microphones and speakers respectively. The array itself consists of five non-coincident pairs of Schoeps cardioid microphones. Each pair possesses a coverage angle that depends on the distance and the angle between the two microphones, these being determined by the curves established by Williams in his research work on stereophony. The five pairs are then arranged to provide a coverage angle of 360°, but without the individual pairs overlapping one another. The overall pickup performance of the array is adjusted either by manual offsets of each pair of microphones or by introducing >

< electronic offsets through delay lines.

The system proposed by Nicolas Jacques and Arnaud Mora uses five non-coincident Schoeps cardioid microphones placed on a single bar, with three front and two rearward facing positions. No matrixing or mixing techniques are required. Again, the aim is to obtain a continuous coverage over 360°. The angles between the microphones were also chosen in function of the corresponding angles between the loudspeakers and final adjustments to the coverage characteristics were made using digital delay lines.

'We have to take the customer's point of view into consideration when asking the question: Why surround?' argued Arnaud Mora 'Will he get any extra benefit from it in order to encourage him to want to buy it? This also underlines the need to have a common loudspeaker format. We also need to consider the aesthetics of recording actual events.'

He continued to say that the system offered a sensible ratio between aesthetic and operational requirements and took advantage of the quality of modern DDLs in order to reduce the phys-

> Jean-Marie Porcher of Radio France is one of the early pioneers in multichannel recording. He opened by stating: 'spectral fidelity is the first priority, followed by localisation'

ical dimensions of the array. He concluded by saying that experiments tended to show that the results were near to 'natural hearing'.

Schoeps' Jörg Wuttke said that he felt it necessary to make clear that the aim of the techniques under evaluation was the recording of classical music—should the recording be an accurate representation of the musical event or made primarily to please the ear? 'We must not forget the correlation that exists between the eye and the ear and that the conditions of a live concert are totally different to those of the living room.'

The characteristics of microphone behaviour were then discussed: for 2-channel stereo there are two main techniques, the coincident pair and AB. With coincident microphones, we are dealing with a difference in levels, but what happens to the reverberant field? We should remember that a cardioid microphone is an omni plus a figure-of-8 and that 50% of the reverberant signal will be picked up as omnidirectional (correlated). With the AB system (or spaced omnis), the RT is decorrelated and this gives a greater impression of space, given by the fact that the differences in arrival

time cause phase differences.

Pierre Antoine Signoret and Jacques Warnier of LGS (Le Grenier & Son) presented a system that consisted of the marriage of two different techniques; Ambisonics (developed by Gerzon), using the Mark V Soundfield microphone, and binaural, using a Sennheiser artificial head equipped with two DPA 4003 microphones. 'We recognise that there is no unique solution. Our aim with this setup was to ally the homogeneous space created by the Soundfield—together with its possibilities of manipulation during postproduction—with the spectral and spatial response of the artificial head.'

The recording was made with the two microphone systems positioned one over the other as close as possible. They also used two Neumann TlM170s at the back of the head for the rear signals, but these were not used in the final mix. The stereo signal recorded with the artificial head was mixed in with that of the B-format signal decoded by the Soundfield DSP 451.

Jean-Marie Porcher of Radio France is one of the early pioneers in multichannel recording and his department is continually looking at honing and refining techniques. He opened his presentation by stating that, 'spectral fidelity is the first priority, followed by localisation'. The array used for the recordings under discussion consisted of the basic Decca tree together with two rear microphones—all Neumann TLM50s. .

Porcher has concluded that by tilting the array up at the front, there is increased depth of the sound image. However, this spatial effect is frequency dependent and tends to fall off above 2kHz. The actual angle used in this case was 42°. It was noted that the front arc of the pickup area is about 100° which leaves holes in the side and rear images. The rear microphones were also placed closer to the front triangle than usual in order to improve the restitution of the lateral sources.

Gunther Theile of the IRT gave a more academic presentation and reminded the audience of the possibilities for 3/2 signals—and their shortcomings. He also pointed out that research shows that lateral phantom sources are unstable. One of the first multichannel microphone arrays developed by the IRT, the Atmos microphone, used four microphones in a quad configuration and was intended for ambience signals on OB events. A development of this is a frontal array of three microphones for LCR (the left and right microphones at 90° to the centre) used in conjunction with the Atmos array, that gives good directional and spatial imaging.

The last system was that of Catherine de Boisheraud and Jean-Christophe Messonnier of the Paris Conservatory. This was made up of 10 omnidirectional microphones placed on a circle in 36° increments. The principle is to create >

Mic arrays: A Brauner; B M Williams/G. Le Dû; C Nicolas Jacques/Arnaud Mora; D Soundfield/LGS; F Sphère Shoeps; H,H', H" Radio France; I Conservatoire de Paris

< a correlation between the localisation of the signal sources at the recording and the playback environment. The signals from the microphones were distributed as follows: the positions 0°, ±36° and ±108° were fed to centre, left-right and rear left-right respectively while the microphones at 180° and ±144° were panned between the rear speakers. The lateral microphones at ±72° were processed via a transaural system developed by IRCAM, the Spatialisateur, which sets out to reproduce the ±72° localisation of the signal source from the front left and right speakers. The system from IRCAM is also known as SPAT.

Florian Camerer of the ORF posed the questions: What do we want to record? Why do we want to try for 360°? What of the aesthetics? The reaction from the assembly showed that there is still little consensus about the technical and artistic aims of multichannel arrays for 5.1 though most people agreed that classical music would be the principal object of the exercise. Another question concerned miking techniques: Why not close mic and then pan, using the speakers to provide the localisation desired? This took the discussion out of the debate of dedicated 5.1 systems but it is worthy of consideration as a production process.

Coverage of the debate would not be complete without mentioning the fact that the ITU recommendation for loudspeaker layout also came under heavy

> I cannot help wondering whether we are chasing shadows with multichannel microphone arrays. If the aim is to reproduce the original listening experience, then it seems logical that the microphone system needs to work on the same principle as the ear-brain combination

fire and it is clear that this setup is far from receiving unanimous support.

Given the level of debate prompted by 2-mic stereo recording, the possibilities for argument over multichannel appear endless. I must admit to being disappointed at the amount of academic discussion because this often seems to miss the point. Theory is good, but psychoacoustics have a mischievous tendency to distance real life from our beliefs. For example, back-to-back M+S microphone deployment is theoretically

suspect yet Danmarks Radio's use eclipses everything else in terms of localisation and spatialisation. Whereas no sensible person would disclaim that research and theory is extremely valuable, our ears (and brain) still remain the final judge and psychoacoustics are still really in their infancy.

I also cannot help wondering whether we are chasing shadows with multichannel microphone arrays. If the aim is to reproduce the original listening experience (which would appear to be the case in this instance), then it seems logical that the microphone system needs to work on the same principle as the ear-brain combination. We already have the 'ears' in the form of microphones so why not use the power of modern DSP to provide the brainpower? Over to you, microphone manufacturers... This said, some of the systems presented were using a certain amount of processing so maybe this does bode well for the future.

What of the recordings themselves? Whereas this could be considered subjective, more than one person was surprised by the weakness of the frontal information and it appeared to me that the rear speakers dominated for most of the seating area. Again, I was disappointed by the results and they were hardly of the sort to send the average consumer stampeding for the shops. The recordings sounded dry and image had an annoying tendency to collapse into the nearest speaker. In general, the percussion recording came over best and this begs the question of the actual placement of the singers for the motet and the acoustics of the room.

Having heard John Eargle's recording of the 1812 Overture which was recorded in 5.1 (and with very little time to put the microphones into position), the artistic and aesthetic possibilities are definitely there and this also applies to other recordings that I have heard. Interestingly enough, it is the recordings made more by intuition and less with the slide rule that win out every time—is this telling us something? ■

THE STEADY and often exponential evolution of audio studio technology—especially during the last decade—has influenced many of aspects of studio life. The visible effects range from what a given budget will buy, through changes in working practises, to the nature of the kit itself and associated shifts in workspace ergonomics. Fundamentally, the tasks are the same—sound still has to be the recorded, edited, matched to picture and delivered to its audience. But the tools have become smaller and more multipurpose, control surfaces have shrunk and become multi-layered, with fewer functions needing—or offering —simultaneous direct access by touch. Armed with a mouse, monitor screen and QWERTY keyboard, you could accomplish most tasks, apart from positioning a microphone or plugging the cable into the wall box. The kit has evolved, the software responsible for much of the action is invisible, and so rooms are changing as a result.

The scale of recording, mixing and editing equipment has been tending towards smaller footprints for some time, with compact-format digital consoles becoming relatively uniform in

size. Mixing consoles, workstations and edit controllers, along with computer keyboards and pointing devices, can all fit into similar space envelopes. The comparative ergonomics of the equipment are also similar, allowing a single, seated operator to be within an arm's reach of all regularly used controls. Given this trend, you could argue that there is now far less of a requirement for custom-made studio furniture, as the prerequisite to fit furniture around a disparate set of equipment shapes and sizes no longer exists to the same extent.

Commercial aspects have also evolved. Size used to be more important for many music studios, but nowadays, most operations would not invest in a console simply because it was the largest, most visually impressive and client-wowing option available. Profitability is paramount. Kit needs to be chosen and positioned carefully, such that it will accomplish a given task as swiftly and efficiently as possible—so that jobs can be completed and despatched in the optimum timeframe, closely followed by the invoice. Speed is also of the essence when it comes to the initial installation, as most companies cannot afford the revenue lost through any unnecessary downtime.

Changing Rooms

The changing face of technology is changing the face of interior design. **Andrew Sutcliffe** fits furniture to function in the modern studio

The evolution of storage media also has an influence on the workspace, especially in the postproduction environment. Audio storage on reel-to-reel tape has evolved into the use of S-VHS or Hi-8 videocassettes for many applications. DAWs typically extend the choice of media to mag-optical discs and to fully networked systems, in which only the movement of data by cable needs to be accommodated. Physical media are therefore often only required at the entry and exit points of the process, with less need to store media at each workstation, further contributing to the downsizing of workspace furniture.

The brief would therefore seem to be for a design solution that allows a facility to arrange its equipment to meet the workspace requirements, from an ergonomic point of view, while ensuring that the quality of presentation supports a company's image. Future updates and re-arrangements have to be accommodated, with the overriding requirement being one of flexibility. A large percentage of audio equipment may well be similar in shape and scale, but the pace of technical development is still rapid. Most facilities will therefore require the option to re-arrange their workspace to cater for new developments and customer demands, and possibly at frequent intervals. The logical route is therefore to look for an off-the-shelf modular solution, and preferably one that embodies a similar attention to detail as could be provided by custom-build suppliers. A modular approach would answer the obvious questions of future-proof adaptability, provided that approach could also meet a facility's aesthetic expectations.

There is a relatively small number of modular furniture systems on the market that address the needs of audio recording and postproduction. Some have grown more out of the Musical Instrument (MI) end of the market, while others have developed from television and broadcast side of the industry. A prime example of the genre is a furniture range produced by US manufacturer Middle Atlantic Products, based in Riverdale, NJ. Called the Edit Centre range, it was launched at the NAB Convention in 1999, and comprises a choice of modules including

central desks and side-bay racks, as well as overbridge components that incorporate swivel mounts for close-field monitor speakers. The design of the units' work surfaces is based on interlocking curves, allow a flush surface to be achieved, yet with the angle between the units freely adjustable through a wide range. The desk units include cable feed-through holes and cable management facilities, with power distribution strips, while the side bay racks are available in single and twin-rack options. Reflecting good design principles, the Edit Centre's rack units are offered with a built-in ultra-low-noise cooling fan, plus gasketed front and rear

There is a small number of modular furniture systems on the market that address the needs of audio recording and postproduction. Some have grown more out of the Musical Instrument (MI) end of the market, while others have developed from the television and broadcast side of the industry

doors for noise control, making them eminently suitable for housing hard-disk arrays. A brush grommet on the cable exit maintains the integrity of the seal for noise and air-flow purposes, highlighting the design's attention to detail. A choice of hard-wearing laminate finishes for the work surfaces should satisfy most studios' aesthetic requirements.

The manufacturer cites the explosion of PC and Mac-based audio and video editing systems, and the growth of small-format digital mixing consoles, as the main drivers behind its introduction of the Edit Centre. The principal design requirements were for a system configurable by the end user to provide the free-choice benefits of a custom-built design, but without the production delays and budget levels >

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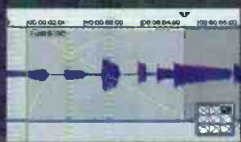
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< normally associated with that approach. Long-term flexibility was a key design issue, enabling straightforward adaptation and reconfiguration when required. Middle Atlantic Products' home market has been quick to pick up on the benefits of the system, with the products now also available in international territories, including the UK. The company produces a lower-cost modular alternative in the form of Multidesk System 2—aimed more at project studio applications—together with a large selection of heavy-duty rack enclosures, rack accessories and cable management products.

Another example comes from the video sector—the Winsted Corporation (Minneapolis, MN) manufactures a large choice of modular systems, aimed predominantly at video applications. The selection encompasses equipment consoles, rack enclosures, file-server housings and media storage—including the Multimedia furniture range. This modular furniture can be supplied with main desk units in a choice of widths, with straight or curved desk-tops, and the desks can be fitted with height-adjustable overbridge risers. Side racks are included in the system, and the design incorporates multiple cable raceways to manage interconnections, as well as below-desk paper trays and CPU support bays.

Preproduction rooms and project studios may find an appropriate solution from the Raxxess products. Located in the US at Patterson, NJ, Raxxess markets a range of studio furniture based around four alternative centre-desk configurations, with free-standing side-rack options. Side wings are available to extend the basic systems in straight runs, or at an angle through the use of wedge-shaped connecting panels. The systems use powder-coated steel for the support structures, with laminate-finished desk-tops edged with a solid oak trim. Cable management facilities are provided, and the main desk systems incorporate retractable keyboard shelves.

Pacific Research & Engineering, of Carlsbad, CA—part of the Harris Corporation—manufactures PrimeLine Modular Studio Furniture, which incorporates design features from the company's custom-built furniture. It features hardwood trims and a wide choice of laminate finishes. Designed primarily for radio and television broadcast studios, rather than being directly suited to music recording or postproduction applications, the PrimeLine is built up from centre-desk sections, with side racks and

various heights of above-desk rack turret. PrimeLine can also be supplied in a choice of heights for seated or stand-up operation. The same manufacturer also markets the QuickBilt II modular system, which is designed to provide a fast-assembly alternative for radio stations.

In Europe, Swiss manufacturer Media Engineering, based in Weiningen, offers the MEFUR extendable furniture system. The modular system has been designed principally for applications in radio broadcast and preproduction studios, and enables straight or angled desk runs to be assembled. Options include over-bridge shelving and rack housings.

The choice of furniture systems highlighted above serves to underline the validity of the modular approach. Custom-building studio furniture for aesthetic reasons, to fit with a particular interior design, will always remain the optimum option—given sufficient time and budget. Modular off-the-shelf systems, however, offer most of the benefits of custom-built studio furniture in a more accessible way and with far greater flexibility. Longer term, the benefits will be accumulative, as each subsequent change or adaptation can be accommodated with the minimum of expenditure or delay. Changing a room needs to be a swift and economically viable process. Modular systems can make this a practical reality, while ensuring that new hardware can always be located in the optimum position for comfortable and efficient operation. ∎

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Audis
Boulogne

It is billed as one of the grandest and most ambitious ground-up builds on the planet and reinforces Paris' film presence. **Zenon Schoepe** reports on work in progress

OINVILLE LE PONT is regarded as the birth place of French cinema as the site, some ten miles east of Paris, was chosen as the location for Charles Pathe's operations at the beginning of the 20th century. It also serves as the location for the Audis de Joinville facilities complex established in the mid eighties. Changes in company ownership and direction more recently saw the creation of the Duran Duboi group which combined Joinville's audio facilities with the Duboi digital special-effects operation and Duboi film studios creating a potent force on the international and local film-making scene.

Having made increasingly ingenious use of available space at the Joinville plot the decision was made to build the Audis Boulogne facility complex in the Boulogne region of Paris. The original Joinville complex has seven rooms of which three are mixing rooms with two SSL 5000s and a Harrison Series 12 plus Foley rooms, recording studios, and a full complement of editorial suites and offices. The initial plan was for the Boulogne site to serve to handle spill over work but the opportunity to grasp true expansion by combining even more audio facilities with the advanced picture capabilities of Duboi will create a complex that will amount to a full service audio and picture facility.

Still being built and nearing completion it represents one of the grandest and most ambitious film-oriented builds on the planet.

Occupying some 5500m² of floor space over several floors, many of the audio facilities were already completed at the time of my visit including a huge mix theatre quite unlike any other in Europe. Measuring 14m wide and a 2500m³ capacity it houses just over 8m of AMS Neve DFC in what is popularly regarded as the largest digital desk ever built.

Technical director and general manager John Rutledge laughs when asked if size really does make a difference. 'I think that it is more a question of comfort for the engineers,' he says. 'We have big ambitions for this room and what we wanted was the comfort of having access to as many sounds as possible which is the way that it is all going. We've gone from 3-track 35mm a few years ago to 32-track workstations very quickly. The ability to have access to all the sounds at the top of the board with the possibility to go back to a previous layer is a great advantage.'

The desk has 120 physical faders in a 160 chassis with Rutledge stipulating in his original spec that the board should be the equivalent to 150 channels of analogue plus the requisite monitoring and all the other bells and whistles required for film. 'You could say that the desk could be made in to a 4-layer deep 480-fader console,' he adds. 'It's a big beast and it has been very well received by the engineers.'

This may seem like a surprising observation given that working film practices in France dictate that the producer brings his own engineer in tow, and editorial staff and talent are taken care of directly by the production company. Despite the fact that a number of other >

< DFCs have recently also been placed in France the odds would seem to be stacked against freelance engineers gaining early experience with the board. Rutledge counters by pointing out that he is in the 'service' business.

'The way I explain it to people is that we are not Renault we are Avis,' he continues. 'At Joinville we have always been technical facilities not a producer. It's not a terribly glamourous analogy, but we are in effect a technical hotel. People check in and check out and we give them a service.

'A studio owner or manager cannot make a decision on a piece of equipment with the provision that they will only buy it if there are already people who know how to use it. Many years back I started Joinville with SSL 5000 consoles and at the time no one knew what they were, but I've always ensured that my staff are trained sufficiently so they can help and assist the engineers in using the new tools. This is a large desk, but it is also a development from other products from AMS Neve such as the Logic 1 and 2 and engineers here know these and the Capricorn. It's not a case of chalk and cheese, it's a further development.'

Gearing up and training his own people involved the loan of a small DFC accompanied by a training team from AMS Neve and this knowledge has been transferred to out-of-time familiarisation sessions for client engineers.

'Mixing a film is not just about pushing the right buttons,' states Rutledge. 'People who think they could do it often don't see the pressure of a session, all the people involved, the constraints and the choices and organisation of the work plus the artistic side. The tools should be as transparent as possible.

'And for us it had to be digital because the products exist and we can now work in this way. Here is the proof.' Audis Boulogne recently reiterated its stance on this issue with the ordering of a further two DFCs.

The room has been working since January with the desk arranged for 2-man operation to reflect and cater for the work practices of the client base. Monitoring is JBL throughout the theatres with THX certification and EX capability plus the unusual inclusion of a reinforced concrete THX baffle wall behind the screen.

Building work on the complex, which retains an older exterior, began in August 1998 and amounts to a ground-up construction with 140 50ft heavy piles for the foundations with the studios constructed as concrete boxes on springs within concrete boxes. Rutledge describes the complex as 'very technical' and has the readouts to prove it. It's a matter borne out by the fact that at the time of my visit complete and utter silence reigned in the completed and working DFC theatre while major building work

continued all around it. Rutledge contributed the technical design with Munro Associates taking care of acoustic design.

Two identical recording studios with variable acoustics and Libra Posts (Audis 10 and 11) are also complete and represent a development and reapplication of ideas from the original Joinville Audis 7. The DFC (Audis 9) mix room will be complemented by the SSL 5000-equipped Audis 8. Some 30 rooms for editorial and production offices are planned.

The machine room complete with a large integral maintenance area runs around in a type of gallery providing projection and sound sources to the rooms below. It is massive and is home to the complex's incredibly comprehensive and future-proofed patching and routeing infrastructure that pipes analogue and digital audio together with telecoms, clocks, time code and remote data around the building. Due consideration has also been applied to often neglected work areas.

'One of the problems with modern technology is that it uses processors and these go fast and get hot and to keep them cool generally involves making a noise which is annoying if your business is sound,' explains Rutledge. 'In all our edit rooms we can remote mount the computing and all the noisy bits in a nodal machine room and run cable round to the room so they just have the control head.'

Duboi's picture-oriented requirements are well beyond the scope of an audio taster on the complex, but are similarly based on integration and future-proofing to the operation's dedicated work areas.

Joinville has amassed an impressive list of international film credits and this is likely to be bolstered by the developments at Boulogne, although Rutledge says that it is imperative that it has a good handle on the French market. He is pragmatic in his analysis of why the new complex simply had to be built.

'We are still refusing clients because we don't have the capacity,' he says. 'What the state of the market is like and how much work is there for how many facilities are other questions, the issue for us is that we are turning away work and we don't like doing that.' ■

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US: What's in a title?

As technology evolves, empowers and destroys established boundaries, new terms and titles need to be defined, write **William Shakespeare** and **Dan Daley**

AND YOU THOUGHT the British were obsessed with titles? Think again. The country that brought you the Internet has developed its own obsession about titles, and it goes right to the root of who we are as Americans. In a WASP country the few words that define you to the world of business carry tremendous import.

It also has some significant implications for those of us in professional audio, as well. During a recent conversation, an old friend was trying to figure out what to call himself for a panel discussion he would soon participate in: 'I'm the president of my music production company, but that sounds generic and maybe even boastful, considering that I'm the only employee,' he said. 'I'm also a bass player, but musician sounds too thin. There's got to be something in the middle that doesn't sound too ridiculous or obscure, and that conveys a sense that my company and I mean something.'

Filling in that blank in the middle has become, seemingly, a full-time job for a lot of people in audio. And the proliferation of technology has hindered more than helped, because if the Digital Revolution is anything, it is amorphous. Here are a just a few examples of titular madness that has gripped the States:

Chief technology officer, chief technologist, chief webmaster-audio, senior vice president of new media and related business development, chief internet audio systems architect, audio & effects designer, director of online & video game audio, and (my personal favourite) media wrangler. The list goes on, each one straining human memory and the capacity of a standard-sized paper business card.

This is explainable in two ways. First, the increasing complexity and amount of technologies—audio and otherwise —require more precise definitions as to what individuals actually do in regards to those technologies. Incredibly specific titles, as ponderous, pompous or preposterous as they may seem, do serve a purpose in that anyone who has to find exactly the right person in an organisation has a better chance of doing so. Just as we never needed the term 'analogue' until digital came along, titles define our sense of place in an increasingly fragmented and specialised universe.

Secondly, and less scientifically, the title becomes part of the person's identity, a somewhat sad commentary on the state of the American psyche, but an accurate one

I think. When you are absolutely obsessed with your work—and we're famous for that —your title can often be viewed as an end in itself, defining someone not only professionally, but personally, as well, a surrogate for self-esteem.

But what this also underscores is the changing nature of the human asset in the pro-audio picture here. The simple term 'engineer' seems less and less valued by those who aspire to the position. At some of the larger professional audio schools in the US, as much as a third of the graduates never go into conventional recording studios, going off instead into bedrooms and basements with their own recording gear. They never have the chance for someone to assign them a title—they invent their own, the amorphous 'producer' being the initial default setting for most of them.

I have a new favourite story about titles. I was recently talking with Brian Pollack, the manager of Trent Reznor's (founder of Nine Inch Nails) studio in New Orleans. When he arrived in New Orleans from Los Angeles five years ago, he needed to set up a bank account. He asked Reznor's management company in New York which of the several holding companies that Reznor

Europe: Audio feat. The Tamperer

Simultaneously assuring the consumer of audio fidelity and the record companies of copyright security is proving increasingly tricky writes **Barry Fox**

TAKE IT AS READ that no-one who cares about music condones counterfeiting, piracy, or home-dubbing over legitimate purchase. Take it as read, too, that it makes sense to make studio recordings to the highest possible standard, either in quad-rate PCM (176.4kHz or 192kHz and 24-bit coding) or Direct Stream Digital one-bit code (at 2.8224MHz).

But the music and electronics industries are going to have the Devil's own job trying to sell the idea of a new quad-rate or DSD version of DVD, when CD already keeps most listeners happy and DVD already offers 5.1 surround. The idea of competing on the open market with two rival and incompatible systems, the 'official' DVD-Audio PCM system driven by Panasonic and the Sony-Philips Super Audio CD Bitstream system, is simply certifiable. Mere mortals can only boggle when the companies behind this madness talk of doctoring the music waveform to control copying, by using technology developed for use on the Internet with mid-fi MP3. But this is exactly what is now happening.

I have already catalogued the extraordinary way in which the Secure Digital

Music Initiative, 4C Entity test group and DVD Forum's Copyright Working Group chose the watermarking system developed by US company Aris—now known as Verance after merger with another US company Solana. As a reminder, though, Verance codes modifications to the analogue music waveform, which are so robust that they will withstand transmission by AM radio and over the Internet, to identify the music source and trigger copy control in future players and recorders, just like SCMS. But Verance was not tested with quad-rate audio and the tests at lower quality were run by record companies using anonymous golden ears. Talk of further tests, to be run by the UK's Phonographic Performance Ltd, has evaporated in silence.

Panasonic and Technics are now promoting '100kHz ready' hi-fi hardware, in the run-up to the DVD-Audio launch now scheduled for this autumn. Because DVD-A is the 'official' DVD Forum standard there

has never been any doubt that the 100kHz system would be used with music that has a doctored waveform. Crazy, but true.

Last August, when Sony and Philips were showing SA-CD at the IFA exhibition in Berlin, Paul Reynolds of Philips pledged: 'We do not want to tamper with the waveform'. He reminded that data pits on the SA-CD disc are both visibly and invisibly patterned so that a player can reject a counterfeit disc. There was no similar assurance from Sony because the company launched SA-CD at the Philharmonic Hall in Berlin with grand speeches from executives who then ducked the usual question and answer session. Foolishly we assumed that Philips and partner Sony were singing from the same hymn sheet. But I have now seen a statement issued by Verance in March that raises doubts.

'All of the five major world-wide record companies, BMG Entertainment, EMI Recorded Music, Sony Music

Foolishly we assumed that Philips and partner Sony were singing from the same hymn sheet

had set up would be his official employer. The secretary replied that he was now working for The Prince of Darkness. Pollack, by now very familiar with Reznor's Satanic stage persona, retorted, 'I already know that. But what's the name of the company?' As it turns out, presenting an American Express Corporate Card at restaurants and airline counters that's inscribed 'The Prince of Darkness, Inc' tends to get you pretty fast service. Pollack's own title, besides studio manager? Acolyte.

So back to my friend's problem. He's going into a situation where there might be some new business to be gleaned. He's going to have a few opportunities in the course of a one-hour panel to let the world know who he is and what he does and how good he might be at it. The title is supposed to say it all. It can't. At least not anymore. No wonder Americans often long for peerages, snapping up estates at auction just for the titles that come with them. So much easier to be introduced as Lord or Duke so-and-so, an all-encompasing title that conveys in a word a sense of omnipotence and omniscience. But the further audio engineers and producers move into entrepreneurship—and they can't move forward anymore without doing that—the less they are going to have the luxury of simple professional handles. And how you define your title has a significant effect on how you define yourself. 'What's in a name?' asked Shakespeare in *Romeo and Juliet*. The answer is, everything and then some.

Entertainment, Universal Music Group, and the Warner Music Group, as well as Sony Music Entertainment (Japan), have licensed the Verance SDMI Phase 1 watermarking system.'

Verance also claims that Sony (along with the DVD Audio camp) have taken licences to build watermarking sensors into consumer audio equipment. And Verance quotes Sony Music Entertainment Senior Vice President, Al Smith, as saying: 'As the watermarking system chosen by SDMI, Sony Music strongly supports Verance; we believe that the adoption of the SDMI standards by the music and technology companies will speed the growth of a viable, vibrant market for digital music which will benefit everyone involved, especially consumers.'

Says Masao Morita, corporate executive, Sony Music Entertainment (Japan): 'It has been our great pleasure to see the developments of watermark technologies for transmission of music, which help preventing unauthorised copying of music without losing smooth user experience in enjoying music. We believe further developments of watermark technologies will further benefit both music fans and industry,'

So when Sony and Philips relaunch SA-CD this autumn to compete head on with DVD-Audio, the key question we should all ask is—are you tampering with the waveform?

Attacking the sibilant

Establishing acceptable levels of sibilance for digital radio may be more difficult than technology promised hisses **Kevin Hilton**

BEING A TECHNOLOGY writer is often like being caught in a time warp. I am privy to news of the latest developments sooner than others are, so by the time new technologies become generally available, it's like I'm going back in time. A number of issues are giving me this feeling at the moment, but none more so than digital radio. Despite pilot schemes, prelaunches and launches, this increasingly beleaguered technology is still struggling to assert itself.

In recent weeks I have noticed DJs and radio presenters using the words 'digital radio' as a kind of marketing tag-line. 'In stereo on FM, on the world-wide web and in digital, this is…' has become the newest mantra spouted by people desperately trying to talk up to the news. (It may be significant that webcasting gets second billing above digital radio, but that's probably for another discussion.)

There is the argument that such buzz wording is superficial and superfluous—regrettably exactly the kind of thing that appeals in today's radio market. In reality, many radio transmissions have been digital, or at least had a digital source, for several years, regardless of DAB. CDs, MiniDisc and the increasing use of both DAWs and automation servers have created a digital infrastructure; the implementation of DAB multiplexes is the logical continuation and conclusion of the process.

There is still confusion about what digital radio exactly is and what it means for the listener. It has been strange hearing technology pundits being interviewed on the subject, not just because I feel slighted at not being one of them, but because I was giving similar interviews five years ago. Back then there was definite aggression towards the format, to the point where I had to tell one particularly obstreperous presenter that if he had a problem with digital radio he should take it up with his employers, not me.

Digital radio's cause has not been helped by the BBC charging ahead with 'launching' pilot services at a time when the means of receiving their services was not available. There is still a problem regarding digital radio receivers: while units are now on the market, the majority are in-car models; home units are still only produced by a small number of manufacturers at the minute. As ever there is the issue of cost, although, as we always say, this will decrease as the technology becomes established.

More worryingly the format's quality has been questioned, not least by my

'Because digital radio tends to be more transparent, we have to be more careful in the production process'

esteemed colleague to the bottom left. When first demonstrated DAB was claimed to be of 'CD quality'; this crept back to 'near CD quality' and has stayed there. People listening to test transmissions were always disappointed by what they heard: shaking off my aloof technology writer status I can understand why. It's new, it's digital, we're all going to have to buy new radios so why doesn't it sound absolutely amazing?

Audiophiles have made much of this, augmenting their case with arguments used in the CD versus vinyl debate of the eighties: the sound lacks warmth and is too real. The case for the defence uses much the same logic: DAB, like CD, is robust and convenient, going some way towards eliminating interference and doing away with the need for jiggling the dial. It is also consistent, something that can only be said of FM on very good days with the best equipment.

The source of the problem can be traced back down the distribution chain: one of the things digital is doing for radio is making the transmission process less of a villain. Presenters, operators and engineers—the ones who are left—can no longer use the venerable radio excuse 'It's alright leaving me' because, as complaints are beginning to show, it isn't.

Glyn Jones, managing editor of BBC Digital Radio, explains: 'Because digital radio tends to be more transparent, we have to be more careful in the production process. Any faults show through more clearly, so there are several areas where care has to be taken now. We're aware of some problems in the use of ISDN and where microphones aren't right for the job. There's a lot of sibilance being heard at the moment and it's not pleasant. Unfortunately DAB is not forgiving of the production process.'

Jones comments that many radio producers and engineers working with analogue transmission have had to consider two formats, either FM and MW or FM and LW. Now, because there is no processing on digital signals, there is an extra chain to take into consideration. 'It's not a case of throwing long established production methods away and starting from scratch,' he says, 'it's a matter of altering the approach and taking more care.'

This may explain problems with speech radio, but not the instances cited by Barry Fox, where certain types of music sound tinny or muffled. The obvious explanation is the transparency claimed for DAB, that transfers the blame to the source material. I can hear the audiophiles saying 'I told you so' from all the way over here.

Ampex ATR124

THE AMPEX ATR124 is, arguably, the most misunderstood tape machine in existence. Only 50 were manufactured and engineers feel that it is extremely complicated because it certainly was years ahead of its time with awesome specifications and facilities. I regard the 124 as the finest sounding tape machine ever built and today, with the correct updates and modern component availability, it can also be made to be one of the most reliable.

I have had seven ATR124s go through my workshop—two I have kept—but all now give good reliable service. When transporting a machine, it is important to make sure that the centre of the machine is supported by a wooden block, placed between the wheels from front to rear because, if it is pushed down a step, or is dropped by just a few inches, the weight of its heavy transformer distorts the base of the machine.

The first task when refurbishing a 124 is to upgrade the two relays on the Power Failure Braking System (our part Relays K1 and K2). The new relays are much larger than the existing models and space has to be created to enable room to mount them. Please note that only one of the relays needs to be a 3-pole device, so we supply one 3-pole and one 2-pole.

To make the modification, remove the panel from the power supply and strip the old relays, making careful note of the connections—note the path through the relay, not the solder position (take care to wire the new relays, the connection layout is not the same as the old). The top mounting lug has to be removed from the new relays. We do not give a template on new component relocation, but with the old relays off the panel, you can offer up the new relays and see how much space has to be made. The resistors and wire mounting post can be moved to new locations, the mounting holes redrilled, it is then possible to mount both the new relays.

Check with a low ohm meter all the wire wound resisters, paying particular attention to the heat sync type, also check both bridge rectifiers under the power supply [A1 and A2], inspect all the fuses on top of the transformer, look for any oxidising on the fuse holders. The fuse rating is 25A, and you should look for any blackening around the transformer cables.

There are two earth rails on the transformer-PSU assembly, one is digital, the second is common ground, if these are not at the correct connection then you

Tony Arnold looks at the assessment and tweaking of the last machine that Ampex made in his series on the brand

will have all sorts of problems—this is a very common cause of trouble.

Check all TO3 transistors in the power supply including the main MDA power transistors [Q3, Q4, Q5, Q6 = capstan; Q26, Q27, Q9 Q10 = take-up motor; Q7, Q8, Q28, Q29 = supply motor]. If any of these look to be in poor condition, replace all four as related to the motor that they drive. It is wise to check the reel servo card, making sure that R74, R65, R55, and R45 show no sign of overheating, and to reverse check the adjacent diodes CR16, CR17, CR18, and CR19. Also check U3 = IC = UA741CP.

We manufacture a bench-test jig for the reel servo PWA while it is out of the machine, as anything wrong with this can damage other components in the MDAs. While you are in the power supply, remove the battery holder and clean the contacts, or fit a new holder, apply anticorrosion electrical lubrication and change the four AA-type rechargable batteries.

We also recommend an additional fan to ensure improved cooling. The PSU has a tunnel-type heat sink, with a fan at the far end, this fan should be reversed, to enable the hot air to leave the machine and a further fan fitted at the front end of the tunnel to assist in pulling the hot air out of the machine. Both fans should be identical, axial type X 100 CFM X 3,200rpm, 110V AC, wired parallel to the original—do not fit a DC fan. You will have to cut out the front grill to accommodate the new fan.

A thorough inspection of the circuit breaker assembly is a must, including the power relay assembly. Look for high-current tarnishing or blackening, also check the relay contacts and thoroughly clean or replace.

The next move is to clean all Molex connectors, pay particular attention to those connected to the PSU.

Before switching the machine on, check the remote cable to ensure that no cables are shorting and that the screen is made at the remote female end only, on pin letter CS. At the male end, both CR and CS should be disconnected, and the cable must be the proper Ampex recommended cable [Belden Cable part number 9551]

otherwise you will get severe crosstalk problems effecting both transport and the audio functions. The correct complete part No Assy 4051017-02 for this cable, is printed on to black sleeving on the cable itself, the spec for this cable is on page 8–14 in the updated service manual.

For a reliable 124, then, we advise that you check that the offset card [assembly 4050907-04] is changed to the latest auto offset card [assembly 4051221] to reduce extreme capstan motor work by sensing capstan motor current and proportionally adjusting reel motor torques. (Note that modification to both the reel servo and capstan PWAs is required.)

The life expectancy of an EPROM is about 8-10 years if they are the originals, and so they should be changed as they are likely to be fragmented. We do blow new EPROMs and also supply the programmes on diskette for DIY clients. We always fit new EPROMs to fully refurbished ATR124s.

The current EPROMs are CPU PWA = U27 = 4570029-05, U26 = 4570028-05 and U25 = 4570030-05. The Control Panel = 4570026-05. The 100 Point Search to Cue = U14 = 133271-01B; U15 = 133271-02B; U16 = 133271-03B.

Before switching the machine on, remove all PWAs (cards), the only PWA's that should be installed are the Power Supply Regulating PWA, and the CPU PWA, switch on and check all voltages on the front of the regulator PWA with a scope and voltmeter, checking for any ripple. The next move is to install the reel servo PWA on an extender card. You should only add further PWAs to the machine's control bay, making sure that the regulator PWA does not change its voltage at its various test points. Do not add any audio cards or audio control cards at this stage.

If the power supply has not changed under the load of the PWAs being added to the control bay, or the resisters on the reel servo board do not show any >

Normal Crosstable head (wide lamination 13mm) **Lo—Crosstable head (narrow lamination 9mm)**

< signs of overheating, you can carry out all tests from Page 5–10 right through to Page 5–23 inclusive in the maintenance section of the Service Manual Part No 4890425-01 (if you have the older manual Part No 4890419-01

then the pages will be Page 5–42 through to Page 5–54 inclusive).

It is important that the transport is running perfectly because the audio will not work correctly until the transport is perfect. Any problems in the transport will cause the audio to come out of record mode. Other problems with the transport will be caused through poor adjustment, fragmented EPROMs, faulty tachos, faulty LM381 on capstan tach PWA, faulty LED TIL31 on capstan LED PWA, faulty Photo Transistor on Capstan Tach Sensor PWA, faulty DS1 on Position Adjust PWA, and faulty Offset Adjust PWA.

When you are satisfied with the transport, you should now check your Audio Control PWAs. Install your extension PWA into the first Audio Control Position without the Audio Control PWA, check for

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8 Volts on pin Nos 99 or 100 using pin Nos 95, 96, 97, and 98 for ground. If you have 8 Volts then fit the first Audio Control Card into the Extension Card. There are three bays, carry out this test on each. (Warning do not fit or remove PWAs while the machine is switched on.)

Next check that the 5 Volt regulators U49 and U50 are not faulty, also make sure there are no oscillations, the two tant decoupling caps C5 and C32, replace with new items. Carry this out on all three audio control cards. Some engineers recap everything, but we agree with recapping in the right places as a complete recap is a long and expensive task. Before fitting the audio cards check, or replace C303, C310, C604 and C605 as these caps will most likely cause severe problems.

If any of the inductors L602, L603, L304 or L305 have been changed, chances are they have been changed to a higher current version. When any of the above listed caps goes short, the older type of inductor would go open circuit due to excess current to ground but any higher current inductors would place a short on your + or - rail causing the PSU (on that rail only) to go into shutdown mode. This then causes the transport to stay in your last transport command until the tape reaches the end or you switch the machine off.

Check that the relay screening plate on the side of each audio card has not come unstuck and made contact with its adjacent audio card causing a short. Only fit one bay of audio cards at a time, checking volts at the regulator PWA after you have added the 8 Audio PWAs. When you have completed the 24 cards if you have any transport problems such as the machine going into Record illumination only when not requested then this problem can be fragmented EPROMs or a faulty U201 = TL084CN on the Record Timing part of the audio PWA.

Here are a few tips and warnings on various updates. The thing that concerns most owners is the lo-crosstalk heads.

The play head was never updated, although I do prefer the Flux Magnetic Replay Head for its extended frequency response (30ips 20Hz–50kHz ±2dB). The record head is the cause of the crosstalk problem and if you have the old type record-sync head Part No 4350372-01 or 4350372-02, then change it, because this was later replaced with an improved crosstalk head, Part No.4350732-01 or the new Flux Magnetic Type Record Head. We supply Flux Magnetic 124 heads on a 6-week turnaround, ATR102 ^{1}/$_2$ or ^{1}/$_4$-inch heads are ex-stock. You can recognise the lo-crosstalk head without removing it from the machine.

Common fault diagnosis can be found on the website: www.prostudio.com/studiosound ∎

The world's first independent analysis of the international professional audio business...

Studio Sound has now published two volumes of invaluable analysis covering the key trends and issues affecting pro audio manufacturers and facilities across the globe - essential reading for all industry professionals with a need to know their market's growth areas and forecasts to make informed decisions for the future.

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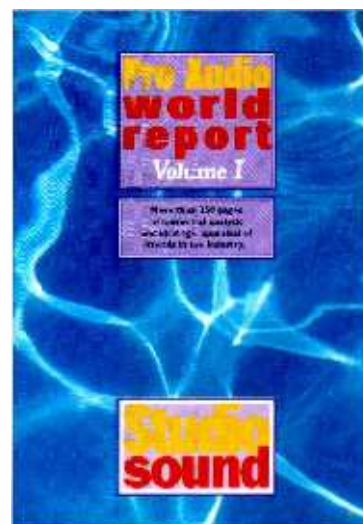
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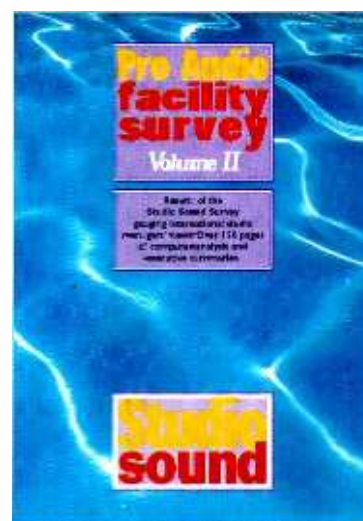
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Transforms and Spectra 3

In the third study of transforms and spectra, we note that square waves do not occur in nature and wonder why we test audio equipment with them. **John Watkinson** explains

THE SQUARE WAVE is a useful signal because it has some interesting properties. Fig.1a shows an ideal square wave which is a waveform spending 50% of its time at each of two fixed voltages. Fig.1b shows the spectrum of an ideal square wave that has a sinx/x shape. The spectrum passes through zero periodically at all even multiples of the fundamental, showing that a square wave has no even harmonics. The remaining odd harmonics alternate in phase.

Note that the spectrum of an ideal
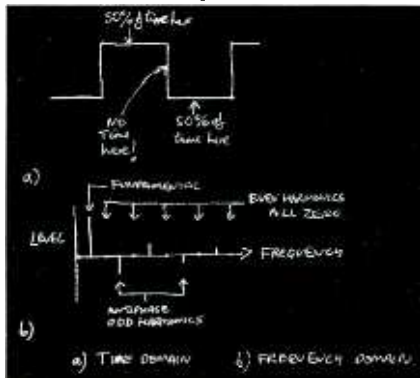
Fig.1: Square waves

square wave is infinite. This is consistent with the time domain waveform of Fig.1a in which the vertical transitions have a rise time of zero, requiring circuitry with an infinite slewing rate. As a result it is impossible to generate an ideal square wave and equally impossible to display one on a 'scope.

In the real world all we can do is to produce a band-limited square wave to display on our band-limited test gear. This is just fine because the whole object of the audio industry is to excite band-limited human ears. The band limit of human hearing has been established beyond question by countless scientific experiments, but periodically the limit is questioned by people who perform experiments which suggest otherwise. These results only show that the experiments concerned are unscientific, but finding the inevitable flaw in the experimental process is good sport.

A band-limited square wave lacks the frequency components that make it square and as a result it will contain what looks like damped oscillation leading to the term 'ringing'. Ringing is often said to be a defect of filters, but this is incorrect. In fact the ringing proves that the filter is doing its job of band limiting. It is possible to estimate the frequency response of a low-pass filter by examining the ringing.

Fig.2 shows a system with an ideal

filter cutting off at 20kHz bandwidth and passing square waves (or what is left of them) of various frequencies. With an 8kHz square wave input, the output (Fig.2a) is a sine wave because only the fundamental is below the cut-off frequency. The third harmonic is at 24kHz and isn't passed. With a 5kHz square wave (Fig.2b), the third harmonic and the fundamental pass, whereas with a 3kHz square wave (Fig.2c) the fifth harmonic can also pass. Thus by counting the number of bumps in each wave the

Fig.2: Result of filtering

order of the highest harmonic to pass can be established.

From the above it should be clear that in an ideal 20kHz band-limited system the same output will result whether the input is an 8kHz sine wave or an 8kHz square wave, and this can be demonstrated with a suitably well-engineered filter. Repeating the experiment with a non-ideal device such as a mixing console or an audio power amplifier won't give the same result. The high harmonics of the square wave will not necessarily be handled well by analogue circuitry designed for audio bandwidths. The thirteenth harmonic of 8kHz is 104kHz—approach-

ing radio frequency. Audio devices are notoriously nonlinear at such frequencies because they have run out of loop gain without which the linearising effect of feedback does not work. Fig.3 shows that in nonlinear circuitry, adjacent harmonics of the square wave, which are 16kHz apart, will intermodulate with the fundamental to produce a series of harmonics at 8kHz spacing, and one of these is at an absolute frequency of 16kHz and is clearly audible, giving the impression of second harmonic distortion.

Of course in the absence of the explanation of Fig.3, the unscientists will claim that the audible difference between a sine and a square wave 'proves' that we have bat-like hearing.

The proper approach to high-quality audio is to ensure that no piece of equipment ever receives a signal it can't handle precisely. As all real mixing consoles and power amplifiers perform poorly with radio frequency inputs, professionalism demands that at some point prior to such a device there is a low-pass filter that defines the system bandwidth.

The presence of this filter, with a frequency response such that it only
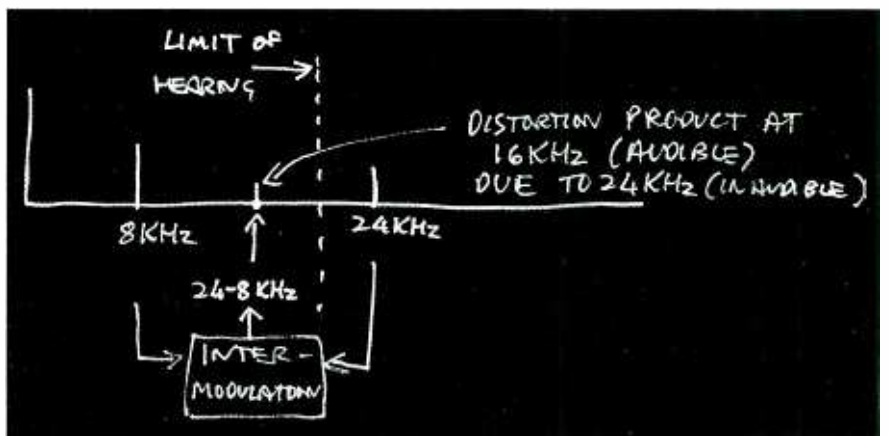
Fig.3: Why signals above the human hearing limit can change the sound: its because the circuitry is non-line

affects signals well beyond 20kHz, may actually improve sound quality because out-of-band input signals are unable to excite high-frequency nonlinearities. Of course the filter has to be of high quality, or the whole object is defeated. A number of high-quality power amplifiers use this bandwidth definition approach, and, of course, digital audio by definition has to have a defined bandwidth to prevent aliasing. Again provided the filter is good enough, the fact that the bandwidth is defined is a further reason digital audio equipment sounds so good, where in comparison most analogue systems misguidedly lack a defined bandwidth.

Square wave testing can also tell us about the low frequency response of a system. The flat parts of the waveform contain low frequency components and if the response is lacking the waveform will 'droop'. Fig.4 shows the result of passing a square wave through a system with a high-pass characteristic. The constant voltage at the extremes of the square wave cannot be maintained. In practice a response down to DC is unnecessary. Loudspeakers can't reproduce it and we can't hear it. DC components reduce dynamic
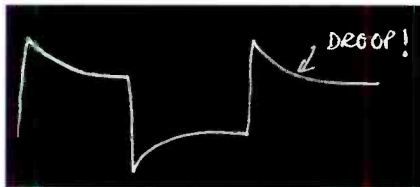
Fig.4: Droop shows poor LF response

range, fry voice-coils and cause thumps at edit points, so we can well do without them. In practice a lower cut-off frequency of 6Hz seems to be a good compromise for analogue circuitry. A number of stages with such a response can be cascaded without audible detriment.

I have commented before that the audio industry's worst failing is its fixation with frequency response, and I'm not about to make the same mistake. Again the square wave comes to our help because it provides a very simple means for checking for phase linearity and time response.

Audio systems should always display linear phase because this is the only way to avoid timing errors between different frequency components which damage realism. A system that is non-phase-linear causes what is called linear distortion of the waveform. In other words the waveform is changed without any harmonics being generated. As a result the spectrum remains unchanged in the presence of linear distortion because the same frequency content is present in both input and output. However, when a system is not phase linear, different frequency components experience different delays.

The square wave shows up these different delays very clearly, whether or not there is any band limiting. Fig.5 shows that when, say, the third harmonic doesn't time-align with the fundamental, the waveform is quite different. Returning to transform

fundamentals, any circuit having a certain frequency and phase response will have a corresponding impulse response in the time domain. It is fundamental that when a system displays linear phase, the impulse response must be symmetrical. Thus when a square wave is band limited by a phase linear filter, the preringing before the transition looks like a double-mirrored version of the post ringing after the transition. If this isn't the case, the filter is not phase linear. If we can't achieve phase linearity, at least the phase should change smoothly across the band. In conclusion, the humble square wave may not occur in nature, but it has a lot going for it as an audio test signal. As square waves contain a rich spectrum, they can test for frequency response at both extremes of the frequency range, they can test for nonlinear distortion that shows up as the presence of even harmonics as well as being able to display at a glance any lack of phase linearity—not bad for one signal.

It would be misleading to suggest that square-wave testing is all we need; far from it. The audio industry still lacks a comprehensive set of audio test signals which will reveal what a device really sounds like, although there is no fundamental reason why this should be so. What can be stated with confidence is that if a device fails a square wave test, it will impair the audio quality, whereas if it passes a square wave test it may or may not have other deficiencies.

That said, many everyday devices such as mic amps, mixing consoles and power amplifiers have quite respectable square wave response, unlike many microphones and loudspeakers which generally don't. The reason generally given is that it's not possible. This sounds to me like giving up before starting. In fact mics and loudspeakers with an excellent square wave response are perfectly feasible and only require some good engineering.

When two systems have the same frequency and phase response and the same percentage of THD, why can they sound different? In my view that's not putting it the right way round. Why should they sound the same when THD only measures the amount of unwanted stuff in the spectrum and not its distribution?

Put it another way. Would you prefer to be hit on the head with a brick or the same weight of sand? Both weigh the same on the scales, but the subjective effect is quite different. ■
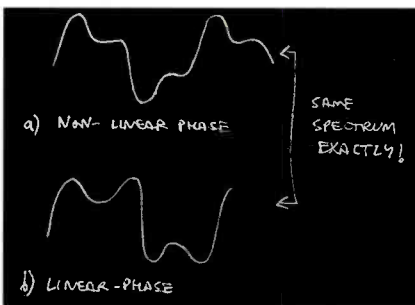

Fig.5: Linear distortion

Maintaining the standard

As managing director of CEDAR Audio, **Gordon Reid** has no objections whatsoever to increased audio quality

SCENE 1: THE NIGHTMARE (Part 1). You drive a nice, sensible car. Nothing too flash, but it is well designed, reliable, not too greedy for fuel, and it gets you from here to there with a minimum of fuss.

The larger car manufacturers have decided that they want more of your cash. So they mount an extensive campaign to persuade you that you require one of their newer and more expensive models. Sleek and sexy, these are capable of double the speed of previous cars, although they require a great deal more petrol to achieve this. You would prefer to spend your money on something else, but you think that you have no choice. So you dump your dependable, reliable vehicle (which has years of life left in it) and replace it with something newer and more expensive that, in essence, performs exactly the same task. Unfortunately, just as you fork out thousands of pounds-dollars-euros-yen on your new car, the maximum speed limit in your neighbourhood plunges from 70mph to 7mph. It's a nightmare, but fortunately, it hasn't happened.

Scene 2: The Nightmare (Part 2). You work in the pro-audio industry. You have a studio with a good selection of equipment. Nothing too flash, but it's well designed, reliable, and it gets the job done with a minimum of fuss.

The larger equipment manufacturers have decided that they want more of your cash. So they mount an extensive campaign to persuade you that you require several of their newer and more expensive units. Sleek and sexy, these handle double the amount of data as previous models, although they require a great deal more processing power to achieve this. You would prefer to spend your money on something else, but you think that you have no choice. So you dump your dependable, reliable equipment (which has years of life left in it) and replace it with something newer and more expensive that, in essence, performs exactly the same task.

Unfortunately, just as you fork out thousands of pounds-dollars-euros-zloty on your new equipment, the audio delivery format in your neighbourhood plunges from 16-bit, 44.1kHz to 128kbit MP3. It's a nightmare. Unfortunately, it's happening right now.

Scene 3: A Dream Within A Dream. You are an engineer within a pro-audio equipment manufacturer. You have developed a good selection of equipment. Nothing too flash, but it's well designed, reliable, and it gets the job done with a minimum of fuss.

The larger equipment manufacturers have decided that they want more of your customers' cash. So they mount an extensive campaign to persuade them that they require a new audio data format. Sleek and sexy, this format requires double the amount of data space as the previous format, and it requires many times the processing power to manipulate it in the same ways as before. Your customers would prefer to spend their money on something else, but they think that they have no choice. So they dump their dependable, reliable equipment (which has years of life left in it) and demand that you provide significantly more powerful hardware platforms that in essence perform exactly the same task.

Unfortunately, just as you fork out tens of thousands of pounds-dollars-euros-dinars on your new development projects, the audio delivery format in your neck of the woods plunges from 16-bit, 44.1kHz to 128kbit MP3. Never mind. Your customers will still cough up for the replacements whether they need them or not. It's a dream. And it's happening right now.

Scene 4: Real Audio? You are a consumer who listens to music. You have a number of music systems comprising a good selection of equipment. Nothing too flash, but they're well designed, reliable, and they get the job done with a minimum of fuss.

The larger equipment manufacturers have decided that they want more of your cash. So they mount an extensive campaign to persuade you that you require a new audio data format. Sleek and sexy, this format is incompatible with the previous format, and it requires new equipment to play it in the same environments as before. You would prefer to spend your money on something else, but you think that you have no choice. So you dump your dependable, reliable hi-fi equipment (which has years of life left in it) and replace it with newer, more expensive systems that in essence perform exactly the same task.

Unfortunately, just as you fork out thousands of pounds-dollars-euros-shekels on your new systems, the audio delivery format in your neck of the woods plunges from 16-bit, 44.1kHz to 128kbit MP3. Never mind. You're buying the latest thingy whether you need it or not. And you're doing it right now.

Scene 5: Unreal Audio? You are a Director of Really Quite Big Audio Inc. You have a sleek, sexy new car funded by the sales of new, high-sample-rate audio equipment. While driving, you listen to MP3s of your favourite music. As you accelerate past 100mph, you grin. You are well aware of the benefits of the 24-96 masters from which they were derived. ■
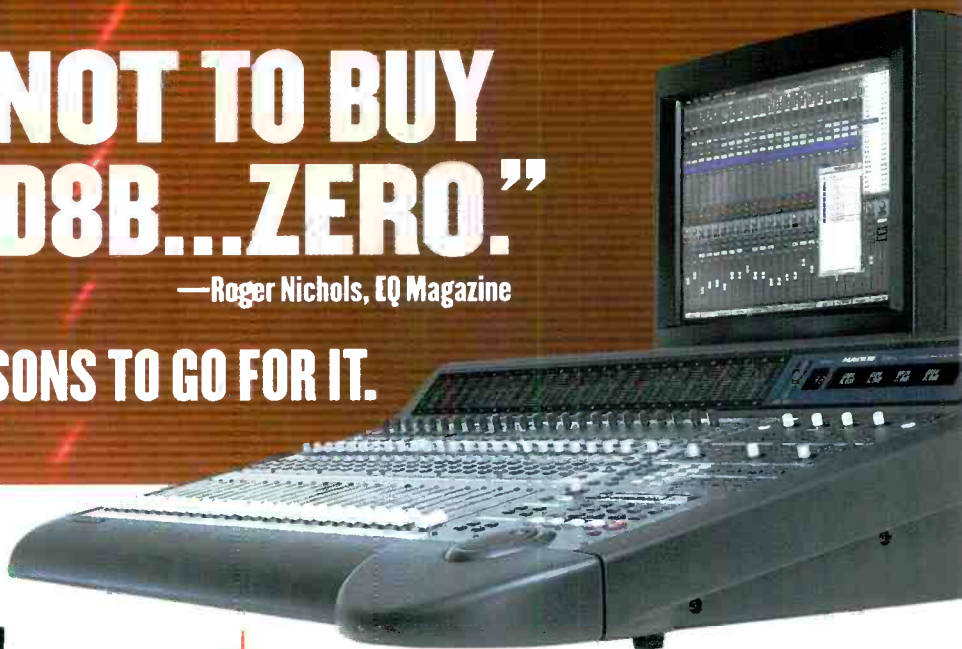
"REASONS NOT TO BUY A MACKIE D8B...ZERO."

—Roger Nichols, EQ Magazine

PLUS 3 MORE REASONS TO GO FOR IT.

1 FREE UPGRADE! NEW OS 3.0 ADDS OVER 30 NEW FEATURES!

Our Programming Department has been chugging the double lattés to create Mackie Realtime OS™ Version 3.0, packed with more new features and enhancements than you can shake a mouse at. Here's just part of what 3.0 adds to the already amazing D8B.

- **New key (sidechain) inputs** for all 48 onboard dynamic processors featuring soft knee architecture and single band 20-20k parametric EQ for frequency dependent processing such as de-essing
- **3rd-party plug-ins** via our new UFX card. Up to 16 simultaneous plug-ins on the first 48 channels, pre or post DSP, pre-fader via up to 4 UFX cards. Each plug-in is available twice — once when tracking, and again at mixdown!
- **Multiple Undo List** — 999 levels!
- New **Snapshot libraries**.
- Externally or internally accessible **inserts across Mains and Buses** plus **channel inserts pre and post DSP**.
- **Updated GUI** including 48-channel fader bank view screen.
- **Time Offset** (delay) adds a delay of up to 999 samples to the signal at the pre-DSP (dynamics / EQ) point in the signal path.
- **New surround capabilities** including depth-of-center control (LCR mixing with divergence), multiple surround panner window, individual LFE channel level control.
- **Multiple direct outs** per channel.
- **Optional level to tape** fader control.
- **Assignable, bidirectional MIDI control** of all parameters.
- **Cross patching** allows substitution of channels between various banks.

The list of top engineers and producers who use the award-winning Mackie Digital 8•Bus is growing daily. For info on the D8B, new UFX and Optical•8 cards, 3rd-party plug-ins and how D8B owners can get their free OS upgrade, visit www.mackie.com or call your local D8B dealer.

2 DSP PLUG-INS!

Antares' Auto-Tune for the D8B uses advanced DSP algorithms to detect the incoming pitch of a voice or solo instrument as it's being tracked and instantly pitch-correct it without introducing distortion or artifacts. Fully automatable.

Massenburg Parametric EQ. MDW 2x2 High-Resolution Parametric Equalizer plug-in from Grammy-winning engineer/producer George Massenburg. Mono/stereo EQ at 96kHz sample rate for unprecedented clarity and high frequency smoothness.

Drawmer offers two dynamics packages for the D8B: ADX100 includes their industry standard frequency conscious gating, plus compression and limiting; ADX200 adds variable "Peak Punch" and further Drawmer innovations.

IVL Technologies' VocalStudio provides real time vocal doubling, multi-part harmonies and pitch correction in an easy-to-use interface. A free demo is built-into the Digital 8•Bus. Just add a second MFX card to own this innovative plug-in from a world leader in vocal processing.

TC Electronic Reverb (bundled with the D8B UFX card) provides Reverb 1 and Reverb 2 algorithms from the renowned TC Electronic M2000 Studio Effects Processor. TC FX upgrade package contains an expanded set of M2000 reverbs plus Delay, Chorus, and Pitch. TC 2000 adds the TC M2000's Reverb 3, de-essing, tremolo, phasing, and panning.

3 1999 TEC AWARD WINNER!

Normally we don't name competitors in our ads. But in this case, Mix Magazine published the other nominees for the 1999 TEC Award for Outstanding Technical Achievement in Small Format Consoles: Allen & Heath's GS-3000, Digidesign's ProControl, Panasonic's WR-DA7, Spirit's Digital 328 and Yamaha's 01V. Thanks to all who helped us win this prestigious award.

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Aphex Thermionics™ Model 1100
Discrete Class A Tube Mic Preamp with 24 Bit 96kHz A/D

This superbly crafted new breed of advanced tube microphone pre-amplifier reaches far beyond mere technical excellence to deliver sound that is uniquely involving, compelling, and real. Our incredible new 2 channel Model 1100 gives you up to 20dB more headroom than conventional preamps, allowing you to record hotter tracks with the highest possible digital resolution. This unprecedented amount of headroom, combined with an EIN of -135dB, allows you to take more gain without the pain of overload distortion or noise.

MORE GAIN-NO PAIN

To achieve this amazing performance, we developed new proprietary circuitry - here's a taste...

- **Discrete Class A Bipolar PNP Solid State Front End**
- **Second Stage Reflected Plate Amplifier™ Tube Circuit**
- **Sweepable Low Frequency Cancellation Filter (LoCaF™)**
- **MicLim™ limiter on the microphone itself makes the 1100 virtually crash proof**
- **Drift Stabilized™ A/D Circuitry eliminates the need for high pass filtering in the digital domain**
- **Third Stage Reflected Plate Amplifier Tube Circuit Discrete Class A Impedance Balanced Output Stage**

MORE GAIN-NO PAIN

Compare this to any other mic preamp in the world - you'll quickly appreciate the benefits of 'More Gain with No Pain'.

For more information visit our web site or contact us for an in depth brochure.

APHEX SYSTEMS

Improving the way the world sounds℠

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