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JIM
REEVES
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  - No head wear or media degradation during repeated use, long shelf-life media

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  - 24-track punch in/punch out, no need to transfer to an editor for basic edits

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**R-1 MULTI-TRACK RECORDER**

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**Replaces real-to-reel 24-track recorders**

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The hammer

YOU'RE NOT ALONE. It is too easy to become blinkered and preoccupied by constant reminders of the neglect of the audio quotient within the much wider context of picture driven productions and to overlook the fact that vision professionals and their industry are not without problems of their own. This was pulled into sharp focus for me with the outcome of a panel session I sat in on at the recent IBC.

We can caw on about the lack of appreciation and importance attributed to audio within broadcast or film release but the other side of the equation has its own voices of dissent. You'll hear talk of the computer industry leading the broadcast and production industries by the nose, talk of the lack of standard consensus leading to hijacking by those with the most money and the most aggression, and talk of dissatisfaction with the progress and evolution of system solutions once users have bought into them on a grand scale.

There is clear objection to the fact that the majority of leading-edge technology whizz-bang picture boxes—and there are now very many—concern themselves with the generation of special effects and object generation, processing and animation despite the fact that the greatest bulk of programme material still concerns itself with the altogether more mundane production requirements of news, documentaries and live action. They resent it, even though they acknowledge that it is more exciting and more creative, because they believe it is unrepresentative of real-world, day-to-day. Most of all, they dislike the imposition of tools that are not so much specific for a task as an adaptation of technology that happens to exist and is not better than what it attempts to replace.

A quote I particularly liked was the observation that if the only tool you have is a hammer, it is strange how quickly everything starts to look a bit like a nail.

Perhaps the most damning comment was the selection of the developments made by the very few remaining film-oriented manufacturers as the highlight of IBC's technological extravaganza. It all sounds very familiar and analogies are simple to draw. No, we're not alone. We're actually fairly well sorted.

Zoen Schoepe, executive editor

Trust matters

WITH BILL CLINTON testing the American peoples’ trust in their President on the world stage, it seems timely bring this up.

For where Bill Gates went up against the American Anti-trust laws, he'd have found the Monopolies and Mergers Commission, and the Trading Standards Association serving the same purpose in the UK, and similar setups in other territories—their common purpose being to ensure that business is conducted in as 'fair' a manner as possible. For example, the BBC is restricted in its use of its own television channels to promote its activities outside of television on the grounds that, since the BBC is a public service broadcaster, nobody else can use it for advertising.

In operation, these mechanisms are usually brought into play by those believing themselves to have been wronged by a competitor. And Sony's NYC studio complex found itself in exactly this position when it considered installing a Sony OXF-R3 digital console before it was available on the open market. I'm not about to take sides over the matter, and Sony's solution to the situation is a matter of record—but there is another side to 'unfair' competition.

Sometimes it takes a rich corporation to fund aggressive R&D, and to accept the consequences when it's got it wrong, since there is a low ceiling on the risk a small concern can take. So Sony is likely to feel intimidated by their weight, without the Sonys, Philipses and Matsushitas of the world, we might find our business a less exciting place to work.

Having recently written on the value of peoples' passion—over commercial vine—in driving pro-audio forward, it seems to me that, if Sony wants to risk its cash and reputation on a pretty revolutionary desk, I'd like to sit back and watch. Easy to say, I know, and I don't know how the law might be better conceived, but I am concerned that its use could cost us more than it pays.

Tim Goodyer, editor
Not Just
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E-mail: sales@solid-state-logic.com
http://www.solid-state-logic.com
Multichannel Sound Forum

France: Studio Sound has given its support to next month's First International Multichannel Sound Forum (5-6 November) which will be held in Paris during the course of the French SATIS national exhibition held 5-6 November at Paris Expo, Porte de Versailles.

Organised jointly as a fringe event by SATIS and Radio France, it aims to bring together manufacturers and end-users to encourage education and exchange.

Copyright co-op

US: Following news of Microsoft's collaboration with Matsushita over the future of broadcasting along with that of domestic computers and the Internet, comes a deal between IBM and NTT on DVD piracy.

The companies' collective interest in the new delivery format and expertise in IT technology has caused them to combine their watermarking efforts in an attempt to establish an industry standard—replacing present systems such as CSS and APS with a watermark and corresponding circuitry for inclusion in domestic DVD players. The results will be presented to the prevailing GISTWG group in due course.

Dolby down under

Australia: The Australian Digital Terrestrial Television Broadcasting selection panel has declared Dolby AC-3 its preferred system of audio encoding for broadcast.

The announcement is expected to speed the introduction of HDTV broadcasting to the territory. The selection of the DVB-T modulation system follows evaluation exercises undertaken by Australia's Government Communications Labs on behalf of the DTTB, and contrasts with the European DVB standard by combining MPEG video with AC-3 coding.

It is, however, in line with the US, Canada and Taiwan, all of which have opted for AC-3.

Speaking volumes


Comprehensively addressing the tricky matter of machine testing drivers and systems in a way that has not been available before, the book is intended to assist in the design and assessment of multiway systems for both amateur and professional applications.

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IBC round up

The Netherlands: Amsterdam again hosted the largest broadcast gathering outside of the US' NAB at September's IBC, with a predictable, but at press time unconfirmed increase in attendance. It was a show of two halves: a noisy and rather sensational half, which成功地 exceeded the NAB's encountered at the PLASA pro sound and light show held in Earl's Court London earlier in the month, and a quieter and more civilised half which incidentally also contained the majority of the audio representation.

The IBC organisers should attempt to curtail the distressing and inappropriate increase in volume at what is after all a broadcast event with the picture companies emerging as the offenders. If anything the non-stop and largely distorted and unintelligible demos from the new audio manufacturers served to underplay the importance of good audio in all related disciplines.

IBC saw little that was truly new in the audio sphere. The event is relatively late in the calendar for the audio manufacturers who now seem to have their R&D cycles synchronised for Spring launches followed by world tours of the products.

Nevertheless it remains an important and extremely popular one-hit exhibition that has consistently grown in strength to strength.

Zeven Schoepet

UK: The 'biggest dance show ever staged' in Britain showcased Riverdance star Michael Flatley in his last performance as Lord of the Dance, Feet of Flames. To capture the Irish magic, the Fleetwood mobile joined the 84 supporting dancers, 25,000-strong audience, and 25 television cameras necessary to cover the massive stage set for the broadcast and video recording. Keith Mayes, Tim Summerrayes and Ian Dyckhoff piloted the 72-input Fleetwood Euphonia CS2000, and a pair of Sony PCM-3348 machines were in attendance to record their efforts. The project is currently being mixed at Air Studios in S.1 surround.

■ IBC has taken delivery of a Cockwood Mastering Bricks console. Installed in a new flagship mastering suite, the console includes a custom 2U-high vu-metering unit that has been incorporated in the Cockwood catalogue.

■ LA's Disney, Warner Brothers and Todd AO film facilities have bought AMS Neve Logic Digital Film Consoles, while Paramount has added 40 channels of Uptown System 990 moving fader automation to its Randirk Symphony LN and upgraded the Neve VR Legend in Scoring Stage M to handle 96 inputs (making it the world's largest VR-series desk). Disney's 88-ladder, 200-input, 3-operator DFC is destined for Buena Vista Stage A and will be used for feature-film dubbing, while Warner Brothers has ordered 96-ladder 250-channel DFCs for its Hollywood Stage D, Burbank facility, and new Stage 8 all for feature film dubbing and Todd-AO has ordered a 3-operator, 100-ladder desk for film recording at Stage A. AMS Neve, US.

Tel: +1 818 753 8789.

■ Uptown Automation, US.

Tel: +1 410 381 7970.

■ Telecoms, Melbourne has contracted Barco exclusively to supply ETSI MPEG-2 140Mbit/s and 155Mbit/s codecs until 2001. The deal, worth US$ (more than 200 RE3400 34Mbit/s codecs, which were used on the Commonwealth Games broadcasting, provides the Malaysian Ministry of Information is to implement digital TV transmission by 2000.

Barco, Belgium.

Tel: +32 56 233450.

■ Louis's renowned Pederenales music recording studio has installed a 48-channel SSL SL4000+ console with Total Recall. Co-owned by Walid Nolen and nephews. the facility was built in 1976 for country music star Nelson's exclusive use but now operates commercially for clients such as Neil Young and Phoebe Ramone.

SSL, UK.

Tel: +44 1865 842300.

■ America's Armed Forces Radio and Television Service Broadcast Center is set to update its automation system with a Broadcast Electronics AudioVault system and Automation, Gainesville, Florida. Located at the North Air Reserve Base in Quincy, Illinois, the AFRTS' BC serves over 800,000 US servicemen and their families in more than 100 countries and on US Navy ships.

BE, US.

Tel: +1 217 224 9600.

■ Ohio's Acoustic Music studio has recently completed a new mixing and mastering suite containing a 12:8 Audioscience Grandson 110B console. Atox NK-faller Pro-2400 monitor system, Designig Audio Media 8 Tascam DA-88DA digital RCM, and Sonic Solutions AES/EBU optical converters.

Acoustic Music, US.

Tel: +1 440 775 3681.
October
12–November 6
ITU Plenipotentiary Conference
Minneapolis, Minnesota, US.
Tel: +1 22 730 5969.

14–18
2nd Expomusica 98
Fira Mapa, Santiago, Chile.
Contact: Juan F Moreno.
Tel: +56 2 231 6515.
Fax: +56 2 231 4981.
Net: www.puntodez.cl.

20–24
Telev. Kino, Radio Technologies
Exhibition Centre, Sokolin, Moscow, Russia.
Contact: Ekaterina Zotova.
Email: man@admit.ru

22–25
Reproduced Sound 14
‘Surrounded by Sound’
Hydro Hotel, Windermere, UK.
Contact: Institute of Acoustics.
Tel: +44 1727 848195.
Fax: +44 1727 850553.
Email: Acoustics@clus.ucl.ac.uk.
Net: www.essex.ac.uk/ioa.

27–31
Broadcast India 98
World Trade Centre, Mumbai (Bombay), India.
Contact: Manoj Meac Samar.
Trade Fairs & Exhibitions.
Tel: +91 22 22 15 1396.
Fax: +91 22 215 1269.
Email: samac@bom2.vsnl.net.in.

27–28
DVD Conference Europe
Fira Palau de la Musica, Barcelona, Spain.
Contact: Cheryl Aggett, Understanding and Solutions.
Tel: +44 1582 607744.
Fax: +44 1582 477303.
Email: U8@undandsol.co.uk.

31–November 2
15th International AES Conference
‘Audio, Acoustics, and Small Spaces’
Scanticon Conference Centre, Snekkersten, Copenhagen, Denmark.
Contact: Jan Voetslenn, Delta Acoustics & Vibration.
Tel: +45 45 93 12 11.
Fax: +45 45 93 19 90.
Email: aes@delta.dk.

November
4–5
23rd Sound Broadcasting Equipment Show (SBES)
Hall II, National Exhibition Centre, Birmingham, UK.
Tel: +44 1398 323 700.
Fax: +44 1398 323 780.
Email: info@sbes.org.uk.
Net: www.w-way.co.uk/~dncl/sbes.htm

3–6
SATIS 98
Contact: Alexandra Tholance Conseil.
Tel: +33 1 45 45 65 25.
Fax: +33 1 45 45 65 35.

4–8
News World 98
Fira Palace Hotel, Barcelona, Spain.
Contact: News World Exhibition.
Tel: +34 4 717 49 0880.
Fax: +34 4 717 49 0990.
Net: www.newsworld.co.uk

5–6
First International Sound Multichannel Forum
Tel: +33 1 45 45 65 25.
Fax: +33 1 45 45 65 35.
See panel right.

17–19
Digital Media World 98
Wembley Exhibition and Conference Centre, London, UK.
Contact: Digital Media International.
Tel: +44 181 995 3632.
Email: digmedia@diilas.co.uk.

20–23
20th Tonmeister Convention
Municipal Hall, Karlsruhe, Germany.
Contact: Ernst Rohlf, Banglungsverk.
Des VDT.
Tel: +49 2204 23959.
Fax: +45 45 65 2080.
Email: vdt@tonmeister.de.

25–28
Apple Expo 98,
Total Design Technology
Olympia, London, UK.
Contact: Liz Scriven, Showcase Communications.
Tel: +44 171 381 2442.
Email: appleexpo@sol.com.
Net: www.apple-expo.co.uk.

December 98
9–10
Cable and Satellite Asia 98
Singapore International Convention and Exhibition Centre.
Contact: Reed Exhibitions.
Tel: +65 299 8992.
Fax: +65 299 0913.

9–11
5th Broadcast Cable
& Satellite India 98
6th Comms India 98
Programming, Multilingual New Delhi, India.
Contact: Mr Bhavuk Savioz, Exhibit India.
Tel: +91 11 463 8680.
Email: exhibition@inds.virl.net.in.

1999
March
3–7
MusikMesse
Prolight & Sound
Frankfurt, Germany.
Contact: MusikMesse Frankfurt.
Tel: +49 69 7575 6130.
Fax: +49 69 7575 661 1.
Net: www.musikmesse.de.

April
10–12
16th International AES Conference ‘Spacial Sound Reproduction’
Arklow, Kowenin, Finland.
Contact: Jukka Backman, Nokia.
Tel: +358 (005) 91 410.
Fax: +358 (005) 5378.
Email: aes@acoustics.hut.fi.

13–15
PLAS Light and Sound Shanghai.
Institut Shanghai, 88 Loushangan Road, Shanghai, China.
Contact: Marcus Renne, P&O Events.
Tel: +44 171 370 8221.
Fax: +44 171 370 8143.

May
8–11
16th AES Convention
MOC Centre, Munich, Germany.
Contact: Martin Woehr, Bayrischer Rundfunk.
Studiorproduction.
Tel: +49 89 59002434.
Email: o6@chamain.eis.org.

10–15
21st Montreux International
Television Symposium and Technical Exhibition
Montreux, Switzerland.
Contact: Patricia Savoz.
Tel: +41 21 963 32 20.
Fax: +41 21 963 88 51.
Net: www.montreux.ch/symposia.

July
8–10
11th PALA 99
Singapore International Convention and Exhibition Centre (SICEC).
Contact: Ann Tian, IRF Exhibitions.
Tel: +65 227 0688.
Fax: +65 227 0913.
Email: ann@irx.com.sg.

October
8–17
Telecom 99
Palaprobe Geneva, Switzerland.
Tel: +41 22 730 5969.

November
2–3
24th Sound Broadcasting
Equipment Show
NEC, Birmingham.
Contact: Point Promotions.
Tel: +44 1398 323 700.
Fax: +44 1398 323 780.

November 5–6
First International Sound Multichannel Forum
Organised by SATIS and Radio France with the support of Audio Visual Cable and Satellite Laboratories and Solid State Logic.
The Forum will open within the SATIS exhibition on 5 November and the presentation of multichannel TV, radio, and record productions will take place on the Masion de Radio France in Studio 105. Each presentation will be commented on by Radio France and invited radio and TV stations. Shuttles are available between the SATIS exhibition and Radio France. The SATIS Sound And Image exhibition will be held 3–6 November at Paris Expo, Porte de Versailles, Hall 4. Opening hours: 10:00 to 18:45 with a 22:00 late night on 4 November.

Programme
Thursday 5 November
SATIS exhibition
10:00 Presentation of the Forum and discussion of multichannel sound recording and mixing
12:00 Screening of the CST test tape for audio control of TV programmes
13:00 Radio France Studio 105
17:00 Welcome to participants by Francois Rochicoli, technical director of Radio France. Surround demo of NHK production by Kimio Harasaka, technical director.
19:00 Debate
Friday & Saturday
Radio France Studio 105
10:00 Surround examples of German and Austrian productions by Gunter Theile, head of audio systems, IRT.
11:30 Surround examples by Radio France.
12:00 Debate
Radio France Studio 105
15.00 Visit to digital studio 103.
Surround demo examples of international productions
17:00 Surround demo of Danish Radio productions by Lars Christensen, sound engineer and member of EBU multichannel audio group.
18:30 Surround examples by Radio France.
19:00 Final debate and conclusion.

Cost
1 session: FF400
1 day session: FF800
2 days FF1000
A whole-day registration includes lunch.
Contact
Tel: +33 1 45 45 65 25
Fax: +33 1 45 45 65 35.

October 1998 Studio Sound
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USA • TRANSMERICA AUDIO GROUP • TEL: 805-241-4443 • FAX: 805-241-7539
JAPAN • TEAC JAPAN • TEL: 0422-52-5091 • FAX: 0422-52-6783
GERMANY • BEYERDYNAMIC • TEL: 0049 7131 617 0 • FAX: 0049 7131 60 459

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Ribbon blues

JOHN ANDREWS' comments (Letters, Studio Sound, July 1998) prompted me to pull four well-known (and loved) classic ribbon microphones from my collection and measure the lengths of their ribbons. The result was as follows: BBC AXB1—2½ inches, RCA 44BX—2½ inches, STC (Coles) 4058 & RCA 77DX—1 inch.

Memory does play tricks on us old codgers which, no doubt, accounts for the 100% exaggeration of a 5-inch long ribbon which would indeed sag under any conditions. However, while I had the covers off, I looked carefully but could not detect any sagging on my particular examples of these microphones.

Having asked other engineers for their experience on this subject, I will concede that it is advisable to store ribbons in the vertical plane (or on edge) for long periods of time.

But I do think that Mr Andrews missed the point I was trying to make, that undue bureaucratic pedantry can discourage us from necessary experimentation.

Malcolm Addey, New York City

On the case

THE PICTURE of Neil Hillman's flight cases at the United Baggage Claim at O'Hare Airport in Chicago (Studio Sound, June 1998) was a sight for sore eyes. Now the whole world knows the glamour of location production for video—I've been collecting my own gear from that baggage area three times in the last month.

More to the point, I'm glad Studio Sound is branching out a little to show what location recordists are up against and I couldn't agree more about stereo field recording. I've had to educate some clients about the advantages of stereo and have had others who asked why we didn't do it sooner. The biggest problem is not in recording M-S, X-Y, or any other method of stereo recording, it is getting a budget to include time in an audio suite to take advantage of what's been recorded. Since X-Y can be accommodated easily enough in an Aud suite, that's often a far as it goes. However, I agree with you that it's a good fight worth the effort, and the reward is an inexpensive way to greatly increase the production values on a show.

Bruno Strapko CAS, Strako Recorders, Schaumburg, Illinois, US

The stereo scam

I FOUND John Watkinson's article (Studio Sound, July 1998) on microphones particularly interesting. His explanation of co-incident microphones was succinct and well done. This got me to speculating on why it is that the vast majority of professional classical music recording engineers prefer to use some form of spaced microphones. This is in spite of a long tome by the brilliant Stanley Lipshitz published some years ago in the Audio Engineering Society journal, citing the evils of using spaced microphones and praising the merits of co-incident microphones.

The term 'fat mono' is often applied to co-incident techniques by the spaced microphone people, and M-S seems to stand for Maybe Stereo. Apparently the mathematical elegance of Blumlein's invention escapes them. Perhaps a clue as to why some recording engineers prefer spaced microphones can be had from the mention in John's article to direct, early reflected sound and reverberation in relation to stereo. Of course, we all know this is a useful but gross oversimplification of what is really happening, thus the clue to that preference for spaced microphones might be hiding in that oversimplification.

Although our hearing system utilises the direct sound to locate a sound source, that is not the end of the story. A musical instrument radiates in an incredibly complex way, the fundamental and the various overtones emerge in a multitude of directions—then these combine with their individual reflections from all six boundary surfaces before they reach the listener, this total sound being the true timbre of the instrument. The brain uses its cognitive sense to analyse all of the incoming sound for purposes such as the enjoyment of music.

As John says, spaced microphone techniques result in a spatial effect that some people find pleasing. Maybe the spaced microphone advocates think it more nearly represents the timbre or three dimensional radiation patterns of musical instruments. After all, everyone knows that ersatz stereo can be synthesised from mono by simply delaying the sound to one channel by 20ms, isn't that built-in when you use spaced microphones?

It is no secret that the best recording techniques in use today cannot rival the experience of a live event. Michael Gerzon once stated it would take a million channels to do it and some years ago Sir Thomas Beecham, at an event Columbia Records gave celebrating his 30 years of recording, told his audience that they has been 'engaged in a fantastic public swindle', and that their records didn't sound anything like his music.

What I think, this points out is that neither spaced nor co-incident techniques are adequate for 21st century technology. DVD and surround sound are almost here and a industry-wide dialogue should begin on how best to utilise multichannel sound for the home listener—before we plunge into another disaster like quadraphonic sound.

New delivery channels, including DVD, satellite, cable, digital TV and the internet, are providing an explosive increase in the number of routes available to deliver material to an ever-more enlightened audience demanding complex levels of audio format.

This in turn has created a requirement for powerful audio tools capable of generating and controlling these significantly more complex formats, effortlessly combining and distributing the increase in the numbers of audio channels.

Manual operation is no longer an option.

Are you ready for this?

People who work with Soundtracs DPC-II Digital Production Consoles are:

They’re assured of 150 automated channels with comprehensive digital audio processing controlled by an intuitive worksurface.

They’re also dubbing in all current and known future mix formats in 24 bit and feeling secure with the knowledge that future proof 96kHz resolution and 7.1 operation is built in.

And they’re saving time by opening up creative possibilities with each new project they produce.

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The PreSonus M80 Eight Channel Microphone/Preamplifier with Mix Bus.


- Greatly improve the quality of any console, analog or digital, by adding eight very high end mic preamplifiers.
- Amazing front end for two, four or eight channel analog to digital converters such as the Digidesign 888® or Soundscape SS810-1®.
- Add warmth to any microphone using the proprietary FET IDSS adjust.
- High headroom summing bus allows true stereo imaging utilizing more than two microphones.
CreamWare SCOPE and Pulsar

After the success of its tripleDAT family CreamWare is boldly going into new realms with its Pulsar and SCOPE cards. Simon Trask previews the next generation, in Germany.

Over the last weekend of August, German company CreamWare chose to preview its grand new design for the computer-based digital studio era in a grand design from a former age. Recently purchased by the company as a base for its development activities, the long-disused Stockert radio telescope is still an imposing edifice situated high amid hills in remote countryside and visible for miles around. The original opening of the telescope in 1956 ended the ban on radio wave measurement in Germany following the Second World War, and was considered an important enough event for representatives of the Allied forces to attend the ceremony.

It was a somewhat more modest gathering of journalists and distributors who attended the preview of SCOPE and Pulsar. CreamWare heralds SCOPE as marking 'a new milestone in the way we generate and produce audio in the digital age,' and describes its new system as: 'a complete studio design environment,' and simply, 'the studio of the future.' Bold statements indeed. There is no denying the ambition of these latest products from the company. With SCOPE (or 'Scalable Object Processing Platform'), CreamWare sets out to combine synthesis, sampling, recording, editing, mixing, effects and multiple I/O on a single card; Pulsar is essentially a scaled-down version of SCOPE, offering mixing, effects, sample playback, predefined synthesisers, and multiple I/O.

Both take the form of DSP-based PCI cards for Windows 95/98 computers; the heavy-duty audio processing work is done on the cards by the latest Analog Devices SHARC DSP chips, while frontend graphical user interface work is handled by the PC. Although Mac versions of the two products have been mentioned in publicity, there were no Mac-based systems in sight at the preview. In fact, from talking to CreamWare representatives it soon became apparent that the company, which has always specialised in the PC, is concentrating its energies on getting the Windows version working properly to the Mac.

Since the introduction of SCOPE in an alpha version at Frankfurt earlier this year, CreamWare has upgraded the system with the latest SHARC DSP chips, which run 50% faster than those used in the original version, and has also upped the number of chips from 6 to 15; Pulsar, meanwhile, runs on four SHARC DSPs. The company is quoting a price of $1298 for Pulsar, and around $6500 for SCOPE—positioning the former at the upper end of the I/O card market (though clearly it has more to offer than straight I/O) and the latter in a more ambitious niche.

SCOPE is a late-nineties version of the dream of a self-contained music production system that fuelled the development of early 'computer music' systems, like the Lambda CMU, NED Synclavier and PPG Wave. And by implementing virtual synthesisers in software, complete with freely configurable graphical front panels, Creamware's SCOPE also realises the dream of another PPG product, 1980's abortive Realizer, an instrument that was way ahead of its time.

Only where all these early systems were stand-alone products, SCOPE takes the contemporary route of using a generic desktop PC to host a system-on-a-card. CreamWare is also adopting the modern-day approach of letting third parties take some of the strain, rather than trying to do everything itself. This approach ranges from letting existing products do what they do best (for example why reinvent the sequencer when you can integrate SCOPE with an established program such as Steinberg's Cubase VST?), to encouraging other companies to develop plug-in effects for the SCOPE architecture, to allowing SCOPE users to put together their own virtual synthesisers and modules from pre-existing elements, and customise the graphical presentation.

As software effects, third-party or otherwise, were demonstrated by CreamWare: instead, the company was using SCOPE I/O in conjunction with effects routing on the virtual mixing desk to integrate external effects processing into the SCOPE environment (a valid feature in its own right, as the standalone effects processor is far from ideal). By its own admission, CreamWare was showing alpha software at the preview event, not a finalised release; it would, perhaps, be more accurate to call the weekend which was billed simply as 'SCOPE Event 1998' a proof-of-concept showing. A rolling program of workshops covered system overview, studio integration, the Pulsar Modular Synthesiser, sound design, and (control) surface design. The elements being highlighted were the virtual mixing desk and virtual synthesisers. The desk provides 32 channels in Pulsar; 48 in SCOPE, complete with 1-hand EQ and four inserts per channel. The release version of the mixing desk software will include channel grouping, plus the ability to control all desk parameters via MIDI using SysEx commands—enabling mix automation from a MIDI sequencing package, and live control of the mixer from a hardware control surface.

ASIO input capability within SCOPE and Pulsar means that audio tracks...
mates, as the company is aiming to develop a new generation of digital audio recording software built on the SCOPE technology. However, SCOPE and Pulsar owners using the PC platform will be able to integrate the functionality of their boards with CreamWare's tripleDAT system to release 3.0 of the tripleDAT software. This December, interest will feature the necessary support capability.

On the virtual synthesiser front, 'Mini-moog plus', Blue Synth and Easy Synth analogue-style synths, and a versatile modular synthesiser complete with graphical patch cords were all demonstrated at the Innovation Forum, along with the functionality in SCOPE (but not Pulsar) for custom creation of synths and modules from pre-existing elements. These are not physically modelled synths or modules, but digital versions of familiar synthesis functionality; however, the sounds emanating from the synths were impressively crisp, powerful, well-rounded and dynamic.

Also shown was the Pulsar sample player, which includes pitch, filter, amplifier, LFO1 and LFO2 functionality, the initial release will support Alesis CD-ROMs only, while later releases will support additional formats. SCOPE users will be able to do their own sampling. Multiple synths and samplers can be run simultaneously, each accessible on a separate MIDI channel, with up to 16 voices of polyphony per instrument depending on its complexity; total polyphony is expected to range from 32-64 voices.

Another workshop showed how you can customise the graphical environment, from selecting colour schemes to importing graphical elements created in external design programs. CreamWare foresees SCOPE users making custom environments, along with custom synths and modules, available for download over the internet for use by other SCOPE as well as Pulsar users.

While the Pulsar board comes with built-in I/O functionality, SCOPE users will be able to add on various I/O options—one of which will be the Pulsar I/O board. This will also be possible to use multiple SCOPE and Pulsar boards together, hooked up via a dedicated TDM bus that allows DSP processors to load up to a hundred channels, each with its own hardware.

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Otari RADAR II

For a hard disk to be accepted as a multitrack replacement it must combine the skills of the DAW with the front end of tape. Dave Foister looks at the next generation

WHILE THE WORLD debates whether the tapeless studio is yet a truly practical proposition, one company has been quietly installing the great success since 1994. Otari's RADAR hard-disk multitrack now has well over a hundred satisfied users in the UK alone, yet this has been achieved with so little fuss that many (including myself) remain hazy about what the system does and how. Its image and intended market slot have been hard to define, and even the fact that entire hit albums have been produced on them (George Michael's Older being the first) has not raised general awareness of what RADAR is. Now a new version, RADAR II, is with us, providing an opportunity to take another look at the system.

RADAR stands for Random Access Digital Audio Recorder, and what sets it apart from most hard-disk-based systems is its determination to behave like a tape machine. Although there is a PC motherboard inside the box, it doesn't feel like a computer running recording software; rather it is a recorder that happens to look a little bit like a computer. And while this ideal has made further appearances since, with tapeless-8-track MDsMs from the likes of Akai and EMU, RADAR remained the only one to provide the tapeless equivalent of a 24-track studio recorder in a single box until the announcement of the Euphonix RI.

Its resemblance to a computer is only passing, although the appearance has changed with the new version. The word is a design an anonymously black box, distinguished by a big blue illuminated Otari logo, and three media drives. Here is the first evidence of the new features of RADAR II: the original had three internal hard drives to handle the audio, while II has a single removable SCsi drive available in 9GB and 18GB versions. There's a floppy drive for software upgrades, and the software itself is now kept on a small internal hard drive rather than on the audio drives as before. The final slot is a built-in Exabyte drive—an Eliant 820, the current fast version—for project backups.

Operation is carried out from a dedicated remote panel, the RE-8, which carries transport controls, track arming buttons, scrub wheel, editing controls, and many other functions directly accessed from the keys. The owner's keyboard is only there for running projects and locators, and in a way it is a shame it takes up so much space as its dominance of the control surface carries with it the suggestion that operation is more complex and computer-like than it actually is.

Above it, a new bank of keys brings some commonly used menu options to the surface, and has space for many more. Comparisons with the Fairlight MX3 keyboard are unavoidable, and like the Fairlight the layout is clear and obvious. It is important to note, however, that the RADAR controller can be operated simply as it is, with no display screen, like a conventional multitrack. The 2-channel meter bridge is standard, making the whole system complete and self-sufficient. There is a display screen, which makes editing more intuitive still, but there is nothing on the system that cannot be done without the screen. The controller is small enough to be mounted on a lightweight rack trolley, which makes it look all the more like a tape-machine remote.

ALL THE EXPECTED FUNCTIONS are here on the RE-8, and many more besides. The transport controls are small, but clear and illuminated, and while it could be said that a random-access system does not need conventional transport controls it all helps to add the required familiarity. The 'wind' speed is fully adjustable, and a double-push on the buttons gives wind at three times the set speed. This kind of programmability extends to the initiation of record, which can be set to be either a single push on the record button or a 2-button operation like most tape machines. Since the record button has no protective fence around it, this sounds a little alarming until you remember one of the features made possible by the hard disk approach—undo. A major point in disk's favour over tape is the ability to undo what you have done, whether by accident or design, and RADAR currently has one level of undo. This would allow an accidental recording to be deleted and the original restored, and also allow overdubs to be carried out on the same track without losing the previous version—if the new one is no better it can be thrown away and track comes the first one. Work is in progress to increase the undo capabilities, perhaps to 10 levels. As it stands, the undo function applies equally to the machine's powerful editing functions.

There is a whole bank of keys dedicated to the editing possibilities, and the operation centres on two keys to mark the in and out points. These select points for automatic punching when that is needed, but also select a block of audio that will have other processes applied to it. The points can be entered on the fly or numerically, and in this context it is worth noting that the time display can operate in bars and beats, given a tempo map and a start point. It also offers frames or feet in addition to the conventional time read-out.

For precise identification of the required points there is a jog and shuttle wheel, which gives one of the best emulations of analogue rock and rolling I've heard, with a directness to the wheel's action that makes its use positive and reassuring. This is not a double-function wheel in the conventional sense, with a push-button action to switch between jog and shuttle; what it does is determined by which key is pressed before you grab it. This also means it is not live all the time but must be selected before use. A big bonus on RADAR is an on-board solo function, allowing any selection of tracks to be soloed to help identify edit points and

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18
problems. Obviously this mutes the rest of the outputs so can’t be used like a console AFL system, but during track laying and editing it puts even more of the process under the hard on the recorder remote. It even offers three types of solo: momentary, additive, and interlocking.

Having selected a block of audio, the full range of editing functions is available on the dedicated keys. The block, comprising a chosen set of tracks from between the marked points, can be cut or copied on to a clipboard, which can itself be played for checking. The block can then be inserted elsewhere, used to replace existing audio elsewhere, thrown away, looped, and even reversed. Crossfades are fully adjustable and tracks can be slipped in time, forwards as well as backwards. All these actions can be carried out very quickly, and all can be undone. Nothing is destructive, although with only one level of undo it is possible to get so far away from your original edit that you will have trouble getting back, but the project management functions mean even this is no problem.

The system can handle up to 99 projects, each with 99 locator positions that can be marked up as easily as the edit in and out points, to bars and beats if appropriate, and named using the keyboard. The power of the system is that a project can be copied in its entirety, duplicating the complete EDI, but not the audio tracks. Since nothing is destructive, like a conventional DAW, several completely different versions of the same piece can be created without ever losing any of the others and without having to copy any audio. There are already tales of people buying a RADAR system, hanging on to an existing DAW on the assumption they will be needing to fly stuff out to it for edits and spin it back in as with conventional tape, and then finding their DAWs have become redundant as RADAR can do it all with less fuss.

A good example of the kind of thing it can do is the idea of doing your mutes on the source tracks instead of on the console. Chunks of audio can be accurately identified and then ‘erased’ very quickly, without actually permanently removing anything; doing this on a copy of the project, coupled with the undo capability, makes it fast, flexible, and safe, and probably more accurate than most console automation.

This kind of work is inevitably made easier by the flat colour-LED display screen. Panels around its edge show various items of status information, such as sample rates, remaining space, and so on, and while all these can be viewed on the IE-8’s display it is also useful to have them all visible at once. But the screen is mostly taken up with the tracks display, a powerful aid to managing the audio and editing functions. Audio on the 24 tracks is shown as pale blue strips which scroll across the screen; the current position is the centre, and contrasting shades show the area defined by the in and out points. Future plans include showing the audio waveforms in the blocks, but even without this facility, scrubbing and locating is very intuitive.

Audio can be shared between different projects, and even the process of copying a section from one and pasting into another is quick and straightforward. The project management tools are smart enough to flag the fact that you have done this, so that when the original project is deleted, the audio needed for the later one is retained.

The fact that RADAR II is 24-bit is an important reminder that this is not a system that appeals purely on the basis of operational convenience. The audio quality of RADAR has always been a source of some pride to Otari, with an enormous proportion of the cost of the original model accounted for by the converters. Crystal semiconductor converters have been used from the outset, and according to Otari there are users who bought systems for the features and found they ousted their established multitracks on the basis of the sound as well. RADAR II builds on this reputation by incorporating 24-bit converters, although 16-bit operation is still selectable where appropriate.

The rear panel is crammed with connectors showing the unit’s potential for interfacing with almost anything. Analogue and digital signal connections conform to Tascam formats, with TDIF digital and balanced pro-level analogue on 25-pin D connector, so a RADAR box can be hooked directly into a studio already wired for TDIF machines. Any pair of channels can be assigned to AES-EBU and SPDIF connectors, and similarly signals coming in on these can be routed to any tracks.

The SCII bus appears on the back for connection of further drives, and seamless recording across these is possible. Synchronisation handles all types of time code, plus MTC and full 9-pin machine control. Chasing is simple and reliable; this is not a lock-and-let-go system but chases code constantly, even apparently staying with dodgy sequencers without compromising performance. Word clock in and out are provided, alongside a video-sync input complete with loup-through and termination switch. This kind of detail shows that RADAR is intended to be a fully professional machine with applications just about anywhere multitrack recording is needed.

The success of this aim is shown by the variety of installations already using it; although the system’s low profile remains a source of frustration to its adherents—there seems to be a reluctance to take it seriously in some areas, perhaps born of scepticism about whether it can actually do what it says. For myself, soon after I was introduced to it, I felt completely at home and started to realise that perhaps it offers the best of both worlds, combining the simplicity of tape with the power of hard disk. RADAR II may be the industry’s best-kept secret, but there are an awful lot of facilities that would be doing themselves a favour by investigating it.
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Merging Technologies Pyramix v2.0

Recently upgraded and supported by new ancillary products, the secret of the Pyramix is unravelling. Terry Nelson peers inside a box of Swiss tricks.

PYRAMIX FIRST ATTRACTED attention some two years ago with the release of v1.0 software. Since then, the release of v2.0 software has built on the system's considerable power and flexibility, yet Pyramix falls into the 'best kept secret' category.

As previously reviewed, Pyramix is a complete virtual studio running on Windows 95 and NT (recommended), and comprises a flexible mixing console, an 8-track recorder, a pool of DSP units and a CD-ROM reader, facility to RedBook standard. It can either be supplied as a complete rackmount package with computer, or as the Kefren ISA card with the software on CD-ROM. In this case, I installed the Kefren card in a Pentium II (266MHz), and this was accomplished in the time it takes to open the computer case, plug the card into the socket and reassemble. Before loading the software, I recommend that you set your graphics to high resolution for viewing comfort. At that time, the computer was running Windows 95 and has since been upgraded to Windows 98 without any problems.

Inserting the CD-ROM starts an auto-run procedure where you are asked what you want to install.

As well as the Pyramix software, you also get the operation manual, Acrobat Reader 3.0 and Internet Explorer. The user manual is well written and quite comprehensive. However, the Help File is also very clear and this would be sufficient in many cases.

As one of the advantages of being a fairly small company, Merging Technologies has a good dialogue with users, and much of the fine-tuning of the system is the result of feedback from customers using Pyramix on a day-to-day basis. Fairly obvious, you may think, but the turnaround time is often very fast, and this is illustrated by the fact that I started with Pyramix earlier this year with v1.2. soon to be followed by v1.3, and now I am up to v2.0.

It is clear that the evolution of the system is resolutely towards AV applications and mastering as compared with the original 8-track studio-in-a-box approach. This is shown by the speed at which it is possible to carry out edits, the assembly of recorded material, with an easy operation. The first software release allowed comprehensive recording and editing facilities, the latest version offers improvements including dynamic automation on all functions including EQs, pans and DSP effects.

For example, a slight adjustment to the EQ of the music background during the voice-over of a commercial might just give that edge to the voice without having to juggle with levels. Another much requested feature, crossfade editing, has been implemented and can be...
tions can be marked and raised or lowered by the necessary amount.

A clip function allows you to see how near to zero level the signal is at its highest peak. If, for example, the loudest peak is +7.5dB, you could raise the overall level by +5dB in order to maximise the signal while secure in the knowledge that you will not clip. It is clear that this feature of gain adjustment could be a global to a whole project, if required, and not just to defined sections.

Certain aspects of the information displayed on-screen have been refined, and among the more notable are active track display and what could be called the overall project display. With the former, it is now possible just to display the tracks with recorded material on the screen rather than a mixture of recorded and empty tracks. This helps clarity of working and avoids undue clutter. The lower edge of the main Pyramix edit screen shows the overall project in a condensed form, which allows zooming in on any point of the project without having to do much searching. This now displays in red the part of the project that is currently active on-screen, a small point, but very handy. Other display features include a fine zoom control that gives a resolution in fractions of a sample. Whereas the this may be seen as being a little excessive, it really does allow you to line up all tracks right on the button for perfectly synchronised starts.

Synchronisation is an important feature of Pyramix, and the unit features full time-code implementation plus machine control. The buttons for the latter fall nicely into place with the right-hand numeric keypad of the computer keyboard, and makes for easy and logical operation. Various sync modes are available including hard and soft chase, where Pyramix will either resync to jumps in incoming time code or stay on internal sync.

A nice feature, harking back to the Nagra-T analogue recorder, is the ability for the machine, in the sync delta display. This provides a visual display of when the system is coming into sync as well as any variations. The default setting is 8 frames, but a zoom function allows the display to be adjusted from numbers of frames to fractions of frames. Also useful is the ability to move the reference sync point of programme material. Suppose that you have fixed your sync reference to start the first cue for a commercial. You now decide that you want things to start earlier or later according to the picture cue by marking this point and making it the new reference sync position. To let you know when you are actually recording Merging Technologies has included an eye-catching on-air logo complete with the graphic of a well-known microphone to appear on the screen. Another small point but one to go down well with studio clientele.

Projects can be saved to CD-R and this provides a neat way of playing back a completed project to a client at short notice. As well as the programme material, the CD-R also records all setup information which means that a project can be quickly loaded and reworked with the minimum of setup time. Pyramix works with most popular CD-R recorders, and provides CD mastering facilities to Red Book standards. The same composition editor is used for marking CD start, stop and index points as for projects, and the real-time processing of the DSP facilities—together with the automation—provides a very complete mastering package. Depending on the final destination of the CD-R, useful features include post roll (broadcasters may want 10 pre-roll whereas CD masters 40ms), digital copy inhibit, readout of track time or total CD play time elapsed, SRC for setting wordlength (multimedia productions), and import and export of various file formats.

In parallel with the development of Pyramix, Merging Technologies has developed a range of hardware and stand-alone products to complement the system. One of these is the new Keep PCI bus card. This supports 16 simultaneous hard tracks with 32kHz-96kHz operation. Version 2.0 Pyramix software already supports Keep PCI, so upgrading from the Kefen card to Keep PCI is plug 'n play. The other two principal items are the DUA convertor-breakout box and the Sphynx modular convertor. The DUA has been designed as a dedicated A-D/D-A convertor for the Pyramix, and consists of two sections, each containing all the necessary A-D/D-A and AES-EBU I/O circuitry, and plugs straight into an ISA bus PC slot. This is possibly the first time that an A-D/D-A convertor is available to plug directly into the computer, due to the extremely hostile environment for audio (lots of 'dirt' flying around). The card features a dynamic range (or signal-to-noise ratio) of 100dB and this is improved by only 0.3dB when using it externally with the computer.

The DUA provides 4 analogue inputs for 20-bit A-D conversion and 6 analogue outputs of 20-bit D-A (outputs 5 and 6 being for a headphone). 2 AES-EBU inputs and outputs at 24 bits (which can also function in a dual mode for 96kHz), internal crystal lock and external AES-EBU and wordclock sync. A 1U-high breakout box completes

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the system by providing all inputs and outputs on XLRs plus headphone monitoring jack.

About to begin production is the Sphynx modular 24-bit high resolution digital audio interface. This is a 2U-high A-D/D-A convertor that has the advantage of being a stand-alone system, and fully user-configurable depending on the modules installed. The basic configuration consists of 4 analogue inputs and outputs (24-bit up to 48kHz), CPU-PSU board, 2 optical ADAT inputs and outputs and an SPDIF I/O. Other modules available include 24-bit AES-EBU inputs and outputs plus 24-bit A-D and D-A inputs and outputs—either at 48kHz or 96kHz. A TDIF interface for Tascam DA-88 will also shortly be available.

The Sphynx can either be controlled remotely via the Pyramix software, when used as part of the system, or via the front panel when used as a stand-alone. In the first case, the convor is connected to the Keftren/Keops card via the ODI optical link.

The front panel consists, from left to right, of a mains switch; remote card indicating remote control, a group of LEDs to indicate configuration; status together with silence button; a second group of LEDs to indicate sample rate (32kHz, 44.1kHz, 48kHz) plus x1, x2, x4 indicators and select button (selecting 48kHz and x2 will provide 96kHz operation); a third group of LEDs act as input level meters for the 8 channels showing signal present and overlevel; a fourth group of LEDs for the Monitor section plus select button to switch either the inputs or outputs in pairs to the headphone output. A volume control and the associated headphone jack complete the panel.

The rear panel features 2 rows of 8 XLRs for line inputs 1-8 and line outputs 1-8. A clever use of the connectors is that, depending on the configuration of the unit, the odd number inputs outputs may also be the AES-EBU inputs—outputs 1/2, 3/4, 5/6 and 7/8. Other connectors include an XLR for AES-EBU reference, three BNCs for video-word-clock (input-loop-output), two RCA sockets for SPDIF I/O and four optical connectors for the ODI-A and B inputs and outputs. The panel is completed by two blank panels marked aux 1 and aux 2, which will be used for the TDIF interface when available.

The Pyramix Virtual Studio is certainly a system to be reckoned with and it is surprising that it has not got a higher profile in studios. This said, OEM use is quite extensive and this include all of the digital audio facilities in the video workstations from Softimage. If you are looking at digital workstations for your studio to include Pyramix in your checklist. Merging Technologies is at the forefront of developments and next year will see some interesting new products.
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VFX 3D
Midias Broadcast XL4

Live sound desks are frequently pressed into broadcast duties but few manufacturers have gone the extra mile to add the bits that can make the difference. Zenon Schoepf reports on the modification of an industry standard.

While by no means a new product, the XL4 has fulfilled its expectations as the Midias flagship desk. The subtle broadcast variant of this console has now been in existence for around 18 months and has accounted for some 10% of the circa 1,604 XL4 user base.

The fact that live consoles are regularly pressed into service in broadcast environments is not surprising given that the fundamental requirements of both applications—major strongly on the objective of the stress-free single-take, be it for live transmission, live recording or live-to-the-air—usually served by the broadcast manufacturers still swear by this standard.

For decades the XL3 was to bring it more in line with more traditional consoles through the inclusion of aux patches and audio subgroups. In fact, you are offered 24 auxes and 16 audio subgroups. EQ on the XL4 is handily parametric rather than the slightly more restricted arrangement on the XL3, even though you rarely hear anyone moan about it.

A quick roundup of the XL4’s CV is in order. It is a 16-bus desk with an additional 18 x 8 matrix comprising, 16 mono auxes, 16 audio sub., 4 stereo auxes, and stereo master. AFL and PFL buses. Inputs up to 36-channels are supplemented by an additional 16 aux line returns. Other additions over the XL3 include a direct output on each channel with a level control, and the meterbridge houses the mic pres fed by separate mic and line XLRs selected by an automated switch.

Indeed, automation on the XL4 is a leap forward over what has been offered by the company before and includes a snapshot fired channel mic-line switch as already stated, phase reverse, EQ bypass, all aux cuts, insert, assign to stereo and mutes switch automation.

The automation on the desk is the result of a collaboration with the UK’s Outboard Electronics, a connection that started with the addition of automation to the XL3. According to Midias the XL4 was not originally intended to be automated; although a change of mind resulted in the fairly advanced level of automation that the desk now sports. Snapshot scene creation and recall is simple, governed by a plain-looking master panel.

Storage is to internal memory and removable PCMCIA card.

A neat inclusion in the fader-panel section of channels is a collection of group-assign switches which can, via global mode switches located in the automation control panel, be used to assign audio subgroups (16), mute groups (8) or VCA groups (10).

It is interesting to note that the XL4 was not designed with the broadcast additions in mind, but the modifications were the result of requests from broadcast-oriented users who saw the potential of the board for their line of work. It is also significant that the knowledge gained in this exercise is likely to be applied in similar panels on existing and future consoles from the company as Midias has clearly spotted the potential of broadcast outside of its traditional live sound market. For example, a similar, but stripped down, variant of the XL4’s broadcast panels module is planned for the
It might not seem like a lot, but these thoughtful additions may be enough to change the complexion of the XL4 from a supremely able, pure live-production console into a live-production desk with a practical broadcast-specific twist.

An ordinary XL4 can be converted to the full broadcast spec and off-the-shelf. The broadcast version asks a circa 10% increase in price, but Midas is at pains to point out that not all channels need to be run motorised faders in the same way, that not all need to have remote starts, so savings can be made.

Outwardly, it is extremely difficult to spot a broadcast XL4 from a distance unless your eyesight is good enough to identify the presence of the expanded comms module. This module has two external inputs with level control and soloing works differently in that you can reverse phase, listen to both in mono, swap outputs over, dim them, mute L or R and A-B speaker destinations and a separate PFL output is supported. Meter outputs are provided and the talk mix can address external and internal with separate level controls. The oscillator has its own output.

Talk to all groups, matrix, mono master and LR. A Comms switch directs output to its destination along with the coordinating buttons in such a way that the talk mixer, headphones and PFL speaker-output can send and receive Clearcom signals, and the headsets.

The XL4 remains a hardcore console. Midas has honed its console's brooding presence to perfection making it the definitive article for live-sound engineers to impress the girls with. As such it's going to be wasted sitting in a locked-up truck.

Front to back reach is substantially less than the far less well-equipped XL200 due to the meterbridge precam mounting. Pot access and switch visibility is superb and every control has the sort of 'buffered' and 'clamped' feel that accompanies quality and would shame many a fixed studio console.

The outline of the comms module features, and other broadcast bells and whistles, will mean more to some readers than others, but in no instance do they detract from the desk's fundamental soundness and suitability to its original application.

It is ridiculously simple to use. I'd challenge any manufacturer to put forward a desk with this degree of flexibility that is this approachable. Only a novice would stumble at getting a bank load of levels to stereo, and even then, providing they had actually seen a mixing console before, they must learn quite quickly.

In live production, no matter how well rehearsed, you still fly by the seat of your pants and the last thing the station wants to encounter is surprises.

Here is an interesting statistic for you that puts the appeal of the Midas brand name into the context of broadcast. Who, would you say, is the single biggest user of Midas consoles in the world? Probably one of the super-league sound companies based on one side of the Atlantic or the other? Wrong. It is Italian broadcaster RAI, which had 19 desks at the last count. That, I think, says quite a lot.

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Focusrite Blue 300

The appearance of the Blue 300 mastering controller prompts an alternative to the usual eclectic style of mastering suite. Dave Foister is in Blue heaven

A Mastering Room is traditionally the most individual and diverse control room you are likely to find. While there are some things you expect to find in most, there are probably no two the same, partly because of differing markets, but mainly because a room reflects the individuality of its engineer. It is a bold move for a manufacturer to try to produce a whole suite of equipment to do the job, and perhaps an even bolder move for a facility to build a whole cutting room around the result. But when that manufacturer is Focusrite then the idea possibly makes more sense; and when the facility is Tape to Tape, with several highly specified rooms to choose from already, then a themed suite like this is an attractive concept that reinforces the high-end flexible image and would bring in clients.

If a single piece of Focusrite equipment is eye-catching then a console full of it has to be seen to be believed. Tape to Tape has a large room, conventionally laid out with no mixing console as such, but custom furniture fitted with rackmount bays filled with appropriate gear. The difference is that apart from the Neumann lathe controls, two DAT machines and two signal processors (a BSS de-esser and a Summit compressor, for the record) all the bays are filled with items from the Focusrite Blue range. In all there is 6.5 units of Blue, and in the middle of it all is a blue oval that for once does not say Ford; it is the remote unit for the Blue 300 Mastering Controller.

The system makes sense because Focusrite has always taken care to address the mastering market. Its EQ and compression has just the right combination of control and sonic quality to make it attractive, and the Blue 515 EQ and Blue 550 compressor/limiter are specifically mastering versions, with stepped switched controls in place of pots for precise channel matching and repeatability. The recent addition of top-flight converters in both directions completes a set of equipment that is ideal for the application and provides virtually everything the transfer chain is likely to need, with impeccable credentials in terms of its quality. The logical next step is to integrate the lot, with a central controller to select sources, insert processing and feed the monitors. This is what the Blue 500 does, with a clever combination of routing, switching and remote control. The last element in particular is intended to promote the idea of a complete Focusrite system; although the signal routing facilities could be used with any suitable equipment. The signals are handled by a 24-bit rack-mounted box with enormous LED meters and a headphone socket, and control of the system lies with the oval desktop remote unit.

Despite its forays into digital, it is with analogue that the name of Focusrite is most closely connected, and all the signal handling through the Blue 500 itself is analogue. In the context of a cutting room this makes perfect sense, as the final output has to be an analogue signal to the cutter head. For other applications the only option with digital systems is conversion to analogue, processing and routing, then conversion back again, and Focusrite is one of an elite of companies that could ask us to do that and get away with it. For those who believe that EQ and compression can only be done properly in the analogue domain, and that Focusrite does both at least as well as anybody else, the Blue 500 is surely irresistible.

The central router has inputs for eight analogue sources, and there are separate signal paths to the record output and the monitors. The monitoring control is at least as comprehensive as the average mixer, with feeds for three sets of speakers (main, mid and mini), summed and differentiated mono, polarity reverse, adjustable dim and mute. The level control is a rotary encoder with a big 100dB read-out for easy repeatability, and the three feeds have level trims under a panel on the main rack unit for matching their volumes. Also hidden under the panel are level adjustments for the eight sources, which can be +10dB or -10dB and have gains set individually for precise matching. Therefore the control panel has seamless switched attenuators for each input with overall trim controls, and a Sum mode allows them to be mixed—a suggested application is adding reverb. For overall level control there is an optional master fader to be mounted separately in the console.

Although all the inputs are expected to be analogue, it is assumed that input 1 will be fed by a Focusrite Blue 260 D-A convertor, and Tape to Tape has two of these. The suggested setup is to use one for the record chain and one for monitoring, but this is by no means compulsory. In between the two selector switch banks are six buttons marked digital, and these come into their own specifically in conjunction with Focusrite's own convertor. The Blue 260 has four AES inputs plus a TOSlink optical and an SPDIF coaxial, and any one can be selected from the front panel. The IC Controller goes one better by offering full remote control of source selection on this bank of six buttons. This means that six digital and seven analogue sources can be selected from the oval remote, with fully independent switching of all 15 to both outputs.

The record signal path has two sets of analogue inserts, again switchable from the remote. In Focusrite's ideal world the first would have a 515 equaliser and the second a 550 compressor, to be switched independently when needed, and indeed this is what Tape to Tape has. It has to be said in fairness that at present these are not actually hooked up to the 300; based on the 'if it ain't broke, don't fix it' philosophy, the record chain in the room is still handled by the original home-built set of big coloured push-button while the Focusrite handles the monitoring. Having said that, it is certainly envisaged that eventually the room will be using all the 300's facilities—there's no reason not to other than the time taken to rewire in what is evidently a very busy room. The result, in accordance with Focusrite's suggested layouts, would be an entirely Blue signal path, from analogue or digital source via Blue EQ and compression to analogue or digital destination. The system is...
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completed by the Blue 2/5 A-D converter, a 20-bit unit with dithering down to 16 bits if required. Tape to Tape's engineers profess themselves entirely satisfied with the quality of both D-As and A-Ds—this in an environment where most rooms use Prism Sound.

Metering of the signals at various points is also controlled from the remote, with the chosen signal appearing on the main rack box on the biggest pair of 150 vus you are ever likely to see. These can show the recording or monitor signals, or tap in before or after either of the inserts, and the whole sensitivity can be switched to cope with high levels. The only disadvantage of the eye-catching styling of the remote is that the control layout does not immediately yield its secrets. The divisions between sections are all at odd angles, and nothing is vertically aligned—knobs and switches are sometimes arranged in a curve and sometimes raked at an angle. There are patch-bay-style labelling strips below the selector buttons but it takes a moment or two to get your bearings. Once you find your way around you notice a couple of extra details: the monitor source can be set to automatically follow the record selection, and conversely the record feed can be linked to the monitor switch bank. This leaves open the apparent possibility that a device could end up fed back to itself, which could disrupt the signal passing through for recording—source selection, faders, trim and insert switching.

Andy Crump and Pete Norman, who share Tape to Tape's Focusrite room, are clearly very happy with what it does. When asked if he missed the variety of kit normally available for cutting, Crump's answer is an unqualified 'No'. There are no reservations about the signal integrity of anything in the chain, and no feeling of being limited by having just one compressor and one equaliser. Add the Mastering Controller and the integrated whole obviously works very well for them, and now that word is getting out that Tape to Tape has the room, there are indeed clients booking just for the Focusrite kit. In all respects therefore the move has been a success, and no doubt we can expect to see the Blue 300 appearing at the heart of more and more suites like this.

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Yamaha 01V

Yamaha has already defined price and performance expectations in digital mixing, but it is trying to do it all over again with a new addition to its family. Zenon Schoepé reports

It is important not to confuse the 01V with the Programmable Mixer 01, but by the same token it is important not to confuse it with the operational elegance of the 03D or 02R even though it shares their processing genes.

The feature list is predictably impressive given this desk’s lineage, but it is yet another example of still further trimming of the original offering in order to create a new price point. Convertors are 20-bit with 52-bit internal processing and 44-bit EQ processing. Out of the box you are presented with 12 mic-line inputs with switchable phantom power in blocks of six, plus two stereo returns. However, a single rear-panel slot can take 8-channel TDM, ADAT and AES-EBU-format 1-0 interface cards with the inputs arriving at channels 17–24 and the outputs assigned for function within the desk. There is also a 4-channel analogue 1-0 option card, and for the record none of these cards are compatible with the 03D and 02R.

As a point of interest only channels 1–16 have access to full channel processing, option inputs have only 2-band EQ, for example, but to counter this it is possible to swap channels from the option interface with fully featured channels 1–8.

In terms of group buses the 01V has four and these can emerge via four so-called ‘omni out’ sockets or via a fitted interface card. Omnit outs can also be assigned to deliver aux outputs, extra stereo outs or as direct outputs from the first 16 channels.

Throw in two internal stereo returns from the two internal multi-effects processors with full EQ and routing and the self-contained nature of this desk cannot be denied.

Internal effects take in 12 presets and 57 user-located while a total of 22 dynamics processors are arranged in 40 factory and 40 user-patches. Onboard automation is restricted to 100 complete desk snapshots, but if its dynamic automation you want then it is to the sequencer and MIDI you must go. There is no surround or multichannel mixing capability and frankly there shouldn’t be.

Channel faders—true 60mm motorised types with associated on, solo and mute switches, the last of which assign the chosen channel to the adjustment of its internal parameters.

In terms of non-fader controls you get dedicated buttons for utility, MIDI, setup, view, dynamics, EQ-attenuation, phase reverse, delay, and pan-routing—some of which will be familiar to 03D and 02R users. Repeated pressing of said buttons calls up successive screens which are adjusted via cursor movement and dial.

A cluster of fader mode buttons carry over the established practice of being able to set effects 1 and aux 1, 2, 3 and 4 levels from the moving faders. The Option I-Q key accesses the under layer of 8 inputs derived from the optional rear-panel interface board, if fitted, while the Remote key assigns the faders to internal and external MIDI data generation duties and can be used to create faders as group masters. All pretty established stuff, at least in Yamaha’s digital-desk land.

On the other side of the LCD a pot is dedicated to pan on the selected channel, while two more handle part of each band’s fully parametric EQ. Annoying things include a noticeable lag in the EQ curve display when administering tweaks—long enough on first encounter to make you wonder if there is something wrong. You also have to contend with the vagaries of doing EQ on two dedicated pots for frequency and boost, selecting bands on dedicated keys, admittedly, while the data dial is used for Q value. You get used to this eventually, but the dial has a peculiar resistive feel to it that is not helped by its size. It is much smaller than the dial found on the 02R, and this means you are turning it from closer to its axis with the usual mechanical laws applying. The dial performance is a pity as you have to use this control an awful lot.

I would have to wonder what I would have made of this box had I not already become well acquainted with the operation of the 02R and 03D before it. I am inclined to think I may have regarded it as something of a culture shock rather than regarding it first and foremost as a derivative of existing consoles, and therefore being a little too eager to explain things away.

However, Yamaha has the benefit of other mod-
els in the series to its credit, and as such the 01V avoids many of the pitfalls that might have enticed it. If this was anyone else's first stab at a desk with this sort of power, for this sort of money, then I would suspect that it would have made its compromises in far less acceptable areas than Yamaha has here.

Once you are into its mind-set then operation becomes logical and predictable even though the business of actually administering an idea can be painfully protracted. You will know how to do everything on this desk fairly quickly, it is just that you are not exactly encouraged. For example, with the 01V I have realised why you would want to have EQ and dynamics memories because with them you shoot for the nearest approximation of what you want and after as little as is possible rather than having to go through the convoluted business of starting from scratch each time.

IF ANYTHING you have to use an 01V to truly appreciate the 03D much in the same way that returning after a brief, yet intense, flirtation with an 03D to the welcoming and forgiving arms of the 02R makes you want to stay home a lot more often.

Indeed the biggest message the 01V delivers to me is the reminder of what a truly fantastic product the 02R still is so long after its launch.

Comparisons to the 03D and 02R are ultimately unfair as they occupy significantly higher price points. As such the 01V targets a different type of user, one that does not want to work to picture, nor mix in multichannel on a compact but intelligent work surface, rather it is someone for whom the fact that the desk is digital and can interface to digital multitracks cheaply and directly is the overriding concern. It has a killer little submixer, perhaps for keyboard rigs or instances where things are present, and the clever stuff is achieved predominately on the faders instead of within the channel processing. It has a good little tool box, too, with coaxial digital I-Os, phono 2-track I-O, headphones, main stereo and monitor outputs, and the ability to cascade two desks digitally.

For the money it is quite phenomenal value and that is something that has to be restated repeatedly, and is pro enough in the right hands to perform a useful role in the right job. It is not a toy.

By the same token I would not want to leave anyone thinking that the 01V is in some way the 03D (or the 03D) in its terms of operational ease and functionality. The 01V is a different proposition altogether, and, in my opinion, makes the original Programmable Mixer 01 obsolete despite the fact that the new model is more expensive.

They'll sell loads.
The inclusion of a hopper for blanks once differentiated CD copiers according to price. Tim Frost continues our look at desktop roll-your-own CD writers.

Time used to be that CD-R copiers which included multi-disc feeder systems were quite expensive. In the last couple of years, though, the huge demand for desktop CD-R copiers has brought less expensive units to the marketplace. TraxData's TraxCopier is a case in point. Costing just under £6,000, this unit has a hopper system that can feed up to 150 blanks into the machine for unattended copying using its two CD-R drives.

Internally, the TraxCopier has two hard-discs. The hard-disc can archive up to three CD images, any of which can be recalled from the menu and used as the master. You do not have to tell the TraxCopier which format the CD is; as the TraxCopier will check for itself and make sure the copies follow the same standard. Most CD formats are covered; although we were warned that CD+G for karaoke-style CDs is not supported.

The master is then ejected into the Accept hopper and the turntable feeds both drives with blanks, so that copies can be burnt two at a time at 1x speed, taking around 20 minutes to copy two full discs. These are ejected into the Accept hopper when they are done and more blanks loaded; and so on, until either the number of copies programmed have been completed or the machine runs out of blanks.

Using its Batch Mode, the TraxCopier can also automatically copy from three different master CDs. This is done by loading the input hopper with the first master followed by some blanks, then the second master and some blanks, followed by the third master and blanks. In the copy process, the master is detected and copied to all the blanks until the next master is detected. The copier then automatically loads an image from this new master and copies it onto the next batch of blanks until it sees the next master, and so on.

Faulty discs are sent to the Reject hopper, but this function has to be treated with a certain amount of caution; as CD writers do not have read-after-write ability and discs with faulty sectors in the middle of them will not be detected. Blanks will only appear in the Reject hopper if there was a problem with the initial interrogation of the blank disc, prior to writing to it.

If you need to be totally certain that all the copies are good, then there is a separate Compare function with takes the batch of copies and compares them one by one with the master image on the hard-disc. The limiting factor is the number of masters that can be copied in the batch process is set by the capacity of the internal hard-d as the master images are not erased ing the batch process.

The setup menus give various options, including specifying read write speeds. Write speeds can be reduced to 1x for those who are not that it makes a difference when writing audio, but, of course, the copy process then takes four times longer.

The system copied well with all drives except for a non-conforming 'R'-pressed audio CD that caused the temperature to lock up; although the test CD discs seemed to copy OK. Issues handling new or modified disc formats can be solved in a later firmware revision. The TraxCopier can easily up firmware revisions directly from CD and does not have to go through the distributor for updates.

For 'normal' audio, data, video CDs formatted with a mix of file types on them, the system copied everything smoothly—and quietly. Even so, there were still a few reservations about TraxCopier specifically in terms of a disc music business setting. There, for example, no audio input or output functions, so all mastering and churning would have to be done on separate CD readers and writers.

It would also have been nice to have a way of feeding the master separate from the blank hopper. The TraxGo could then be permanently loaded with blanks and copies of new masters off whenever they are needed—or having to always be putting a master at the bottom of a new pile of blanks. Lastly there is the general comparison between this type of hopper copier and a tower system with eight drives.

With two drives, TraxCopier will take hours to copy three discs, whereas a tower system will do it in half the time. Advantage of the TraxCopier is that copy process can run completely unattended, where obviously some needs to remember to feed the tower with blanks every twenty minutes so you pays your money and takes your choice.
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JBL LSR32

Studio Sound's 'bench test' loudspeaker reviews continue with the LSR32. Keith Holland reports

The JBL LSR32 is a 3-way, passive loudspeaker comprising a 12-inch (300mm) low-frequency drive-unit with a carbon-fibre composite cone, a 5-inch (125mm) mid-range with a Kevlar composite cone and a 1-inch (25mm) high-frequency drive-unit with a damped titanium-composite dome that radiates through an elliptical, oblate, spherical horn. The LSR32 is the largest loudspeaker we have tested so far in this series, weighing in at 21.3kg and having external dimensions of 645mm x 490mm x 292mm deep; the internal volume is specified as being 50 litres. The mid and high-frequency drive-units, along with the bass reflex port, are mounted on a square sub-woofer allowing rotation through 90° for either horizontal (landscape) or vertical (portrait) orientation: this review was conducted with the loudspeaker in the landscape orientation as supplied. The crossover network is specified as having 4th-order Linkwitz-Riley alignment with crossover frequencies of 250Hz and 2.2kHz. The manufacturer specifies a long-term maximum (IEC 265-5) power handling of 280W rms, that when coupled with the quoted sensitivity of 94dB SPL@1m for 2.83V input (which is confirmed in Fig.1), endows these loudspeakers with a (theoretical) maximum continuous output level of 116dB SPL, each at 1m under free-field conditions.

Fig.1 shows the on-axis frequency response of the loudspeaker. The response is held within ±2.5dB from 70Hz to 20kHz with ±6dB at around 4kHz. This is a commendable result. The low frequency roll-off is seen to be fairly smooth and gradual, having an approximate 6th-order slope. Fig.1 also shows the harmonic distortion performance. This result is remarkably good. At worst, the second harmonic distortion rises to ±5dB (0.5%) at 60kHz, but has disappeared to ±0dB (0.1%) by 200kHz, with 3rd harmonic distortion peaking at ±6dB (1%) at 3kHz and remaining below ±5dB (0.2%) at higher frequencies. This model displays the lowest harmonic distortion of any yet tested (at least at the test level of 90dB SPL at 1m).

The horizontal off-axis responses (Fig.5) are fairly well controlled with some evidence of lobing (crossing-over of the plots) between 1kHz and 3kHz. Very little mid-frequency narrowing is evident though, due to the 3-way design. The vertical off-axis performance is dominated by a dip at the crossover frequency between the mid- and high-frequency drivers; this result is expected with spaced drivers. The waterfall plot (Fig.7) shows that the low frequencies decay fairly rapidly, due to the gentle low frequency roll-off, and that ringing at higher frequencies is confined to very minor problems at 500Hz, 800Hz and 1kHz. The acoustic centre result (Fig.2) confirms the effect of the gentle roll-off; it shows a maximum group delay equivalent to a shift of the low-frequency source to only 2m behind the high-frequency source. The group delay in the lower-mid-frequency range (200Hz to 600Hz) is, however, higher than that found in equivalent 2-way designs; a result that is confirmed by the step response chart (Fig.3), which shows that the transient response occurs later in this frequency range than at higher frequencies. As expected from the flat on-axis frequency response, the power cepstrum (Fig.4) shows very little evidence, if any, of reflections, indicating that the low frequencies are well controlled. The lower intermodulation distortion is excellent, and the on-axis frequency response is very flat. A mid-frequency group delay, possibly due to the high-order crossovers, spoils the step response somewhat, but otherwise there is little evidence of any problems with the design. The loudspeaker is also capable of high-output levels, but its large physical size must be borne in mind when comparing it with other loudspeakers in this series.

October 1998 Studio Sound

The AT4060 combines premium 40 Series engineering and vintage tube technology to deliver the exacting, versatile performance required in the most demanding studio applications. With a dynamic range that far exceeds that of other tube mics, the AT4060 provides the coveted sound of value design without compromising the specification standards necessary to excel in today's diverse recording situations.
In Studio Sound's 'bench test' loudspeaker review Keith Holland retests the Meyer HM-1S

A test of the Meyer HM-1S previously published in the June 1998 edition of Studio Sound, concluded that a severe dip in frequency response at 80Hz was probably due to a phase reversal in the crossover between the satellite loudspeakers and the subwoofer. Meyer has since admitted that the test loudspeaker had been demonstrated, prior to being sent for review, in a situation that required the subwoofer to be placed away from the satellite loudspeakers. In this setup, the demonstrator found an improvement in response with the phase of the subwoofer reversed, and re-wired the connecting crossover package is built-in to the cabinet, but the 100W power supply is separate. The ported cabinet is relatively small, measuring 8.0 inches (200mm) wide by 9.7 inches (246mm) deep by 11.5 inches (292mm) high, and weighs 5kg. The drive-units are both magnetically shielded. The review system was fitted with an optional subwoofer that comprised a 10-inch (250mm) driver (not shielded).

The electronic crossover is specified as having complex roll-off shapes with a crossover frequency of 3kHz and an improvement in the internal power amplifiers are rated at 200W each into 4Ω giving a claimed maximum sound pressure level of 115dB at 1m (120W with subwoofer) with a pink noise or music input.

The loudspeaker is constructed to a high standard and finished in a real wood veneer (natural oak optional), and is small and light enough to mount on most monitor stands.

Fig. 1 shows the on-axis frequency response for the Meyer HM-1S with and without the optional subwoofer. The response lies between ±5dB limits from 10Hz to 20kHz with the subwoofer, and 80Hz to 20kHz without, except for a narrow dip of some 5dB at 7kHz. The low-frequency roll-off is very rapid, suggesting the use of a high-pass protection filter. The low-frequency harmonic distortion performance is unspectacular with a peak in second harmonic to +0.03dB at 50Hz and in third to -22dB at 1kHz.

The levels of the harmonics fall rapidly above these frequencies to below -50dB at 80Hz upwards, except for a peak in third harmonic to +53dB (1.7%) at 700Hz.

As before, the off-axis response measurements (Fig. 5) show very wide dispersion at high frequencies evident from the similarities between the 15°, 30°, and 60° measurements. There are no significant peaks or dips in any of the off-axis response curves indicating a freedom from side-lobes in the polar pattern; although a narrowing of the directivity of the woofer can be seen at high frequency end of its range between 1kHz and 3kHz. As the loudspeaker is the dual-concentric design, the directivity is the same in the vertical plane as it is in the horizontal plane.

Fig. 5 is the step response of the loudspeaker without the subwoofer. Again the HM-1S exhibits excellent time-alignment between the drivers with a sharp rise to the step and steady decay, although, as confirmed by the acoustic test results on the Studio Sound website, there is a delay of some 1ms in the low frequency energy, corresponding to a shift in acoustic centre of more than 3° behind the loudspeaker at low frequencies. This result is an improvement, however, over that for the previous test with the phase-reversed subwoofer, being more similar to that without the subwoofer. The power spectrum (Fig. 1) shows some evidence of smeared reflections at 200Hz and 700Hz, but is otherwise fine and a slight improvement over that for the previously tested example. The waterfall plot (Fig. 6), which has been introduced since the last test of the HM-1S, shows no serious ringing above 15kHz with an even decay rate at all mid- and high frequencies, but the group delay at low frequencies is very pronounced.

As concluded in the last test, the Meyer HM-1S is a mixture of strengths and weaknesses. Amongst its strengths are excellent time alignment and high frequency directivity control, but this performance is let down by a ragged on-axis frequency response.
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The news coincides with a tie-up with Sonorus that uses Mytek's 8-channel 24-bit, 96kHz 8X96 series converters and Sonorus' Studio card in a new Mytek - Sonorus DAW 9624. This multichannel hardware package is capable of running various third-party 24-bit 96kHz software packages for Mac and PC. Depending on the software chosen, DAW 9624 can be configured as a general purpose multichannel DAW, or CD/DVD mastering, Foley, or film sound workshops. Current 24x96-capable software choices include Cubase VST 2.4, SMultrude 4.5, Safari Pro, Wavelab, Cakewalk 8, and such plug-ins as Waves NLP. DAW 9624 will run all 48kHz and 44.1kHz software as well. Mytek US Tel: +1 212 234 9191. Sonorus US Tel: +1 212 232 7700.

SPL-Brauner 3D recording

In a joint venture with mic manufacturer Brauner, SPL has launched the ATMOS 5.1 Surround recording mixer, which works in conjunction with Brauner's AS45 adjustable Surround microphone. The ATMOS 5.1 Surround recording mixer features five high-precision microphone preamps providing up to 70dB of gain.

The circuit incorporates Lundahl input transformers as well as pads, phase reverse, phantom power, low-cut filters and switchable gain controls. All switching uses high-quality switches, and relays with gold-plated contacts used throughout. Comprising a main unit, and a separate power supply, the mixer section occupies 5U of rack space with the PSU being just 1U. Mixing and 5.1 panning are possible from each.

San Francisco AES

Despite the developing trend for spring show releases, San Francisco did have new product to boast about, some of which will be of wide appeal.

Yamaha announced the sub-US$3000 D24 24-bit, 96kHz digital recorder based around removable 3.5-inch magneto-optical disks. The machine offers 16-bit, 20-bit, 24-bit, 8-track simultaneous recording and play capability at 44.1kHz and 48kHz, and 4-track record-play at 96kHz. Up to eight units can be synchronised together to create a larger system and delivery is expected to begin in the first quarter of next year.

The D24 is intended to serve as a multi-track recorder for music recording and mixdown and live audio applications for television, film, and live playback for theatre and other live sound applications.

Editing options include copy, move, erase, delete and merge, and tracks can be merged. Variable speed is ±6%. Each track also includes 8 virtual tracks for a total of 64 while 8-track recording time is 15 minutes at 16-bit, 44.1kHz and 9 minutes at 24-bit, 48kHz on a 640Mb disk. Four-track recording time is 9 minutes at 24-bit, 96kHz and 512 songs can be recorded on each disk. The machine can be connected to external hard drives via a SCSI-II connector.

The D24 syncs to MTC and SMPTE video sync, and word clock, and is controlled via MMC, 9-pin or an optional remote.

Thoughtful features on a box of this nature are time compression-expansion (50%–200%) two tracks at a time and pitch can also be altered two tracks at a time without changing the tempo or duration of audio material.

The machine offers 01V-sized interface cards in ADAT, TDM, AES-EBU and analogue formats.

Tascam bridged the gap between its TM-D1000 and TM-D8000 digital desks with the TM-D3000 model, the first to sport motorised faders and 100mm ones at that. Price at US$499 it is expected to hit the street in the first quarter of next year.

San Francisco AES, October 1998
Sennheiser Evolutions

Launched at Frankfurt as a complete product line,

Dave Foister asks if the dynamic has evolved

O

The FEW classic dynamics, a good number of them have Sennheiser written on them. A whole new range from the company is therefore bound to be of interest, and the Evolutions have been arousing much curiosity.

The Evolution family comprises a mixture of new and updated models; harmonised in terms of finish to the same choral-satin effect familiar from the current radios and others. All feel very light — you could be forgiven for thinking they were plastic; although they are in fact metal — but experience suggests they will survive better than some heavier constructions encountered.

The vocal microphones in particular are not structural lightweight, having die-cast zinc housings with shock absorption built in. A feature shared by all is the use of Neodymium magnets, and this gives every model the capacity to handle levels in excess of 150dB SPL. A surprising presence is the E904, an upgrade on the MD 904 recently introduced to such great acclaim (well I liked it). I didn't get sent one of those, but five of the rest of the set turned up for evaluation, covering the whole range from general-purpose microphones to vocal and bass specialists.

The most recognisable model is the E909, clearly descended from the highly popular MD 409 and aimed at the same applications. The distinctive flat side-on body is retained, with the stem and its integral XLR now colour-matched to the main body, and the capsule has been updated incorporating a hum- and click-bucking coil. The original was much in demand for jobs where a natural frequency response and the ability to get in close were needed, and the E909 continues to excel in the same way. It is by far the smoothest of the lot, with a trace of brightness that is likely to be helpful rather than harsh, and plenty at the bottom. Its supercardioid pickup is reasonably consistent with frequency, and it looks set to be as good an all-rounder as its predecessor.

There is one surprise in the bunch and that is the E902. You might guess from looking at its flat, square body that it’s meant for less drum: really just a smaller version of the ES 85; but its behaviour is even more extreme than might be expected. There’s no need to look at the frequency response plot to tell that there’s a huge amount of upturn at the top as well as at the bottom, with a big hole in the middle. The graph shows it to be no less than 10dB up at 700Hz and down at the top end, with the bottom extending all the way down to 20Hz (and you can feel it). This produces the same kind of effect as a high-toudbound control, making the right source sound bigger and fatter with no help from the console. It has an enormous bottom end with surprising clarity, and the exaggerated top gives it bags of punch and bite. For general use it comes across as very thin as there is so little body in the mid range, but in its intended role it has a powerful character all its own.

Also specialised but not to quite such a jaw-dropping extent are three hand-held vocal models. The entry-level end of the range is represented by the ES 85, another one I did not get to try, and next up is the cardioid ES 95. A lot of people are touting this as a serious competition for the SM 87, and its resemblance in terms of build and sound is clear. It has a typical vocal character, with a very flat response from the middle down, and a modest upturn at the top end, with the 4kHz. This translates into a good present sound, but not exaggerated enough to make it hard, and the character is retained off-axis with a commendably consistent polar pattern. It is also available with a lockable noise-switch as the ES 94.

Next up is the ES 99, physically there’s little to tell it apart from the ES 95, but it is altogether more even, with a more extended top end and a generally fuller sound. Its pattern is supercardioid, seemingly as smooth off-axis as the others. Top of the vocal range is the ES 955, only available unswitched. This, too, is supercardioid, and is the smoothest of the lot. It clearly is not truly flat, neither is it intended to be; it has in fact got a more pronounced HF lift than the ES 95 on paper, but as the frequency response does better at the extremes this comes across as a clean bright presence without detracting from the overall openness of the microphone.

There is one more, the E908, claimed to be the world’s smallest dynamic microphone and mounted on a miniature gooseneck. This was not available to me, but if it shares the central characteristics of the rest of the range it should be quite a contender.

This is quite a family, with the courage to include extensive deliberate tailoring as well as the neutral models. All impress, and some can expect to join the 121, 112 and 109 as members of the elite.

NEW TECHNOLOGIES

< channel via a front surround-channel, an LCR pan control and a divergence control. In the Subwoofer matrix, the sub signal can be selected from the front, surround, and centre signals, while in the Surround matrix a stereo mix can be fed to the front, central and surround channels. Furthermore, the stereo soundstage from the front and surround channels can be widened using all-pass stereo spreading. Two phase meters are used to display LR and SR SL correlation, and the monitor section allows each bus to be independently monitored. Beamer's AS5 microphone system is based around five matched microphone heads, three of these are arranged as a Deca tree to handle the LCR recording.

A further two heads handle the surround information. In addition to supporting the INA 5 standard, the AS5 allows continuous adjustment of the polar characteristics of each microphone from omni-directional up to figure-of-eight. These adjustments are made directly from the Atmos 5.1 Surround recording mixer. Furthermore the positions of the SL-8R microphone heads are variable up to 90° (+45°). The AS5 includes 25m of 12-pair screened multicore cable to connect it to the SPL Atmos 5.1 Surround recording mixer.

SPL Germany Tel: +49 2163 98140.

Dual CD-R

With a UK price of less than £400 line VAT, the Philips CD765 dual-deck audio CD recorder has a fully specified CD-R, CD-RW, CD-ROM player transport, and a CD, CD-R, CD-RW player transport in one unit allowing mastering and double-speed duplication.

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A Harman International Company
Brauner Valvet

A new mic from this young but already respected manufacturer is to be welcomed. Dave Foister opens the box.

Dirk Brauner’s microphones have made a splash with the single VM1 model winning awards, and being acclaimed by no less a person than Bruce Swedien as „phenomenal“—it sounds like the best U47 that you’ve ever heard in your life.

They are big and bold, with a precision build and the best kind of valve character, but the small size and low price make them an obvious contender for a lot of people. The design is a clever one that replaces the VM1’s continuously variable polar pattern with a choice of cardioid and omni. It can also be upgraded to a full-blown VM1 if required, and then there is the Valvet, offering the same rather unusual pairing of polar patterns in a very different package.

The enormous bomb-proof flight case gives a slightly misleading impression of the size of the microphone, though the cradles the microphone in its suspension mount and the power supply, in its lid is the Tuchel-terminated multicore cable, a big hit, but substantial and generous in length.

At one end of this is the power supply, a small, sparsely equipped, but clearly robust box. In line with the standard Brauner image is the fact that silver metal throughout, and its front panel contains only three switches, for power on, polarity reverse, and polar pattern selection. The first is accompanied by a blue indicator LED (apparently Dirk hates red LEDs) and the last is a simple toggle between cardioid and omni. Clearly the circuitry takes some time to settle when this is changed as the manual says not to use the microphone for 30 seconds after switching it over. Evidently this is the month for dodgy German-English manuals; the few sheets explaining how the Valvet works are riddled with such remarks as: “If there is a sparkling buzzing noise at the start, do not care about it, which whilst quaint does not enhance credibility.

Fortunately, this does not really matter a toss as the microphone is so straightforward. The solid cylindrical body has no controls whatsoever, and looks a little like a scale model of a proper valve microphone. Its mounting assembly is remarkable in that it does not quite encircle the microphone body; it is a C-shaped structure that grips the microphone essentially by the interference fit of its rubber suspension elements. Clearly the intention is that the two components should not routinely be separated, an idea of which I heartily approve. The mount’s swivel is locked by means of the same level-like handle as on the much heavier VM1, making it securely effective against droop. As with the bigger microphone, the manual carries a warning against over-tightening this handle as it could break under its own leverage. There is certainly no doubting its ability to support the Valvet, even though it is heavier than it looks.

The 1-inch dual-diaphragm capsule is clearly visible in the basket, and is likely to remain so as there is no frosted shield. The size of the body means that there must be plenty of socks that will fit it, but none is supplied by Brauner. It may simply be that there’s enough internal wind protection—with the same performance. I used it when I was not aware of any problems.

The only remaining adjustment is an earthing switch, an eminently sensible provision on a valve microphone. Like that on the VM1 this has three positions, allowing the audio ground to be shorted to mains earth, completely separated, or linked via a small capacitor. One of these options will surely cure any hum-loop problems.

If the size of the Valvet is scaled down from the VM1, the sound is not a proportional reduction. It is open and clear, and characterised by an unusual smoothness in the upper registers—less clinical than some solid-state microphones, but without the obvious presence lift exhibited by some valve designs. It has a pretty good bottom end too, and an off-axis response that sounds almost as flat as the front pickup, in cardioid as well as in omni. I especially enjoyed the VMIs as spaced omnis, and the Valvet appears to share the essential character. It is also extremely quiet, and only hums when the earth switch is in the wrong place. Its lack of a pad is the more surprising in view of its sensitivity, which is quite high, but even with the most powerful soprano I’ve recorded in a band that was two feet away from it it never wavered.

The Valvet can only build on Brauner’s existing reputation. Its laboratory appearance attracted comment from everybody who encountered it, and its performance matched up to expectations. It deserves to do well.

Brauner, Germany.
Tel: +49 2856 9227
Fax: +49 2856 92719

A&H digital

Allen & Heath has launched five new consoles including the digital ICON series DL1000, DP1000, and MixWizard series WZ12.2DX, WZ16.2DX and WZ14.4.2.

The ICON series is a range of compact digital mixers for live-sound applications. The first two models in the series are the 10-input, 4-output DL1000, and a powered version, the DP1000, that comes complete with 600W into 4Ω stereo power amplifier. Both consoles will begin shipping in October, with prices starting at £1,000 (UK). DL1000 and DP1000 offer 6-mic line inputs with 4 sweep EQs with fully parametric mids, plus 2 stereo inputs that can double as mono mic inputs. In addition to the LR output, ICON consoles provide configurable AB amp outputs, plus monitor, aux and LR recording outs.

Target markets are performing artists and live-sound venues. These can program and recall settings in song patches, which can then be sequenced according to a playlist and triggered by footswitch, pushbutton or MIDI control. This preprogrammed sequence can be overridden when requiring special patch sets and levels, and effects for between song announcements.

Key settings such as gain, levels and mutes are dedicated controls for instant access, including 100mm faders for all inputs and main outputs. Other console settings are created and adjusted via a strip of rotary controls used in conjunction with a large backlit LCD screen. The consoles’ onboard dual 10-band graphic and amplifier configurations are stored separately in venue memories. ICONS include two built-in effects processors plus noise gates and compressors.

Symetrix processors

Symetrix has released the 300 Series range of half rack-width boxes for the contractor and broadcast industries. These include the 301 mono compressor-limiter, the 302 dual mic preamp update on the popular 310.
Building on the phenomenal success of the DP200, the DP226 continues the reputation for sound quality in a product aimed squarely at the most demanding applications of the sound reinforcement, installation and studio market.

The DP226 features 2 inputs and 6 outputs. Both inputs have an 8 band parametric equaliser, base delay and gain control. All outputs feature crossover filters, 5 band parametric equaliser, high and low shelving filters, limiter and delay. Full metering is provided for inputs and outputs, with mute and access buttons allowing quick set up and gain adjustment. The DP226 can also be controlled via PC with our popular AudioCore Windows™ control software.

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Höf Höfex Spectral Exciter

The best efforts in the realm of excitation tell a lot about a company, as Dave Foister discovers

Enhancers come and exciters go, with varying degrees of success and credibility— or are we hand in hand? Some specialise in them, some throw one into the catalogue for the sake of completeness, some do not wish to be associated with such things—and some are so out of the mainstream that you know sooner or later they are going to have one, and that it is going to be a little different.

Höf Audio belongs in the last category, by dint of its Dynamic Master, a comprehensive stereo dynamics processor with an approach all its own. This is now joined by the Höfex Spectral Exciter, a 2-channel enhancer package that adds a couple of new twists to the familiar parameters.

Operationally it is simple. The 1U box has XLR in and out on the back, controls for the exciter functions on the front, and that is it—no level controls, no metering for instance, showing remarkable confidence in the user's ability to drive it properly. There is an overload LED on each channel flashing at +18dBu, and this is placed after the processor to show its effect on the output level. It is worth noting that the Höfex is entirely analogue, so the lack of pinpoint level setting is, perhaps, less surprising.

The exciter itself comprises three separate circuits, with a fourth process available on the end having little to do with the central job. The most familiar element is the HF exciter, working with phase shifts and added harmonics to increase the perceived HF content without EQ. The actual processes involved in all this remain something of a mystery, partly because Höf does not want to give too much of its game away, and partly because the manual is written in such German—English that it is often hard to fathom what it is driving at. I am not much of a flag-waver for English as the global language, and I am certainly not saying I could do a better job of writing in German, but if a company is going to write an English manual it really ought to get it checked out by a native English speaker—this one is a classic example of how wrong they can get it.

Like many of these processors, the effect of the HF exciter is tailored by a Spectrum control, determining the frequency above which it will operate. The range is considerable, allowing choices from near infinite to be produced. The manual describes the results as "soot as silk", and it is genuinely characterised by an unusual smoothness. It is possible to make the sound much higher than would ever be needed, but it never seems to acquire the hardness that can often accompany this kind of treatment. Setting it up is helped by a global Solo function that removes the original signal from the output, leaving all the processing.

The high frequencies are further affected by a single control called Glitter, which turns out to be a very appropriate name. This too adds overtones to the original, working at the upper end of the range, and although it is similar to some settings of the exciter itself, there is enough of a difference to warrant its inclusion as a subtle adjunct to the main process.

Frequency response manipulation is completed by an LF exciter, again with a single control to adjust the amount of effect. This seems to attempt to add bass that is not there, working very low down indeed to give a big thumping bottom end that manages not to turn into a muddy mess. This is the only effect that significantly alters the levels, upping the output by as much as 8dB when cranked right up on a signal that already has plenty of bass.

The final process is not something you would normally find on an exciter, or anywhere else for that matter. Called REX (for Room Exciter), it tries to simulate the response of a room around the sound using reflections and phase manipulation. In fact the output of the circuit is completely out of phase across the two channels and extremely unpleasant to listen to on its own. The intention is that the Höfex should be put in solo mode, fed from an aux or subgroup, and the REX signal mixed back in as required—it has no adjustable parameters on the unit itself. The contributions of the main exciter circuits can then be adjusted in proportion. Used carefully this can give some interesting results, with a definite opening up of the stereo image and added liveliness.

The approach of the Höfex is decidedly unusual, and with a manual that in one of its more lucid moments suggests that the treble of the source might be deliberately reduced so that the exciter can put it back again, it is more likely to detract than attract. That would be unfortunate, as the Höfex is genuinely, smoothly different from the run of the mill. I have tried a few, and been less than impressed with several, but the Höfex would get plugged up more often than most.

New Technologies

< SX202, and the 303 stereo level matching bidirectional interface amp. Other units are the 304 2-in, 4-out headphones amp, 305 1-in, 4-out distribution amp with individual trim pots, and the 307 dual isolation transformer.

Traditional 1U rackmounts from the company now include the 581E distribution amp and the 565E dual compressor-limiter-expander. The former is a 4-in, 16-out device that accepts stereo audio feeds and other configurations via simple rewiring. The 565E offers two channels of simultaneous in-line processing with proprietary Dynamics Squared circuitry which claims to reduce distortion when automatic gain reduction is at a maximum.

Symetrix, US: Tel:+1 425 787 1222.

Soundcraft Series 15 desk

Designed for on-air use, Soundcraft’s Series 15 console comes from the same family as the Series 10 radio desk, but adds facilities for general production. Available in 16, 24 or 32 frame sizes, the desk is modular and has as standard 4 stereo groups, insert points and comprehensive monitoring.

Mono and stereo inputs are provided in broadcast and production versions and a choice of telco input modules is available form an option list that numbers 35 different module types.

Soundcraft, UK: Tel:+44 1707 665000.

Barco monitor convertor

Barco’s Uno processor reproduces digital video signals on a standard computer display (52kHz) or on an analogue RGB monitor (15kHz). In contrast to simple SDI-VGA convertors the device maintains broadcast picture resolution, provides full colour matching capabilities and adds broadcast functionality to a standard computer display.

Barco, Denmark: Tel:+45 39 170000.

Cinemix adds film panel

D8R has added a film-style master section option to its Cinemix console which can handle mono, stereo, LCFS and 5.1 channel formats. The new section offers discrete assignment of input signals to all individual buses and joysticks, but formats are selectable between the aforementioned...
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Power Technology DSP FX

Rob James tests the outboard plug-in that turns a personal computer into a dedicated effects processor.

After DSP FX plug-ins were recently reviewed, another package arrived from America containing one of the Power Technology processor cards, together with the latest version of software. The software has been revised to be more efficient. Indeed, Power Technology claims that on a Pentium II 300 a reverberation now only uses 7% of the processing power and one user has already had a total of seven reverbs running during a 24-track mixing session. If you need a bunch of reverbs running simultaneously the economies of this approach start to really make sense.

This version adds hardware in the form of a half-length ISA card with a daughter-board piggy-backed on top (which I assume carries the optional AES-EBU digital interface circuitry). There are a couple of connectors on the card for Fs Link In and Fs Link Out. A TRIP DIP switch selects any of 8-bit addresses for the card. If you have enough slots up to 8 cards may be installed in one PC. As I had previously suspected from the manual, installation is made much easier than usual by a (very) small piece of software to avoid the use of interrupt requests (IRQs) and direct memory access (DMA) channels. The only configuration issue is memory address—on my machine the default box 240 address was already in use by another sound card, but thanks to intelligent board layout, the DP switches are accessible without removing the card and it took but a moment to pick an unused memory address and change the card setting. Once installed, the external connections are four 7-inch jacks for 100kHz analogue in and out, and a 9-pin D-connector that either connects to an optional external +12V solid state 20-bit converter, or, if the AES-EBU option is fitted to a pair of XLRs for breakout.

The software installs without fuss using the dongle supplied with the WIZARD. The only item to watch out for is not to install the dongle driver when invited to do so. With the optional AES-EBU interface audio quality is subjectively high, much better than the onboard converters.

The algorithms employed are identical in the hardware and plug-in versions. However, unless you are on a very high budget I think the hardware solution has a lot to commend it. The main reason there is no longer any need to concern yourself with how much processing power the effect is using since all the work is done by the onboard DSP. The PC is simply providing the interface and housekeeping of the presets. The quality of the effects is also high, and they all seem to have been designed to interact with the software. The hardware interface is a lot easier than the ISA one.

As with most effects units, hardware or software, it pays to experiment. Some very subtle effects can be achieved with patient tweaking, and of course, these hard won presets can be named and saved for future reference. Microsoft-WAV files may be processed in real time by the DSP FX hardware and the Virtual Pack plug-ins. The other advantage is there is no need to concern yourself with how much processing power the effect is using since all the work is done by the onboard DSP. The PC is simply providing the interface and housekeeping of the presets. The Virtual Pack can be used for this purpose stand-alone—that is, without using host editing software. The inputs are achieved by simply not saving the output file. In fact, most effects units that delay effects alone. The various Leslie speaker effects are also impressive. In fact, if the choice of effects on offer have a weakness it lies in the lack of novelty. The effects supplied are apt to maintain what might be termed the standard effects rack kit. If perhaps less superficially exciting than some of those offered by other manufacturers. However, the real novelty here is the use of a PC to host a suite of effects. Even better, the PC does not need to be the latest Pentium II flying machine. Any reasonable PC with virtual ISA slots and capable of running Windows 95 at a sensible speed will do a perfectly good job.

If you can find a mother-board with enough slots to take the maximum 8 cards you will have a very serious effects processor.

If the company keeps the price reasonable, and continues to produce high-quality effects to add to the existing armoury, this is a very attractive way of acquiring a rack full of effects. Plug-ins are more suited to your way of working Power Technology has been running a special offer on the Virtual Pack that is almost too good to resist.

A free demo may be downloaded by pointing your web browser at www.dspfx.com.

NEW TECHNOLOGIES

<16-18mbit formats, access to group output buses for surround pan pots and input modules, access to group output buses for both joysticks, and an extensive selective muting system of individual channels.

The film master allows engineers to work on a maximum of four premixes-stems of six buses each simultaneously, and a stem-return module is available as an option. The monitoring matrix module is 24 × 2 × 6 allowing stem-premix returns to be fed into the console's control-room monitoring section. Using this module no inputs (traditionally the upper faders) need to be sacrificed for monitoring purposes.

The monitoring matrix module caters for four groups of six inputs each, and all four groups are fitted with bypassable level adjustment trim pots. All inputs are provided with Direct-Playback, mute and solo switches.

Described as the Airmix's little brother, the company's Airmate radio desk is designed for the self-op DJ.

The 19-inch frame is built up of eight triple input channels with one mic and two stereo-line inputs per channel. Each channel is additionally equipped with 3-band EQ and phantom power. A master section provides a clean feed output for interfacing with telephone hybrids with talkback.

D&R, Netherlands. Tel: +31 294 418014.

Stramp boxes

Peter Struven has launched a number of new Stramps including the STRAMP SR-X (a countdown box for film and video post) and the STRAMP CP-3 (a SSL parallel to Sony serial converter with track select for SSL 4k, 5k and 6k series).

The STRAMP DAB01 is a Sony 9-pin RS422 serial to Tascam format serial protocol converter that is suitable to use with the STRAMP ADX SuperSet, The STRAMP CP-3, and the STRAMP AV, MFCX. Users can connect up to six H/8 machines to >
When we changed to SM 911, 7 or 8 years ago, we got that sound back. It has a really good musical edge.

When **BASF SM 900 maxima** came out we started to use that on the 24 track – it gives me that sound I want.”

Producer of the Brit Awards “Album of the Year” 1997
“Everything Must Go” by the Manic Street Preachers and Winner of the Music Week “Producer of the Year” 1997, Mike Hedges has produced hits of artists such as Texas, Everything But The Girl, Siouxsie and The Banshees, The Cure, The Beautiful South, Geneva and McAlmont and Butler.

For more information contact +44 01295-227838 or visit EMTEC Magnetics’ web site at http://www.emtec-magnetics.com
Antares ATR-1 Auto-Tune

Tuning can often mar the perfect take when under detailed examination. Dave Foister tunes in, drops out and cheats.

This should have been another one for the April issue. It's a standing joke, isn't it, along with the Better Knob and the Funky Control—have you got anything that can make me sing in tune? It's the recording equivalent of the philosopher's stone—a box that can correct vocal intonation in real time, surely too ridiculous to contemplate. Yet that's the aim of the ATR-1 Auto-Tune from Antares, a new hardware box derived from an already successful software plug-in.

The ATR-1 is a single channel processor with analogue in and out, and its sole function is to correct in-tuning on the fly. To do this it analyses any incoming periodic signal milliseconds processing delay) without glitches and with a virtually immediate response—it only takes a few cycles for the unit to recognise the pitch and deal with it. There is a big red display on the front that shows how much retuning is happening, and most of the time this is the only way you'd know it was working. If it starts dragging things into the wrong place because you're using the wrong scale, it's only the fact that it no longer sounds like the harmonic of the key away.

This takes a bit of getting used to, but it doesn't stop there. There are a few user controls to allow for different performance styles, particularly how the correction will deal with vibrato. A response-speed parameter allows short and long vibrato to get through untouched, or set it so it will virtually remove the vibrato. It's even possible to add vibrato in the same way as on a synthesizer, with variable rate, depth and delay. There is a sensitivity control for dealing with less than perfectly clean sources and signals with a lot of variation in the periodic repetition. Overall, detune shifts the fundamental reference useful in those countries that like their A at a rather brighter +2 or instances. It doesn't get corrected to the nearest note in the scale. Notes can also be bypassed so they'll be left alone. The ability to choose which notes to allow makes it possible to set up all kinds of modes and to accommodate jazz scales, blue scales, passing notes and anything else that doesn't conform to straight major or minor scale. Even on a lot of those, particularly on the jazz and classical side, to be hidebound by the fact that some notes are wrong, that the chromatic scale is likely to be the standard starting point in order to avoid perfectly good notes being dropped off by a sensor to fit the scale.

Because that, believe it or not, is what the ATR-1 is capable of doing. Presented with a reasonably clean monophonic line, it really is able to retune notes in real time (with a few features: unbalaced and balanced output, Auto-Tune and MIDI controllers, auto monitoring and timer record and playback.

Cassette deck

Tascam has announced the 130 cassette deck offering 3-heads, 10% pitch control, Dolby B/C and HX Pro in a 3U-high rackmount. Connectors are unbalanced and the machine features automatic tape selection, auto-monitoring and timer record and playback.

Low-pro PMC

PMC has designed the new low-profile version of the B15 for 5.1 channel work with mirrored pairs of speakers and centre channel singles. Options give 1.5dB and 3dB cuts on the HF and a matching sub bass is available.

October 1998 Studio Sound
THE AMEK DIFFERENCE
A TOTALLY FLEXIBLE APPROACH TO RECORDING AND MIXING

Whether you’re working in music recording, broadcast or post-production, the AMEK DMS can be customised to suit your application precisely.

Uniquely, the DMS is built around a revolutionary 32-bit floating point DSP core, a highly flexible chassis design and a fully modular I/O system, which allows it to be designed in any shape and system configuration.

Its powerful software resource enables the same DMS to be reconfigured for music recording, broadcast and post-production – all on the same day, if that’s what is needed. For every different operator, DMS can be a different console.

A modular system, AMEK DMS can expand as your business grows, which means you don’t have to invest in features or hardware until you need them.
StudioComm 68/69

Multichannel mixing can now be realised affordably but monitoring control is different. Rob James reports.

OVER recent years the entry-level cost of mixing and recording in multichannel surround formats has dropped to the point where it is a viable option for studios operating on very modest budgets. Anyone attempting to follow this path will, however, quickly become aware of a problem area: monitoring control.

For those in the market for ‘big gun’ consoles the position is easier but costly. The options are to order a purpose-built film console to spec or to bolt on complex additional monitoring facilities. The Studio Technologies StudioComm Model 68 Central Controller and Model 69 Control Console are designed to bring proper, affordable, surround monitoring facilities to small studios and tracking workstations.

On the back of the Model 68 Central Controller are a row of 16 muftiport trim pots enables input adjustment through a range of ±2dB. This allows for extremely accurate level setting. Two LEDs indicate power on and data receive, and that is it.

Audio I/O uses three 25-pin D-connectors. These, useful, follow the Tascam DA-88 conventions on the inputs, and loosely follow on the outputs. A 9-pin D-connector joins the Model 69 remote controller, and a further 9-pin D-connector is provided for remote control via GPIs in the console. One of the GPIs can be used for for an input switch. This could, for example, be used for automatic PEC Direct switching via a record tally. The others allow interfacing with talkback and so on.

The Model 69 control console is a small, neat, desktop mounting unit which connects through a 9-pin D-connector on the rear.

All the keys have associated indicator LEDs. The top row starts with a mode switch. This toggles the function of the 10 keys and its output between MONITOR (on the left) and STEREO A+B. The right side is dominated by a large white LED knob. Associated keys select DISPLAY, primary reference level and switch the LED on and off in pairs. Everyone has their own preferences as to how monitor controllers should operate. One of my pets is the action of mute solo indicators. I expect the unit to light when outputs mute. I feel this logical and makes it easier to work out why everything has suddenly gone quiet. The 69 mixes the logic in that the input indicator lights up when active, but the individual output indicators are lit when not muted. In some modes the solo output flashes perfectly. After a conversation with Studio Technologies the unit convention is to be changed so maybe my view makes sense.

I would have liked separate meter outputs and Studio Technologies tell me they will consider including them. A luxury addition would be the provision of parallel audio I/O sockets. This would make installation a doubley with desk layout and planning, but no surround monitor outputs.

The 69 console is highly configurable. A tiny recessed button on the rear enters the configuration mode. Only four output channels are required the spaces can be disabled. Similarly, where inputs with less than six channels are used, the redundant input channels can be disabled, and input and mix levels programmable. You can be 10, 15, 20 or 2dB down. A nice safety point is when dim is active, using the rotary control to change level, causes dim to automatically turn off. Another sign of attention to detail, which really appeals to me, is that red flashes when the level pot is at the preset level.

The output mode can be locked into surround only or left selectable via the key. Output muting may be selected to operate either via the output relays or the analogue input switches. The inclusion of relays is most welcome. Although a relatively costly option they provide the best protection against speaker damage during power up/down or interruption. I found a software bug that, in some circumstances, resulted in thumps at power up despite the relays. Studio Technologies assure me no units will ship until it is fixed.

The inputs' keys allow film-style PEC-Direct switching or, by simply pressing two or more input select keys at once, inputs can be on top of each other.

The Model 69 69 would be ideal for a setup where 6-track recording is used for 8-channel surround. The extra tracks can be used for a stereo reduction M/E, or whatever, and the LR bypass input and remote bypass function enables easy coexistence with the monitoring on a stereo console. With very little work, 69 69 can be integrated into an existing studio. They provide a simple, versatile, means of controlling surround monitoring with a enough extra facilities and programmability to grow with the studio, without changing the way stereo monitoring is done. I think this is a good thing.

CD printers

The MediaForm CDP-CP2 inkjet printer offers 1140 x 720dpi printing, and is compatible with Windows and Mac systems complete with templates for the most popular desktop publishing packages.

Camcorder receiver

Described as the smallest diversity receiver and developed for the new generation of digital ENG camcorders, the Sennheiser EK 50/U is weatherproof, the size of a cigarette packet, and slots into Sony SX-Philips LS6K120 and Ikegami HV77 camcorders without additional cables or adaptors.

The unit has 23-channel switchable frequency operation within a 24MHz window and can be supplied for operation in the 900MHz to 950MHz UHF bandwidth. It can be partnered with any current Sennheiser UHF transmitters including the SKM5000 and SKM3072 handheld and SK56 beltpack.

Sennheiser, UK. Tel: +44 1494 551531

Mastering bass limiter

The DB 10 dynamic bass limiter is a 2-channel bass filter that limits excessive signals in the 0-800Hz region to protect cutter heads and converters from overload. It is designed predominantly to be used cutting and mastering.

At normal bass levels the filter is said to be completely transparent and has limiation only ever it peaks exceed the preset level. The rackmountable unit has indicators and selectors for sensitivity, attack and release on each channel.

Etec, Denmark. Fax: +45 36 450925.

October 1998 Studio Sound
TASCAM MMR-8 DIGITAL DUBBER

Multilingual

with unique Multi-DAW format recording and playback

"It's about time the industry had units like this. We ran it straight out of the box for five days solid. We were able to use our existing WaveFrames to continue editing, while the MMR-8 was recording all the ADR in the file format of our choice." Danny Longhurst, MD The Frame, London W1.

* Cross-platform, multi format capability
* Direct plug and play portability, from workstation to mixdown
* Direct read-write of ProTools Session files, WaveFrame, Sound Designer II, broadcast WAV files and OMF Composition
* Native operation with both Mac and PC disks
* Forward and reverse play lock to Biphasé
* lock to SMPTE, VITC, MIDI Time Control and MIDI Machine Control
* 8 channels of 24 bit PCM playback and record from single or multiple drive configurations.
* Analogue, AES/EBU and 25-pin D-sub TDIF I/O
Korg D8

With the latest in the line of digital multitrackers, Korg has entered the fray. Rob James says it is uncompressed yet compact.

The KORG D8 joins the crowded portable personal multitracker market with rivals from Akai, Fostex, Roland, Tascam and Yamaha all vying for a piece of the action.

An immediate impression is made by the size of the D8. It is small. The power supply is a separate (non-plugtop) brick that goes some way to explaining how this has been achieved.

In essence the D8 is an 8-track recorder/editor at 14 kHz uncompressed with a 12-channel, 4 bus mixer, and an effects unit. A maximum of two tracks are recorded simultaneously with up to eight available at playback. The total of 12 mixer 'channels' is arrived at by adding the 8 playbacks and the four inputs.

Once recorded all processing is in the digital realm. Converters are 18-bit linear and internal processing is 24-bit. The two analogue inputs are balanced 1/4-inch jacks. Input 1 has switchable impedance to allow direct connection of a guitar. These jacks are located on the front edge of the unit, recessed with trim pots, the headphone output, and level control, and a footswitch socket. The headphone level is, in some parts, perfectly adequate. On the right-hand side of the unit is a SCSI socket for external storage. Removable drives are supported in software. The panel also houses optical S/PDIF I/O and phono sockets for analogue aux in and I/O aux out. The rear panel carries MIDI and output jacks. Songs can be loaded up to DAT and restored using the optical I/O.

The work surface follows the conventional pattern for this type of device. Mixer section on the left with six mono strips which can be paired for stereo and one stereo strip plus master. Transport controls are bottom right and the housekeeping, effects, display, parameter wheel, locators and edit keys occupying the rest of the space. The display is reasonable, but not the best I have seen, and there is no back light or track display.

Up to 50 songs can be recorded per disk. The internal storage is a 1.4Gb hard disk that gives a total recording time of 34 minutes if all eight tracks are used. Twenty scenes, three locations and 100 marks are allowed per song.

EQ is 2-band on each channel, and offers ±15dB at 100Hz and 10kHz.

The scene memories provide a powerful automation tool. Unlike several of the D8's competitors scenes may be recalled on the fly at specific time points. Scenes are editable, but transition time is fixed.

Audio editing is basic, but fairly easy to use. The three locate markers are used to specify the in and out points and, where appropriate, the start point of the scene. Creating may be performed on single tracks, adjacent pairs of tracks or the top or bottom group of four tracks.

A 2-channel internal multi-effects unit gives 50 types of effect with 88 possible chains of up to four processing blocks. Sixty factory presets are OK. Nothing jumps out, but equally there is nothing less than adequate. The rotary speakers are above average, never will not win any prizes but then, a really decent reverber would cost several times as much as this entire unit. I really did like the 'turnable effect complete with 35/5pm click!'

In operation the D8 is a mixed bag. The basic stuff is fairly quick to get to grips with. Songs are easy to load and save, and general navigation is good despite the lack of a time-line display. However, the effects editing takes a while to become instinctive involving multiple key presses which are not particularly intuitive. As I have noted before, the price you pay for a big feature set in a compact and cost-effective machine is the steepness of the learning curve.

The little Korg has much to commend it. If you are looking for the most compact do everything multitracker this one must be in the running. It also scores with scene memories which can be recalled against time. Another point in its favour is the absence of data compression which validates the possibility of using it for serious work.

There are now many broadly similar machines in this market. The differentiation is mostly subtle and it has never been more difficult to decide which one may be right for you.
Legendary Sound to Make Your Dreams...

The Studer V-Eight is an 8 channel 20 bit digital recorder based on the ADAT™ type II format, using S-VHS cassettes. The V-Eight is 100% compatible to all current ADAT formats with over 100,000 units sold. The professional design and reliability will give you a cost effective, faithful workhorse for all professional audio recording applications. The V-Eight features a professional S-VHS Tape Drive for extremely fast and gentle tape handling which leads to substantial time savings. The V-Eight also has the convenience of an integrated TC generator and chase synchroniser. Unique features are: 24 bit Studer converters based on the legendary D-827 DASH recorder technology, to improve the sound of your recordings and an On-board 9 channel monitor mixer to make live recordings without a mixing console.
Location Recording

Hera's how it works...

Deva's dual disk recording process records up to four tracks of 24 bit audio with up to 10 seconds of Pre record audio (no missed cues) on 2 disks at the same time. At the end of the day, send a DVD, Jaz or Syquest disk to telecine and then to post and keep the Deva internal drive as a backup.

KEY FEATURES:
- Audio quality far superior to Dat or analog Nagra.
- Save up to 25% of the time required in telecine.
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- External SD2 and Broadcast wave recordings compatible with virtually all DAWs and Dubbers.
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USED ON THE FOLLOWING PRODUCTIONS:

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From the Earth to the Moon

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Stuart Little (TBA)

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Clueless
Meet Joe Black

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Sometimes catching a take on location is a matter of life or death—and sometimes its worse. With a vested interest in preservation, Neil Hillman surveys the location recording scene.

I TRY NOT to dwell on my demise from this life; although events can overtake the conscious mind. That 'for it is not death or hardship that is a fearful thing, but the fear of death and hardship' was uttered by the ancient freeman Epicurus, counted for little when the trauma of my holiday accident unfolded. Certainly I rehearsed what my last words should be and what a suitable epitaph for my stone on head- phone hill could be—something witty like Oscar Wilde's. 'Either this wallpaper goes or I do or a simple, moving inscription like that of delta-blues slide-guitarist of the 1950s, Blind Smashing-Pumpkin. 'Didn't wake up this morning.' I did in fact break my ankle. On holiday, in France, running along a cliff path.

I hobbled heroically home and surrounded by my caring and concerned family. I lay and waited for the ambulance 'caring and concerned' because my wife, rather too obviously given my condition, cared that her hair was okay for the dashing French paramedics. My children were more concerned that they would miss their outing to the Astérix adventure park.

At first I thought I was going to die. After 10 minutes I was afraid that I was not; but the administration of a suitable pain-relief injection soon had me drifting into the arms of Morpheus where a man will see his worst, sinful, dust.... replotted so that for one last, delicious time he may savour the sweetness of forbidden fruit, stolen during the profligate periods of his life. That's the theory anyway. What I got as an hallucinogenic experience turned into a narcotic nightmare—wracked by the dilemma over which to record, and on which format, my last words should be captured on.

StellaDAT II has a heavy-of-stone build quality, with 16-bit, 4-track recording at either +4.1kHz or 2-track 9kHz and a lower power consumption than its rivals, but with a price to match its high pedigree. User-configurable setup software offers almost total access to the machine's variables via menus, and an onboard joystick. M-S monitoring and phantom power is offered, and along with the four inputs and outputs, the time code in and out, and two AES I-O pairs are all on XLR connectors; BNC connectors route sync in and out.
would all return less than 10 miles to the gallon.

Tascam's DA-P1 rests at the budget end of the Pro-DAT market, but can still sit comfortably shoulder-to-shoulder in terms of quality with the Sony TCD-D10, selling at more than twice the price. Another 2-channel, non-time-code machine; its balanced mic line inputs are on XLR connectors with switchable +48V phantom power, while its line out and digital I/O SPDIF signals appear on RCA phono sockets. A robust, switchable limiter keeps the lid on heavy mods and the constantly updating headroom margin, shown on the backlit LCD in dBs, inspires some confidence in the absence of off-tape monitoring. Sampling rates are the normal 44.1kHz. 48kHz and 52kHz. The DA-P1 lacks the monitor speaker sported by the Sony TCD-D10, but by saving well over £1500.00 you could go out and purchase the best headphones money can buy. Alternatively, stock-up on recorder batteries.

The Zaxcom Deva, when it was launched, created a huge amount of interest not least of all for being a hard-disk location recorder that was completely resistant to the bump and grind of location life thanks to a RAM store of PCMCIA recorders, with 2-channel progressive technology, are primarily aimed at newsgathering journalists, but (a source of frustration up to now, given the potential of this medium) have yet to court the film and television sector. They use removable PCMCIA cards that decrease the transfer time to a workstation and record either linear or compressed MPEG, or combinations of both, and byte-for-byte give the highest storage capacity at the lowest price.

Simple unidirectional, non-destructive editors are also incorporated to allow the preparation of reports either to be played in live via on-board ISDN codecs or modem connectors, or given as a finished-edited "wax file" on flashdisk to the studio.

The Nagra Ares-C is a comprehensive MPEG recorder with balanced XLR outputs and line microphone inputs offering +12V T. +12V and +48V phantom power, gangable input pots, standard Nagra filters of "FL" 'speech' and "FL", and a truly straightforward editor with an inbuilt ISDN codec, or SMPTE-EBU time code.

Mandozzi's DA-P1 carries more than a passing resemblance to the Nagra B with balanced XLR line-mic inputs with phantom power, XLR line outputs and XLR AES digital output, right down to the large single function key enabling the machine to record or playback its linear audio. Where it differs noticeably from the Nagra is the commumate case with which the machine can be configured to alter recording parameters such as mono or stereo, bandwidth and titling individual recorded tracks.

If the Ares-C is the nearest to looking like a conventional recorder, the Sonifex Courier is by far the best looking. But its beauty is more than skin deep; this is a serious-minded tool, offering linear or MPEG compressed audio, balanced XLR line-mic inputs and line outputs, phantom power and FL filters plus off-disks 'confidence monitoring with a reassuring 'record delay'; dependant on which level of compression has been selected.

Your com's ReporterMate MTR-61 has two, balanced, line-mic inputs with phantom power, line out and AES-EBU digital in-out—all on XLR connectors—to access recording and playback of either MPEG or linear audio. The mixer offers gain compression, limiter, and a voice-over mixing function. With a dual card slot, the PCMCIA cards may be 'hot-swapped' during recording.

The design of the Eela S40 Reportable benefits the extensive experience Eela has in the remote communications field, with the Reportable looking very similar to the company's telephone-mixer reporting products. With balanced XLR connectors for line-mic in and line out, but no digital output, the S40 records MPEG in a .wav format at a sample rate of 48kHz.

The Marantz Easy recorder, like the Zaxcom Deva and the Eela S40, offers time-shift recording which enables the machine to 'record-listen' before the arm-on button is pressed by means of a RAM store that, depending on the settings chosen, can store up to the previous one minute. Using either MPEG compression or linear .wav recording formats, the Easy recorder offers XLR balanced line-mic inputs with +12V phantom power and XLR analogue outputs; two card slots and two integrated speakers.

If we are compressing audio, the Marantz PMD-650 MiniDisc recorder has the location MD market to itself, all other professional location MD recorders being taken from the domestic market and respectively cleared out and replaced with the Marantz S40. This recorder, designed as a field acquisition tool, with balanced XLR line-mic inputs offering +48V phantom power, balanced digital output, limiter, built-in speakers and channel recording, has the ability to record through its line inputs from digital sources with sampling frequencies of 44.1kHz. 48kHz or 52kHz.

If you are still sold on open-reel recording the Nagra-D is arguably the most sophisticated location recorder available today is—like the Zaxcom Deva or the Nagra Ares-C, the Marantz MD-R30 or the Denon DMP-R70. The Marantz, however, has been designed as a field acquisition tool, with balanced XLR line-mic inputs offering +48V phantom power, balanced digital output, limiter, built-in speakers and channel recording, has the ability to record through its line inputs from digital sources with sampling frequencies of 44.1kHz. 48kHz or 52kHz.

The four hours of recording. if you are recording and 8h for replay. Collating four tracks of audio onto a removable 2-inch IBM PC disk, the Deva also offers an inbuilt 4-channel mixer with assignable onboard equalisation and effects and, new for this season, the ability to interface with a scaled Panasonic DVD-Ram drive. These slave recorders can be made on location simultaneously as the Deva's internal drive record, in an attempt to get around the sticky problem of recording media costs. With selectable sample rates of 48kHz, 48.01kHz or 47.952kHz, and supporting all time-code rates, its impressive. If only it looked less like a piece of test gear.
Welcome to an oasis of real satisfaction, where your thirst for the Whole Truth and Nothing But the Truth will finally be quenched.

For Nearly 20 years we've been known for our active monitoring systems, particularly our compact, nearfield bi-amplified ones. But outside the nearfield, where the heat really gets turned up, Genelec's S30C, 1037B and 1038A integrated tri-amp* active monitors are designed for bigger spaces - mucho grande.

*Amplifier modules may be rack-mounted in any soffitt installation.

The Whole Truth And Nothing But The Truth
The film reviews have been damning, but the public has flocked to the cinemas regardless. Rob James investigates the quirky and shady regions of The Avengers sound world.

The television series The Avengers began life as Police Surgeon with the late, great, Ian Hendry avenging his wife’s murder. Patrick MacNee played a well-heeled, suave, debonair civil servant. When Hendry left to do a movie the series was renamed and the clipped tones of John Steed were joined by Honor Blackman as Cathy Gale. Blackman went on to do Goldfinger and Diana Rigg’s Mrs Emma Peel took over as the series moved from black and white into colour, and brought dark catsuits and sharp brains. Rigg was succeeded by Linda Thorson, but MacNee remained until the series was dropped in 1969. It was revived as The New Avengers with Gareth Hunt and Joanna Lumley in support. In its time, The Avengers captured the imagination of audiences across the world.

Since every other similar UK TV series of any note seems to have spawned a movie—sometimes with dubious results—it is surprising it has taken until now for The Avengers to hit the big screen. But where the television plot lines tended toward the fantastic, and the movie is no exception both in plot and production, the director and dialogue writer seem to have missed the point. The TV series were always tongue in cheek, but the characters had depth and there was real chemistry between them. In the film the leading characters are flatter than a surface plate and even Sean Connery appears to be on auto-pilot. The only vaguely sympathetic acting comes from the minor characters including the transparent archivist, a cameo role for Patrick MacNee as I (for invisible) Jones. The spectacular weather and other visual effects not to mention the slick and involving mix sadly cannot redeem the rest.

The early TV episodes were broadcast semi live, and as the series progressed so did the production methods leading later episodes to be shot on film. Given the intervening technical advances, the film was edited by Mick Audsley on Lightworks with the majority of sound postproduction undertaken at De Lane Lea in the heart of London’s Soho. Adrian Rhodes began working on The Avengers in October 1997, initially as effects editor and is currently best known for his sound design work on the Wallace & Gromit animations. Director Nick Park and Rhodes were at the National Film and Television School together. The high profile of Wallace & Gromit has slightly obscured the rest of Rhodes’ work which includes, lead mixer on The Full Monty and Hamil Karesh’s The Fanatic and effects editor on the latest Bond epic Tomorrow Never Dies. Danny Langhans was dialogue editor.

Ian Wilson did Foleys. The team shared two assistants, Guy Hake who, according to Rhodes, was also indispensable as the team ‘boffin’ and Mark Rose. This compact sound team illustrates the way digital tracklaying is revolutionising the business. If the tracklaying for a movie of this complexity is undertaken on 35mm magnetic film the number of people involved could be expected to at least double.

Location recording by Clive Winter was on a standard ¼-inch Nagra job with time code: although the majority of the dialogue was replaced in ADR sessions due to the large number of special effects intensive shots which rendered much of the location sound unusable. The film rushes were transferred to Betacam, the sound synchronised and copied to the Beta tapes and sub-master DATs—all fairly routine stuff these days and caused few problems.

By November the first of many temp mixes took place in De Lane Lea’s Studio 5. This is one of the smaller theatres in the complex, equipped with an AMS Logic 2 console. These mixes are produced for a variety of reasons; to get an idea of whether the film is working, to help determine what dialogue needs replacing and to assist in designing the track. The dialogues were prepared on a Waveframe then digitally bounced to DA-88 since this was a delivery require-
ment. The DA-88 tapes were digitally bounced to Akai for mixing. ADR recordings were made in London, Shepperton and New York, recorded on whatever the individual ADR theatres usually employ and then loaded into the Waveframe for editing. Foley scenes were recorded, edited and delivered on Akai (a DR16 in this case). Rhodes used his Spectral Synthesis Audio Engine for effects editing with the results digitally transferred to Akai for mixing.

It is generally considered undesirable to mix for cinema in a small theatre, but Adrian Rhodes reckons. We did the mix in Studio 5 because it was available and I am used to the Logic. But to be safe we did the SVA (Stereo Variable Area, a term used for a Dolby Stereo Matrix encoded master) in the much larger Studio 5, and then checked the mix on one of the screens at the Warner Village in Leicester. It was absolutely fine. When Studio One at De Lane Lea was designed, the decision was made to eschew rows of magnetic film machines in favour of a bank of Akai DDP digital dubbers. Akai machines are also used as recorders in the other studios.

There were several subsequent temp mixes in editing progressed. This would usually require a complete remix or copying and editing of premixes but Rhodes found a better way. All the mixes were recorded onto Akai machines, he explains, and we tried out the auto reconstruct that was introduced a while ago. It was a highly successful venture. Everything worked and we just had to fill the holes where new shots had been cut in.

The effects gathering and creation started right at the beginning and continued throughout the postproduction period. Meanwhile I was adding to the effects. Rhodes continues, 'I made heavy use of an E4 Tinnitusator and SPL Vitalizer plus various Yamaha MPXs and Lexicon, but, really, it's just using a lot of imagination, you don't need to spend a fortune on toys to achieve results.

The SFX shots were perpetually changing, background lightning hits would happen in different places. Things come and go, and you have to keep updating. At least on The Avengers we didn't get to see all the shots, so The Borrowers some shots we never did see and just had to make the best guess.'

The mechanical bees effects were made from a variety of unlikely parts. 'An electric razor and an electric toothbrush, Doppler shifted and pitched with the f-mu, and I did find a simple way of achieving an organic bee sound using a helicopter and single engined plane up and past with some onboard pitch change courtesy of an Evenside. Mix that lot together and you get some organic mechanical bees.

Many of the backgrounds are constructed from equally imaginative sources. 'I have a huge library of wind sounds, mostly mouth produced. Your mouth is a very versatile instrument for effects work. I am also the voice of the Prime Minister. I did it for one of the temp mixes and they kept it in.'

The score was to have been composed by Michael Kamen, but, if Warners is to be believed, due to the complexity of the visual effects, the film was late and Kamen had another booking that prevented him completing The Avengers. As a result Joel McNeely is the man responsible for the mighty score with its occasional well handled homage to the TV theme. The music recording took place at Abbey Road with Shawn Murphy engineering and some programming done in the US. The resulting material was presented for mixing on Pro Tools 24 with a Genevac backup that, in the event, was never required. It is an interesting sign of the times that there was insistence on staying 24 bit with the music. How long will it be before 96kHz sampling or maybe even 192kHz becomes a production requirement?

The final mix was originally booked into Shepperton Sound, but, again due to the delays, it was eventually decided Adrian Rhodes would co-mix the film in De Lane Lea's Studio 1 with Mike Prestwood Smith who had recently joined De Lane Lea from Goldcrest and brought considerable Harrison and Synclavier experience to the party.

Being a Logic user I found the Harrison Series Twelve quite difficult to start with,' Rhodes recalls, 'but the learning curve was really quick because ultimately, it is a pretty standard console. It's just the automation which is tricky.'
to begin with having learnt on a different desk with a processing pool rather than fully-loaded channel strips. There seems to be an automation mode on the Twelve to suit everybody." Standard maybe, but with a phenomenal number of inputs.

When this desk was specified, I thought full automation seemed like overkill but on this film we used them all. On Reel 3, with the mechanical bees, and on the final reel we even had to use some predub inputs as well. To make it as exciting as possible there are all the elements contributing, fire, wind and water.

To leave things as open as possible the effects were premixed onto 32 tracks with mono, pan groups, two subs and two stereo scenes. Foleys were mostly across 16 tracks, and there is a serious amount of Foley. This is a big effects movie.

Dialogue premixes covered a further 16 tracks with mono, pan groups, split reverb and subs with the music score coming direct from Pro Tools and source music supplied on DA-88. The team used some fairly novel approaches to the final mix.

We had the dream of doing a totally "virtual" mix, but the picture goalsposts were moving to fast so we had to compromise. All the premixing and final mixing was recorded on Akai DDoS onto MD discs, so we never had to bother about disc clearing or archiving. The mixes were done "virtually" reel by reel. In other words, no audio actually gets recorded until the reel is complete then we transfer it to disk in one pass with the automation doing all the work. We even used the time-code automation of the Lexicon 600 to take care of any reverber changes. It all behaved perfectly.

The final mix was spread across three DDS split into Dialogue Effects and Music each with LCR, LS and S channels. The delivery requirements included mixes in Dolby SR and the accompanying SVA, DTS and DDS.

We simply ignored the extra effects channels, says Rhodes. "This is okay with the DDS people and to do anything else would have required a substantial remix."

As a matter of interest, both the DTS and DDS mixes were supplied on DA-88 tapes lending more proof to the ubiquity of the format. I live in hope a more robust disk-based format will supersede it. Meanwhile there are interesting lessons to be learnt by other manufacturers. In particular the relatively easy interchange between US and other time-code formats is one of the reasons for its success.

And the mix is, indeed, remarkably dynamic, clean and punchy. Every word of the dialogue can be heard without the usual strain and sensible use of the available dynamic range has allowed for real shocks.

There is a third of a second around the moment that effects levels are getting too hot in all the digital formats at the expense of dialogue intelligibility and also safety. If the dialogue mix is right, using the available dynamic range is the key, just because you can go loud doesn't mean you have to all the time. In fact, if you do, you lose impact."

Rhodes regards the idea of working as effects editor and mixer on the same movie as a wonderful way to work, but uneconomic from a company point of view. But, if I am honest, I am a mixer at heart so that's what I now am. After the recent change of ownership he is also a share holding director of De Lane Lax along with Paul Hamblin, Dave Old and Mike Solinger. They have some exciting development plans and it will be interesting to see them put into effect. The postproduction route now followed by The Avengers is a good indicator of where things are going. Virtual mixing is almost a reality and will open up a myriad of possibilities. Digital dubbing is coming of age. It is not only viable but offers huge advantages to facilities and producers alike, especially when the film is in an almost constant state of flux until the last minute. □
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Jim Reeves

Manhattan melodies

A star-studded career throughout the sixties, seventies and the eighties has failed to calm Jim Reeves' enthusiasm for music and recording. Dan Daley talks tunes and technology.

Born in Westchester, just north of New York City, in 1943, Jim Reeves comes with the kind of pedigree often found in engineers of his era. He moved, when he was four, to live with his aunt and uncle, virtually in the shadow of Manhattan's 59th Street Bridge, and attended piano lessons aged seven. By adolescence, his family were tolerating the technical curiosity that often developed others into the autodidact recording engineers of early rock 'n' roll. The piano became neglected for the musical mode of the era on the streets of New York: doo-wop under the street lamps.

'We were always getting buckets of water thrown on us from windows,' recalls Reeves, now able to laugh at the musical Philistines that once infested Manhattan. He found himself in good company—with his group the Melodies, he sang at club and school dances six nights a week with rising stars like the Del Satins and the Companions (and doubling with an a cappella jazz group Four Score). did orchestral dates at the Toxedo Ballroom, and once even had a gig interrupted by an impromptu on-stage appearance by Johnny Mathis.

Reeves kept this up throughout high school, and upon switching from the Catholic Cardinal Hayes High to the more practical Brooklyn High School of Automotive Trades, he was able to put what he gleaned about electronics during years of teenage tinkering into practice. He built his own recording console and hooked it up to a pair of creepy, academic-grade Wollensak mono tape recorders, and a Shure SM58 perched on an equally creepy Atlas boom stand. 'I'd been fooling around for years putting together little mixers, but this was the first time I was able to actually machine something,' he recalls.

After graduation, he was working the cash register at his uncle's chemist shop on East 59th Street, slipping off when he could to the recording studio at Kapp Records, upstairs in the same building. Here, chief engineer Grant Elliott showed young Reeves the ropes of real studios. Then, as he describes in the personal narrative style unique to native New Yorkers, 'Leo at the Dover Deli across the street says, 'Let me introduce you to Dave. You want to know him''

Dave Sarser was a violinist in the NBC Orchestra, conducted by Arturo Toscanini, this being the days when broadcast networks still considered philharmonic orchestras to be the audio flagships of their ethereal empires. More importantly to Reeves, Sarser was also technologically inclined, having helped Les Paul put together his first 8-track recording system—assisting in the design of the first headstock and acting.
as the liaison between Paul and Ampex, who constructed the towering early multitrack system.

I had been trying to do sound-on-sound recording at home on the Wollen
saks, so Sarser became my guru, and you couldn't ask for anyone better con-
sidering what I was into,” says Reeves, who continues to stay in touch with his mentor via phone calls and e-mail. Sarser helped Reeves get a job as a porter at Studio 3, the East Side facility owned by the legendary Louis Ton-
derson, who at the time was the band-leader on the original *Tonight Show*,
then hosted by Jack Paar in the pre-Carson era.

While the proximity to clients like Ruby & The Romantics and Vladimir Horowitz led Reeves obsession, cleaning the studio's toilets did not let him get any hands-on recording experience. So he looked around at some of the burgeoning independent studios that were starting to dot Manhattan's West Side, where the Brill Building was the musical Mecca of the time. He stuck pay-
dirt at A1 Sound Studios, where owner Herb Abramson, who had been a partner with Alpert Eritgian in the start of Atlantic Records, gave him the job as an engineer.

He was ostensibly to assist Abramson at sessions, though Reeves remembers, “He would keep saying, ‘You set up and get things started, I'll be along.’ It got to the point where I was doing the entire session and he was just showing up in time to collect the money. But that was fine. That's how I got the experience I was looking for. I also worked up doing sessions for lots of artists, including Ruby & The Romantics, and the Supremes, who came in to do songs on a memorial record for Sam Cooke, who had just been murdered. I remember we were recording a 3-track at the time, and the Supremes were incred-
ible pros, but so nonchalant about it all - they sat on stools smoking cigarettes during the sessions, putting away between lines of the songs, and never missing a cue. I had a Neumann U47 on each of the girls, and I put it up near the control room glass in the studio.

It was a long, loft-like studio design, which a lot of the rooms were back in those days, and we had the band down at one end, and the brass at the other, and the vocals in the middle near the window. So you were getting these great reflections of the drums off the glass and into the vocal mics which gave the whole production a real sense of space and roominess. There was also a natural delay between the band and brass sections that affected the way people played their parts and the feel of the record.

I was having a great time and learning a lot, too. Then Skitch found out I was doing engineering at another studio on the side and asked me why I did not do some engineering for him.

Reeves did, running sessions for some of the same artists he had once cleaned up after, including Benny Goodman (who had a habit of throwing his ciga-
ettes burning on the lid of the classic Steinway given to Henderson by John Steinway himself). He also did numero-
ous commercial jingles-productions for Henderson's commercial company, Clef Ten, were the studio's bread-and-butter clientele—including naming the desk as Sarah Vaughan sang the Tasty-Cakes jingle and editing those jingles, which proved to be a baptism of fire.

The Wollensak tape recorders were really not designed for editing,” he recalls. You had to hold your finger on the capstan to get the tape up against the heads and we were working down at 3 ips.

All this was being done on the con-
sole that had been designed and built by Reeves's mentor, Dave Sarser, who consisted of Alex line program amps on a panel purchased at auction fitted with Bradley 100k faders. 'It was a sort of broadcast design, which is where con-
sole design was really coming from in those days,” says Reeves. 'It had nine inputs and three channels out with a phantom centre because mixing fool stereo was still being worked on at that time. It was a very cool-looking console.'
<Crystal Blue Persuasion> when he brought a Moog voltage switch into that studio one day to experiment with an early morphing effect between the guitar and vocal tracks. It was kind of like toggling between them, but it didn't really work," he says. 'His next record was Crystal Blue Persuasion—which I didn't engineer—but that same idea was on that."

Reeves ultimately bought Studio 3 from Skitch Henderson. However, his first stint at studio ownership did not end well. Reeves partner in the venture, whom he declined to name, brought in what in genteel circles are referred to as 'unsavoury characters', more contemporaneously known as 'wise-guys', who in short order took over the business themselves.

Reeves did not register complaints with the Better Business Bureau. 'I was more interested in staying alive,' he says. Studio 3 on East 57th Street was turned into an after-hours nightclub that was frequented by celebrities, including Judy Garland shortly before her death. Meanwhile, Reeves spent three months in the Caribbean recovering from the episode. It cost him his business and his domicile, since the studio also included a townhouse where Henderson had originally lived.

Returning to the US, he did some record productions but realised he needed a job with more definite prospects. Working with a band at Columbia Studios on West 52nd Street in late 1968, he became friendly with an engineer; you had to be friendly with them, since at the time Columbia was a union house and union rules stipulated that only house engineers could so much as touch the recording desks—who steered him towards Roy Friedman, the studio manager. Friedman gave Reeves a tape to edit as part of the studio's standard hiring test. The two met in one of the studio's numerous control rooms and chatted. A few moments later, Friedman readied to leave and let Reeves get on with the test. Whenupon Reeves handed him back the tape with precise edits already done. 'I used to do a zillion edits a day for Skitch, I did them while I was on the phone. I could do them in my sleep, he says. 'I just did the edits scrubbling at a low volume as we talked. He looked in his union book, found the highest-paying engineering job category, and gave it to me on the spot.'

At Columbia over the next four years, Reeves engineered hundreds of records as the label established itself as a leader in the burgeoning rock industry under Clive Davis' management. These included Johny Winter And...; the Chambers Brothers' New Generation; the Brecker Brothers' Billy Cobham collaboration Dreams; Tony Rish's Wrong End of the Rainbow; and Bob Dylan's collaboration with now ex-Beatle George Harrison on what would become Wallflowers. He got to work with a litany of the best producers of the decade, including Roy Halee (Simon & Garfunkel) and Bob Gaudio, whose Four Seasons' records resonated in Reeves' doo-wop soul.

'Bob Gaudio was a great producer,' he says. 'He was the one who taught me how to record and produce a vocal take. He used a line-by-line approach with Frankie Vallie, and he knew how to cash in on every word. Before Frankie could do the next syllable, let alone the next word, of a line, Gaudio would stop him and coach him. Then we'd record it, punching in line by line, word by word. Later I used the same thing on Tom Jones. You really hone each performance that way, and every word really counted. We used to go till 7am, doing vocals like that and then we'd walk over to Central Park and eat hot dogs for breakfast.'

Reeves facetiously recalls that in those days he was stuck with Neumann U-47s, U-67s and U-87s. 'I mean, that's all there were at those studios,' he says. 'They weren't considered vintage yet. They were what you used every day.' Still, he remained a fan of the Shure SM58, from which he often removed the ball and covered it with foam, allowing singers to get right on top of the capsule. He also refined his mixing techniques during this period. 'I was always into close mixing,' he says. 'I always thought the sound is originating in the speaker, so why put the microphone far away? For guitar amps, I would put an SM58 right on the speaker, on-axis. I never liked the phase...
effects produced when you placed it off-axis to the speaker. Instead of EQ, I would just move it a little to the left or right to reduce the brilliance or sharpness of the sound. On the Jim Factory album, I literally used no EQ at all on anything, relying solely on microphone choice and placement. You learn how things can EQ each other just by how they’re all balanced together,” he explains. At Columbia, he had access to some very interesting consoles designed and built by the label’s own in engineering department on the seventh floor, though he was exposed to more of the off-the-shelf designs when working at A&R Studios, where Columbia sent its spillover work.

At Columbia, Reeves experienced a studio culture that is now only a memory, and not always a fond one, for him, anyway. The unions, in this case the International Brotherhood of Electrical Workers (IBEW), Local 1212, restricted how engineers worked and interfaced with producers. As mentioned earlier, the latter could not touch the consoles. The management used to stare into the windows of the control rooms to check on you as you did sessions, to make sure you were complying with all the rules,” he says. As engineers moved into production themselves, the studio culture reacted by restricting how they were credited. ‘After Freddie Catero and Roy Halee did the first Blood, Sweat & Tears record, Columbia got defensive about giving engineers any credits on records,” he remembers. ‘Columbia was afraid of engineers getting too much control on the records. And if an engineer did get a credit, his name was in lowercase. I always felt like I was just part of the machinery when I was there.”

Reeves found other outlets for his ambitions. While continuing to work at Columbia, he designed and ran the sound system for JPs, an Upper East Side club that quickly became the nexus of Manhattan’s rock culture, drawing all the while on earlier live sound experience gained at Oneline’s where he had mixed shows for Jimi Hendrix and the Doors. A while later, he was involved to a lesser degree in the creation of the sound system at the West 2nd Street club Traxx, of which JPs was a venture partner. While he was ultimately displeased with what he called an overly large sound system in the low-ceilinged space, the venue launched scores of artists, including Cyndi Lauper, Desmond Child and Manhattan Transfer, who liked Reeves live sound work so much they took him on the road with them.

‘It was also a great place to learn more about mixing instruments,” he says. ‘One thing I learned is that the way things sound is as much a function of who’s playing as anything else. I got to work with drummers like Rick Marotta and Russ Kunkel there. I remember one night Rick came in with a tiny drum kit. I mean, tiny—it fit into a regular suitcase. I thought it was a joke kit. Then he started playing and it sounded explosive. It had nothing to do with the microphones. It was the dynamics of his playing that made the sound.”

Reeves also did a dozen or so live records at the Fillmore East, including Sly Stone, Jessie Colin Young & The Youngblood, and Johnny Winter’s classic live record there, done to an 8-track Ampex MJ-1000 deck wheeled into the basement of the venue. During that record, Reeves stepped outside to take a break between shows and impresario Bill Graham’s security team wouldn’t let him back in. Finally, like more than a few devoted fans, he eventually found a way to sneak back in for the second show.

Reeves had also been doing sessions at the Record Plant, another independent studio that Columbia used for overflow. It was here that Reeves saw how much more evolved the culture of the independents had become. ‘The Plant just blew my mind in 1974,” he exclaims, ‘unselfconsciously using the vernacular of the era. Aesthetically, it was totally different from Columbia, which was more like a hospital in terms of vibe. The Plant had a Datamax console back then which had no monitor volume control. But when you hit +2dB on the meters, you were in mixing heaven. And the monitoring was so much better than those nightmare Altec A7 speakers we had at Columbia.”

After leaving Columbia for good that year, Reeves worked at the Record Plant, where he engineered numerous period classics, including Greg Chinn
Class of 67: Reeves at Incredible during a 4-track session with Freddy Scott

< Allman's Laid Back, which spawned the Southern rock anthem ‘Midnight Rider', Reeves used the microphone against the control room window effect again on Allman's vocals into an omnipatterned U67. But he went back to the basics for the strings using Shure SM57s.

The Record Plant became his home base as he recorded an increasing number of live records outside of New York, then brought them back for sweetening and mixing. Along with, he was experimenting with more offbeat microphone combinations, including using Sennheiser 441 for all the vocals on the Mott the Hoople live record.

Perhaps the most interesting of the live gigs that Reeves was invited on was the 1972 Presidential Inauguration Bull, which kicked off Richard Nixon's second, ill-fated presidency. Working for the Record Plant, Reeves recorded performances by country singer Ray Stevens and the Mike Curb Congregation. He also got to meet First Lady Pat Nixon. But the most memorable event for him was sitting by the console as actress, diva, Rat-Pack flirt Joey Heatherton ran back and forth from stage to dressing room doing numerous 30-second costume changes during a single number. That, he says emphatically, was fun to watch.

Reeves was in early on the short-lived Quad freeze of the mid-1970s. Columbia Studios brought him back to do several remixes using the SQ 4-channel system, setting up one of its newest in-house console designs in Studio 409 for the project, a desk that offered one of the first solo-in-place functions, and four of his least favourite Altec A7 speakers. Columbia mandated that all the Quad mixes were to be stereo compatible, but Reeves soon found that the inherent phase anomalies of Quad just got worse when you truncated a 4-channel mix to two channels. The first record to undergo the quad process was a Paul Revere & The Raiders remix, but it was the Super Session recording (Al Kooper, Steve Stills, et al) which got the full treatment.

‘On “Season of the Witch” I wanted to be able to rotate the information symmetrically, and there was no way to do that precisely with the system the way that it was,' Reeves states. ‘So I had to develop a 4-ganged joystick myself.' Reeves once again went to the Columbia Skunk Works on the seventh floor, where chief engineer Eric Porterfield made what Reeves had in mind and was able to achieve the desired effect. However, quad itself didn't fare as well, something that Reeves attributes to poor word of mouth about the format among both consumers and professionals.

'I enjoyed Quad, but I thought it was marketed badly,' he says. The press played up all the phase problems and said it was a novelty flying sound around four speakers. But they didn't say the same about stereo even though it came in people were also doing hokey things like ping-ponging sound back and forth. I really believed it could work, but hadn't anybody killed it.

Reeves believes that current multi-channel formats will succeed where Quad failed because the companies behind it have learned the marketing lesson and are prepared to back that up with money.

The memories, both fond and not-so-fond, have not made Reeves into a museum piece by any means. Though out of the mainstream for a while after working in jingle production in New York for many years, he now runs his own studio and production company from a Chicago suburb, where he moved four years ago, reveling in the ADAT culture of local garage bands and the inexhaustible potential of the music business. And while he is still a bit bitter that credit has not always gone where credit was due—he still smarts that his overdub and mix work on Lou Reed's Berlin album is not properly attributed—he doesn’t let it slow him down. 'I still just really, really like what I do,' he says.

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Africa’s children

If television symbolises Western civilisation, radio is the lifeline of established and emergent society. **Kevin Hilton** charts changes at the Namibian Broadcasting Corporation

Broadcasting is schizophrenic; it has its traditions, with many of the techniques and sensibilities used today, going back to the earliest times of the medium; and its dedication to innovation, which drives it along and helps it progress and develop. As broadcasters start to come to terms with a near fully digital environment, this schizophrenia becomes more apparent: the need to move on, but the equally strong need to keep some of the old ways that have worked so well.

Radio in particular is experiencing such a transition. One broadcaster that is going through changes at present is the Namibian Broadcasting Corporation (NBC); although change has never been far away for either the broadcasting organisation or the country. Currently in the middle of a long-term refurbishment, this state public-service broadcaster is trading in its old reel-to-reel machines for Magneto-Optical (MO) player-recorders in a bid to maintain its reputation as one of the leading broadcasters in Africa.

Bordering Angola and Zambia to the north, Botswana and South Africa to the east and the Atlantic Ocean to the west, Namibia, like many African countries, or indeed, many countries in general, has a complicated, torrid history. The original inhabitants were the Damara people, but in 1884 the country was annexed by Germany, with the exception of the British-Cape Colony enclave of Walvis Bay, Namibia's major port, situated on the Atlantic coast. World War I saw occupation by South African forces, with Namibia being mandated to its neighbour in 1920, from which time it was known as South West Africa.

Being governed by South Africa meant that the first broadcasts in Namibia were under the auspices of the South African Broadcasting Corporation (SABC). The earliest transmissions were merely short-wave relays from South Africa that were then rebroadcast on FM. But as dedicated studios were built, local programming slowly took precedence over centrally generated material, with an independent South West Africa Broadcasting Corporation (SWABC) emerging out of the SABC around 1979.

The centre of operations is the capital, Windhoek, situated pretty much in the middle of the country. Programmes from here were broadcast in several languages: English, Afrikaans, German, Damara-Nama and Herero; studios were later built in Oshakati to serve the Oshiwambo population area in the north of the country in its own language. A station was also built in Rundu in the north on the border with Angola, serving the Kavango region, with another located in Katima Mulilo at the end of the Caprivi Strip in the extreme north-east of the country for the Lozi speaking population. (A transmission service was added approximately two years ago, and there are now plans to institute a SAN or Bushman channel.)

Full independence for the Namibian people came in 1990; although South Africa continued to claim sovereignty over the commercially important Walvis Bay area up until August 1993, when it finally waived its rights to the enclave. With the creation of a free Namibia, the SWABC changed its name to NBC and has continued to expand services to cater for its small, but wide-spread population (estimated at 1.7m). Although not as large as South Africa, the country is still immense, having an area of 824,400km²-318,262miles². (To get an idea of the scale of NBC's operation, Katima Mulilo is 1200km from Windhoek, but is maintained by technicians from the capital.)

Main areas of the country are covered by FM transmissions; although, due to the large sparsely spread population, it is not possible to cover all areas and so new FM stations being implemented all the time. A MW transmitter is located at Tsumeb to serve areas not yet covered by FM, with two SW transmitters outside Windhoek. NBC employs approximately 500 people and has contribution centres in Otjiwarongo and Keetmanshoop, each with a small studio, while a new office has recently been opened in Walvis Bay.

The Windhoek broadcast centre houses around 20 studios, with four studios each in Oshakati, Rundu and Katima Mulilo. The majority of the Windhoek studios are straight-on-air broadcast suites, with one being a music recording studio equipped with a 24-track Studer A827 and Dolby SR. The English language National Service is currently the only stereo channel in the country and runs on the Dalet hard disk-based automation system, with 18Gb of storage (mirrored) giving in the region of 180 hours of audio.

This shift to hard-disk equipment is now becoming more general as NBC looks for replacements for its ageing stock of reel-to-reel tape machines. Phil Schmamann, responsible for the planning and implementation of projects at NBC, explains: 'We have been using Studer B67 and A807 and Revox PR99 tape machines. The Revoxes are mainly used for automated play-out in the evenings and weekends, while there are three A808s and an A827 in our music studio. In total, we have around 220 tape machines in use, most of which are B67s and they are getting really old. As we have seen that the main expense on these machines is mechanical part replacement, we decided to look for an alternative with minimal mechanics.' The choice may have been new MO technology, but NBC decided to stick with Studer, buying 20 so far, with >
or enclosure, channel controllable digital It's Graham in on fade need to maybe you need to ride gain Need to do drives, which give us around three hours mono audio per side and around 1/2 hours stereo per side,' he says. 'We also wanted to replace the automated play-outs done after hours. As most of our services are still mono, being able to put three hours of audio onto a disk, and then simply program a play-out sequence would work like a bomb.'

As the new devices were intended as tape machine replacements, their linear recording capabilities, but with full editing facilities, were important to Schmaman. He adds that as the analogue I-O is now an optional feature, there is an indication as to how radio will be shaping up for the future. 'Like all other Studer equipment, they're built to last. That is, like brick outhouses, he enthuses, and we have had really excellent service from Studer and the South African agent, 8th Avenue Sound.' This was an important factor in looking at a system.

In addition to tape machine duties, the MO units will be used for archiving purposes. Schmaman says that, as the devices have standard SCSI drives, they are essentially dedicated computers, making upgrades easy because they are software-based. All the machines can be networked and have EBU sync, word-clock and video sync.

The MO recorders are part of a project to update NBC's radio studios, some of which, Schmaman admits, are getting old. The typical set-up is three to four tape machines, one Tascam 122 Mk II cassette deck, two Revos (221 CD) players, two Technics SP10 Mk II turntables and Studer 169 or 269 consoles. A variety of other consoles are also used, including Soundtracs FMBS, while the Dalet equipped studios has RS (Radio Systems) desks. Dalet is installed in only two studios and is currently an isolated system; although this could change in the future.

'We would like to extend it to all the other services,' says Schmaman, 'but that all depends on money. A fully integrated system including our advertising department would, of course, be the ultimate. During the night, it is the only service running on all transmitters and our All Night Service runs off it fully automated.'

NBC's refurbishment is an on-going project. The next big job is Main Control; tenders are out at the moment and Phil Schmaman and facilities manager Lutz von Dewitz visited IBC to see what was on offer. The long and ever-developing history of NBC continues.

October 1998 Studio Sound
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Introducing the first digital multitrack production studio in 24 bit
RENOWNED CLASSICAL recording engineer Tony Faulkner describes this as an "ordinary day." Certainly, the weather outside London's Henry Wood Hall is typically British, but the recording session inside contains a few surprises. The first is the ability of Marc-André Hamelin to turn fellow pianist Leopold Godowsky's absurdly difficult studies on Chopin's etudes into phenomenal performances for release on the Hyperion label. The second is the equipment configuration, beyond a typically single pair of Neumann U87c mics. EAR valve preamps and DCS A-D converters sit three cascaded recording chains. One addresses a DAT machine for producer Andrew Keener to vet the recordings, another a Tascam DA-88 for standard CD release, and the third a prototype Sony DSD recording system. The setup is a variation on a theme that Faulkner has used on some 15 recordings over the last 12 months or so, most recently on a 4-CD set of Nicholas Medtner's piano sonatas which were also performed by Marc-André Hamelin.

We have a policy of giving our clients the best quality we can get," he says in explanation. "So we've been creeping up to higher resolution and higher sampling rates over the last ten years. We're using DSD so that we can archive off the 2.8MHz bitstream, so we've got it in the stores for whatever high-resolution carriers come along.

Part of this philosophy is evident in the use of "classic" equipment like the old Neumann mics set in a simple configuration, and part of it in the use of a Prism M110A 24-bit splitter to turn the DA-88 into a 24-bit machine. The replay chain also speaks of quality with its EAR valve power amplifiers and Quad Pro 65 electrostatic speakers. But most obvious is the Direct Stream Digital hard-disk workstation, the Advanced Intelligent Tape backup system and other attendant boxes—the latest high-quality digital recording initiative to appear from Sony's R&D labs.

For the uninhibited, DSD captures a 1-bit datastream at a 2.8224MHz sample rate, avoiding the cakemaking and interpolation stages that hinder the use of higher sampling rates in 16-bit, 44.1kHz CD release, and lays claim to a DC to 100kHz frequency response. In addition to its applications in archiving, replication and distribution, it is part of a larger scheme hatched by Sony and Philips to replace conventional CD with Super Audio CD, a dual-layer hybrid disc adding a DSD (Scarlet Book) layer to the familiar Red Book CD layer. If accepted, SACD will retain compatibility with 10bn CDs and 500-600 million CD players in existence, and offer a route forward for high-quality digital audio. Although still under development, both DSD and SACD are gathering significant support within the professional audio industry, with around ten record labels onboard and the first domestic SACD players scheduled for their Japanese launch in Spring 1999.

At present DSD's appeal is primarily to the classical music fraternity—partly because of classical music's tradition of purist pursuit of quality and partly due to limitations of the prototype equipment; although Faulkner is expecting to have a DSD-capable SADIE system within a few weeks. And with surround sound so clearly on the classical agenda, it is significant that the first multichannel DSD recorder will be introduced at the San Francisco AES. In the meantime, Faulkner has been enjoying the use of one of the eight prototype recording systems—only three of which are in Europe—assisting in its testing and contributing to the fledgling DSD catalogue. He terms it "future proofing."

We did some of the first 96kHz recordings for Hyperion, but we didn't go with that because it seemed to me that going from 44.1kHz, 16-bit to 44.1kHz, 24-bit made more of a difference than going from 44kHz, 16-bit to 96kHz, 16-bit. Faulkner says of the race to raise recording standards, "and the complications in editing and sample-rate conversion just didn't make 96kHz viable. To my ears, sample-rate conversion degrades the sound to a lower standard than if you'd recorded at 44.1kHz to begin with. We live in a world where 44.1kHz is a very important delivery format and the DSD solution means you can downsample without any unpleasant surprises.

One of the key elements determining the sampling rate for DSD was conversion to all likely release rates with minimal degradation. For 16-bit, >
<44.1kHz CD, this involves the use of a 32.699-tap single-stage FIR filter and noise shaper in a process Sony has dubbed Super Bit Mapping Direct. To date, two downsampled DSD recordings have been made by PolyGram at Wisselord Studios in The Netherlands whose engineers regarded it as superior to their parallel analogue and PCM recordings.

We did listening tests in February in Scotland with the Scottish Chamber Orchestra where we ran 176kHz, 96kHz, 44.1kHz and DSD. Faulkner offers, 'We had a whole day with the orchestra rehearsing so the producer and I had the mics up and we flipped from one to another. It was very interesting—there appeared to be definite "thumbprints" on each of the sounds.

The main things you could spot were that the high-frequency reverber time appeared to be a lot different when you went from one sample rate to another. If you listened to 44.1kHz, 16-bit it sounded like a smaller hall with padding down the back as compared to 176kHz or DSD which rang on for at least another half second or so.

'I don't think it's the extra bandwidth you're hearing, it's to do with the filters and the amount of information—it's as if you've got more pixels. I'd quite like to do some listening tests just with the filters so that I could be sure just what it is you're listening to. You could create an analogue filter with the same characteristics and see if they give the same effect on the sound.'

Faulkner expects today's recording to see its initial release from the 24-bit DA-88 tapes if the provisional release date is to be met, but believes that the vagaries of record company schedules could see it become one of the first DSD releases.

'To date none of our projects have been completely edited from cradle to grave on DSD. It's all archived off. We tend to do most projects on 176kHz, DSD or both. The advantage of 176kHz is that you can use ordinary PCM equipment—although it eats a lot of channels and bandwidth—and you can edit it using existing technology. And if you want to surround you just lock more machines up where with DSD it's a bit more awkward while we're waiting for the gear to be fine-tuned.'

Faulkner evidently expects the wait to be worthwhile since, on top of the innate increase in quality offered by DSD, he sees the restoration of the 'professional headroom' that CD took away.

'That he is relieved to have a higher-resolution system than the domestic standard is unsurprising, but his enthusiasm for its editing possibilities may surprise some classical purists.

'If some postproduction is needed then we've got more resolution to play with,' he agrees. 'As it happens we've never EQ'd or put echo on to anything of Marc's, but some artists like that; when we did Kissin's Moonlight CD last year he decided he wanted quite a lot of echo put on it and we were able to do it without losing the quality. If you start off with 16 bits at 44.1kHz and the producer says, "I want it compressed, then I want you to roll some bass off and put some echo on", by the time you've gone through all that digitally you're probably down to a 12-bit recording and it sounds fairly crummy. If you start off with the highest practicable resolution, any meddling you have to do will have less effect on the transparency of the finished recording.'

'Half of this recording will be done in a different venue—it will be recorded in Bristol in October. If, when it's all put together, somebody says it sounds great, but the stuff in Henry Wood Hall sounds a bit more boomy, I wonder anything we can do? If taking half out at 60Hz and starting out with a high-resolution recording means that it won't collapse into a 10-bit recording then we can maintain the quality.'

With certain commentators targeting the limitations of the implementation of 16-bit, 44.1kHz digital audio over the value of higher-definition standards, it is interesting to note the comments of one of classical music's most successful practitioners. Having been closely involved with certain of the manufacturers seeking to maximise the performance of the existing CD standard, Faulkner is ready to welcome the completion of the SACD chain.

'The one thing that really interests me about SACD is that, when we do sessions for higher resolution than 44.1kHz, 16-bit and we go to all the trouble of editing it, what goes to the CD factory has to go through at least one generation of 16-bit,' he laments. 'I always wanted to be able to take the high-resolution layer into the control room and cut directly from that, and the SACD package for the Red Book layer includes that very option so when they cut the Red Book layer they cut it from the high-resolution source material.'

Faulkner is further concerned over the mastering media currently in use. 'Digital tape is a lot less noisy than we're led to believe by some people. We've tried recording DSD using tape, and the amount of error concealment that goes on in an 24-track, ADAT, DA-88 or DAT is far higher than you'd think. And DSD is obviously very intolerant of that—either the data's right or it isn't, and as professionals we should be glad of that. That's one of the advantages to me of moving on to this new carrier—just getting away from error concealment will give a greater degree of listening pleasure.'

As Andrew Keener directs Marc-André Hamelin's prodigious performance of a left-hand-only étude, and Tony Faulkner marks successive takes on his DSD workstation, it is easy to understand why they should be eager to see the back of the 16-bit, 44.1kHz CD standard. And it is difficult to see why they should not get it. Whether or not DSD and SACD are the answers to their needs remains to be seen, but with the weight of Sony and Philips behind them and the complicity of such talents, you would be ill advised to bet against them.

76

October 1998 Studio Sound

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There are many companies in the market place offering multi-channel audio PCI cards, but for most, this is where it ends. Most companies can at best just 'put chips on boards', but from Soundscape Digital Technology there is an unbeatable combination of software and hardware... mixtreme.

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- Wordclock/Superclock in/out
UNTIL NOW I have avoided delving into the specific technology employed to achieve IT-style networking because I do not believe it is relevant or useful to professional audio. While there is a temptation for manufacturers to promote the relative merits of various technologies for connectivity, the important questions remain: what problems am I trying to solve; is networking going to solve them; what about likely future requirements; and what effect will my chosen solution have on the bottom line of my balance sheet? Yet I can no longer avoid looking into some of this technology since they all make great play of the particular networking architectures they employ.

Digidesign supports several different networking options based on Ethernet and Fibre Channel Technology, and Ethernet networks have been used quite extensively with Pro Tools systems. At first, Ethernet backbones were simply used as high-speed data conduits in order to copy audio from one Pro Tools station to another. With the release of Pro Tools v4.0, auditioning files across an AppleTalk network became possible with Pro Tools 24 and Pro Tools III systems. Network auditioning allows you to preview one or two channels of audio before importing a file directly into Pro Tools. In addition to Pro Tools' own audition and import functions, Gallery Software, a Digidesign Developer, offers SampleSearch, which allows you to search extensive storage devices, such as CD-ROM effects libraries, audition selected files, and import those files directly into a Pro Tools session.

Digidesign believes the future of digital audio networking lies in Fibre Channel solutions. Fibre Channel, or FCAL (Fibre Channel Arbitrated Loop) allows multiple Pro Tools clients to access the same audio files and/or session files simultaneously across a network, centralised Fibre Channel storage, or Shared Storage Network (SSN) as it is sometimes called, eliminates the need to redigitise copies of audio for multiple Pro Tools users working in parallel on a project. Digidesign claims the benefits of this are immediate, especially for those companies that have short, mission-critical production cycles such as news, advertising and promotions. Fibre Channel also allows Pro Tools workstations and Media Composer systems to centralise audio file storage and use a single copy of a project on one storage subsystem, rather than multiple copies on local hard drives. Fibre Channel drives appear to a Pro Tools workstation as standard SCSI devices. Digidesign is currently testing the Avid MediaShare FC system with Pro Tools 24. In addition to Avid's storage division offering, third-party Fibre Channel solutions are also selling into the Pro Tools community.

Fibre Channel offers several more advantages over local storage: administrative security, fast throughput, and hot swappable connections. File security allows an administrator to set the privileges for each individual user. As the user logs on, the administrator can control which volumes the individual user sees and the level of access (read-write privileges). In addition to access control some Fibre Channel solutions offer built in Project Management features as well as bandwidth limiters on larger networks. As previously noted, Fibre Channel's 1.3Gbps/sec throughput capabilities allow Fibre Channel volumes to be used as on-line record-playback drives for Pro Tools 24 systems providing from 1–64 tracks of recording and playback. Furthermore, hot swappable connections allow drives or users to be added without bringing down a network.

Fibre Channel is still in the introductory stages, but, according to Digidesign, offers the promise of Fibre Channel switches (fabric switching). FCAL Bridges that will allow you to use existing storage media in a Fibre Channel environment, and the capability to bridge across different OS file architectures. Fibre-Channel also overcomes a Pro Tools limitation—in order for Pro Tools to use network volumes or centralised storage as a record-playback device, the volume must appear as a block level device when mounted on the Pro Tools workstation. Standard AppleTalk or Novell NFS volumes do not appear as block level devices. This means DAE (Digidesign Audio Engine) is unable to arrange the data in the proper order for Pro Tools and subsequently prevents users from even attempting to use the volume as a playback device.

Merging Technologies has introduced a way of sharing material between their Pyramax workstations known as AudioShare. The approach is a server-less one, that the company believes eliminates the biggest single expense and single biggest risk. The IT industry has always had two types of networking: networking that connects machines together and networking that connects devices (storage) to a single computer. The inter-machine networks (such as Ethernet) are suitable for a large number of users over a good distance (500 or more feet); and at good (for typical 'office' requirements) transfer rates (3Mbps effective). The device connections, on the other hand, provide very fast transfer rates (greater than 20MB s), but over very short distances (six feet or so), and with only one computer attached. Another important distinction is that inter-computer networks made an assumption the media would often introduce errors and thus the protocol had to be robust to compensate which introduces processing and bandwidth overhead. Disk networks assume the media to be virtually error-free and can exploit a significantly leaner protocol, further increasing their speed.

In the past year or so, a new disk-connection technology, Fibre Channel, has been introduced that offers increased distances (10km in glass), and high-data rates of 100MB s, and lightweight protocols. These components have finally become available as commodity items, and thus the price }
In a regular server-based system, regardless of how many clients are connected, the entire disk I-O funnels through the single server and the single operating system. Thus, there are relatively few problems managing access to storage. In a shared-storage system, the control of data flow from disk becomes critical, errors will result in the corruption of data. The fundamental reason is that each client host sees the entire network of storage as local, and thus opts to cache data freely. To the Data Processing (IT) community the complexity of solving this problem does not outweigh the benefits. However, for the audio-video industry, there is no suitable technology available.

What is needed is a layer of software that synchronises the access to storage between all the hosts involved. Given the existence of such software, it is possible to have very high-bandwidth access to a shared pool of storage, without having a ‘master’ server. Merging Technologies claims to have addressed these issues with AudioShare. It is a disk-sharing technology that allows several operating systems to work simultaneously using the AudioShare. It is claimed to be less expensive than removable disks or other technologies. It enables disk sharing by making available, to each user, every storage device or devices selected on a per-workstation basis. Scaling is achieved by adding bandwidth to do so in a disk-sharing topology is generally a linear cost increment. The server-type approaches on the other hand can be extremely costly to scale. The use of a switch or multiple switches can increase the bandwidth or number of devices even further (up to 10 million devices). With a switch, the bandwidth is not shared and every I/O port of the switch has the full 10Gbit/s bandwidth.

There are two typical modifications added when scaling is needed: the use of hardware-based RAID-5 storage and the deployment of Fibre Channel switches. Using external hardware RAID allows a set of drives to produce a steady stream of over 70Gbit/s of data. The host also handles the load on the workstation CPU's and offers the added protection of parity storage should one drive die, the others can recreate the missing data on the fly; thus, no real data loss is suffered. The cost of an external hardware RAID system is not really an issue; once an application requires a couple hundred Gigabytes of storage or more, the cost of the RAID controller is quickly amortised over the cost of the drives and the enclosures.

The next step to add power to a scalable system is through a switch. A Fibre Channel switch allows any and all pairs of connections to be simultaneously active as opposed to a hub where only a single conversation can take place at a time. A configuration that has three workstations and three data sources (three drive towers) could potentially have 300MB/s of real performance. Certainly, separate drive towers can be soft-switched together to get even higher performance than a single drive system (1.4Gbit/s and up). A single workstation will not need 1.4Gbit/s of data, but the sum total of workstations accessing that material will demand such performance.

It is difficult to state specifically at which point a given approach would not work, and thus when to switch to a different topology. One site of 12 users may work perfectly well with the use of Fibre Channel hubs and software-striped drives. Another site of eight users might need to use a switch instead of a hub. All of this depends largely on the expected number of simultaneous accesses. Facilities of different sizes and operational situations will benefit differently. Every facility will immediately derive some value from shared storage, and that value will grow rapidly as the facility learns how to exploit the technology and as the market develops new tools for it.

Studio Audio & Video's NASCIA network is claimed to be completely transparent to the users of multiple connected SADiEs. Further, Studio Audio claims no knowledge of the IT side of networking is required to gain the benefits of the technology—a SADIE user can use a networked system as easily as a stand-alone unit with no added emphasis on hardware or software.
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Due to the scalable architecture employed with the SASCiA network, there is no theoretical limit to the number of SADiE users that can be on-line at any one time.

becomes the critical factor. If 30 users tried to access the same audio data from a single drive simultaneously, a single drive would be incapable of delivering the audio. In large installations the network may well require RAID drive arrays to support it and some clever administrative network management to ensure the bandwidth is not exceeded.

ATM is compatible with most of the telephone service providers which raises the possibility of opening up wide area networks using SASCiA.

These particular manufacturers are not alone among the PC-Mac based DAWs in offering networking solutions, they were just the most forthcoming with information. That said, what characterises their proposals and differentiates them somewhat from the majority of ‘hardware’ DAW manufacturers is this: the emphasis placed on the particular technologies and topologies employed. I can only repeat my warnings about the seduction of elegant technology, I admit to having been guilty in the past and even now I am having great difficulty restraining myself from commenting on the side of a particular technology.

The important elements of all these solutions are the ones which take up the least space in the manufacturers’ proposals—the real world capabilities and management and security strategies. This is a little ironic as the basic networking hardware and software is available ‘off the shelf’ and is not proprietary to the DAW maker. There is, in fact, nothing stopping anybody ‘rolling their own’ networking solution for a host of the PC-Mac based DAWs, and many people do just that with results that vary from solid and useful to flaky and dangerous.

The clever bit is the design of software which integrates the networking into the whole philosophy of the particular DAW and from there to the production process. Not forgetting all the security and administration issues. If you are seriously considering a networking solution to an existing problem it is these areas you need to concentrate on. It is going to be unnecessarily difficult for specifiers to arrive at the right decisions for their particular needs until all the manufacturers can present networking in the same way as other audio and video products—relating what they are offering to real world processes or suggesting new ones. The facility manager should be able to concentrate on the design and integration of such processes to improve quality, efficiency, convenience and perceived value. It would be more helpful if the emphasis moved away from the underlying technology to avoid ‘blinding the users with science’.

Networking is, potentially, the most significant development to arrive in the audio and video market for many years. Unfortunately, like the Internet it is shrouded in mystery, hype and sheer hull. The real promise of both still exists more ‘in potential’ than in reality. As some early adopters have discovered it is possible to waste a vast amount of time and money for little real benefit.
Down by the Bitstream

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Sun, sea, surf, radio. **Kevin Hilton** reports on BBC Radio 1's largest ever outside broadcast at multiple venues throughout the sensational, party island of Ibiza.

Ibiza is one of the most widely recognised names in modern cultural society. It conjures up images of hedonism, excess and all-night partying, bound together by the leading DJs currently working on the club scene. It is a focal point for all serious clubbers, particularly those from the UK, and, as it to underline this, BBC Radio 1FM broadcast much of its live output from the White Island, as it is known, over the cusp weekend between July and August. It was the biggest outside broadcast the station had undertaken at multiple venues so far away from home, and ensured Ibiza maintained a high profile over the summer season.

This profile went through the roof when, at the end of August, the British consul on the island, Michael Birkett, resigned his post, saying he was ashamed of his country's tourists. The newspapers in the UK leapt on the story, but it has to be remembered that Ibiza has had numerous incarnations during its history and currently there are two very distinct sides to the White Island's nightlife. Situated off the western coast of Spain, Ibiza is one of the autonomous Balearic Islands—with Minorca, Menorca, Cabrera and Formentera—and was originally a Moorish kingdom.

Under General Franco's benign leadership, the island was forcibly developed: concrete boxes masquerading as hotels were built, together with airports, and the package holiday boom began. Of the 1.3 million people who visited Ibiza and Formentera in 1986, around half were British. This is one side of the modern Ibiza: drunken tourists wandering around the West End of San Antonio, a former fishing village, shouting, vomiting and annoying the locals.

The other side was created in the eastern districts of San Antonioduring 1987 with the arrival of DJs Paul Oakenfold, Danny Rampling and Nicky Holloway, who helped pioneer the new dance boom, attracting in excess of one million people—again half of whom are British—to come and dance all night, drink water and take E.

The Ibiza club season now runs from the end of June to mid-August and the music has entered the mainstream, with early pioneer Danny Rampling a key part of R1's new line-up. The station made its first trip to the island two years ago, when it broadcast for a total of ten hours over 2½ days, a schedule it repeated in 1997. This year it was extended to a full 3-day weekend, cramming in 4½ hours of live broadcasts from four venues: three of Ibiza's leading clubs, and a specially constructed studio in a villa overlooking San Antonio.

But first the BBC Resources OB team had to get there. While clubbers, DJs and musicians can travel light and fly into Ibiza, heavy radio equipment has to go by road.

A convoy of three vehicles—a B-Type OB truck, with a 40-channel Calrec GP desk, pulling a trailer, a mobile VSAT unit mounted on a Peugeot van and a Peugeot Boxer full of equipment—left London on Saturday 25th July, driving down through France, reaching Barcelona on the Monday to catch the ferry for Ibiza on Tuesday.

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**Kevin Greening presenting The Breakfast Show from the terrace of a villa in a secret location.**

**Using a Denon DN961 FA CD player and a Denon DN981 Minidisc player**

Despite the high-tech holiday image of the place, some elements of the Ibiza way of life are still stuck in the pre-Franco redevelopment days. The crossing was made on what Richard Earle, the sound supervisor who co-ordinated the OB, working with executive producer Pat Connor, describes as 'a crate'. To while away the 9½-hour journey, Earle and his colleagues were entertained by videos of *Back to the Future* (in Spanish) and a poorly tracking copy of *Jurassic Park.*
This translated itself into other, more immediate problems once the BBC team arrived on the White Island and made their way to the villa. The biggest problem on Ibiza is the poor conditions of the roads, explains Earle: 'Once you're off the main routes, you're faced by cart tracks and our driver, Paul McCann, had problems finding a place to park the B-Type.' On the Wednesday, technical shortcomings came to the fore, although VSAT was the preferred method of getting signals back to London. ISDN was necessary as a backup, and for any transmissions that were not using the satellite.

We cruised round the venues, checking that everything was okay,' says Earle, and although we had one ISDN 2 connection at the villa studio, one at Cafe Mambo and two at Bar M (which had been ordered independently by the owner), we didn't have what we had asked the provider. Manumission, for at Privilege—Spanish Telecom had run out of ISDN on the island. With live broadcasts set to start with Jo Whiley's Friday midday show, the remainder of Wednesday was spent rigging Bar M, with the villa studio constructed on the Thursday. In temperatures of between 35°C and 36°C, the crew unloaded 2 1/2 tonnes of equipment to create a continuity suite, DJ and performance areas and an edit room. 'I think that each of us drank two or three litres of water,' laughs Earle, 'Everything takes so much longer in those conditions, but we fixed the VSAT link for the villa and put everything else in place. We had an ISDN connection there, but we wanted to save money on international calls by using VSAT, which was on our own link.'

This villa is often rented out as two buildings, as one section built on top of the other. The upper portion includes the terrace and swimming pool, while a customised 16-channel submixer of the main DJ desk was built. Like the majority of BBC OBs gear, the consoles were Glen Sound, in this case augmented by an SPX 1000, a rack of BSS compressors, foldback, and some PA gear. This played host to Linda Hicks on the Saturday, and the Lighthouse Family on the Sunday. Also on Saturday, Joceylin Brown sang live to backing tracks at Cafe Mambo during Danny Rampling's show. The pool area provided an obvious backdrop to Zoe Ball and Kevin Greening's Breakfast Show, with lots of wild and wacky stuff of people being thrown into the water.

The main continuity studio, run by sound supervisor Stuart Veasey, with Kevin Long, was built in the lounge of the lower part of the villa, with the main DJ desk out on the covered balcony, overlooking what Earle remembers to be a 'fabulous view of San Antonio.' The continuity suite housed two S60 Systems Short-Cut editors, both with external MQ drives on a SGIII connection. These 510MB devices give 20 minutes of storage and enabled material to be moved around, not only between these two units, but also a third Short-Cut in an edit suite, set up in one of the bedrooms.

This room was further equipped with a SADiE hard disk recorder-editor, that offered V.3 software, but Earle >
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comments that most of the crew using it prefer V.2 MiniDisc and DAT machines were also installed, with microphones set up in the second bedroom for any necessary voice work. RI's current affairs programme, Newsbeat, and magazine show The Net took feeds from this facility, receiving material either by VSAT or ISDN.

All ISDN connections were through Telos Zephyr codecs using stereo Layer 3 MPEG compression. "This is perfectly adequate for RI," observes Earle, "as we can get away with 128k." ISDN digital 'phones were also used, with analogue converters, enabling the B-channel of an ISDN line to become an ordinary telephone connection, used for both conventional telephony and Internet communication for laptop computers. Aside from its backup function, ISDN was primarily used to receive cues from London during programme junctions, as the VSAT operates with a 1.5s delay.

All day-time programmes came from the villa, which was kept as a secret location. After 4pm, the emphasis was on location broadcasts from around the island at the three featured clubs: Cafe Mambo, Privilege (which many still call by its former name of Ku) and Bar M.

At the last venue, splits were taken from the existing club mixer for transmission, while the desk itself was augmented with 'broadcast bolt-ons': CD, MD and a DJ mixer. In the evenings, Privilege-Ku becomes Manumission, named after the promotion company that organises club nights there. There are two rooms, one that can hold 6,000 people and features a swimming pool, the other a smaller space for chilling out, where Annie Nightingale hosted her ambient trip-hop show.

Earle and senior sound supervisor Steve Richards split the main desk here and installed MD players for jingles and a microphone, as there are usually no speech interruptions during a club DJ's set. A control room was built in what is normally the T-shirt franchise shop, enabling RI IDs and any announcements to be mixed in without them being heard in the club itself.

At the smallest of the venues, Bar M, the DJ booth is not visible to the crowd, so it was moved to a balcony so that the DJs could get a good look at their audience and the audience could debate whether they, like most DJs, have good faces for radio or not. Moving the equipment outside was a good plan while everything was Mediterranean, but it was just before Judge Jules Dance Anthems show on Sunday afternoon that the weather, which had been perfect up till then, finally broke. The skies darkened, leading many among the BBC crew to believe that the end of the world was nigh. As rain and gales lashed the White Island, the technicians only had two hours in which to rig and move the DJ desk back inside.

This was one of only four glitches during the whole weekend, two of which were technical, the other one artist-related. The first break in transmission came on Friday at the villa when the VSAT overheated and momentarily shut down. There was a slight gap but everything was switched over to ISDN in ten seconds, re-establishing the connection with London. At Cafe Mambo, the club's lights went down, which tripped out the breakers in the broadcast racks for around 20 seconds. The dead air was covered by the 'shadow DJ' back in London, who put on a filler track, from which the DJ in Ibiza just picked up from at the end as though nothing had happened.

The only other emergency was when Lisa L’Anson did not make it for her show, with Emma B having to step in.

After the rigours of the weekend — A bit more sleep would have been good. Earle jokes — the Resources crew had the prospect of the ferry journey back to the mainland and then the drive up through France to London. On the journey back, they reflected that things had gone well, considering the number of hours they had broadcast. And the farewell meeting with the mayor and deputy mayor of San Antonio had gone well too. "We had been worried about that but they were pleased," concludes Richard Earle. Something that, if he had known about it, might have convinced Michael Birkett to stay on at his post as British consul.

Continuity from the secret villa. Stewart Veasey, sound supervisor with Glen Sound MX6C Series equipment, Telos Zephyr (on top of the rack in the corner), video O/D of VSAT receiver showing Spectrum 12GHz SAT Rx and Shortcut editor
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A FEW YEARS AGO the ability to dial-up a high-quality sound circuit as easily as a telephone would have seemed incredible but it is now commonplace in broadcasting, and other areas of the audio industry are also finding further applications for this facility. This series looks at the telecommunications and audio developments that made it possible: the uses our industry is making of it and what users need to look for in products on the market.

Until about 1991, the opportunities for sending high-quality audio over distances were meager. A 15kHz international analogue circuit could sometimes be booked in advance with BT in the UK, but this required a good deal of notice, was expensive, and was only available between sites already equipped with permanent wide-band local links from a telephone exchange. Nationally, wide-bandwidth circuits were used by broadcasters to link their own studio centres and transmitters on a permanent basis, and frequently used outside broadcast sites had them. To bring in speech contributions from abroad the crackly old phone line was the principle means. The basic problem was that it was noisy radio that wanted to send high-quality audio over distances, and it was not worth the world's telecommunications providers putting facilities in place for such a small amount of traffic.

From the middle eighties the telecommunications infrastructure went digital. Thus between exchanges and switching centres all traffic went over high-capacity bearer circuits known by the then Post Office as the Integrated Digital Network (IDN). At that stage it became possible for customers to lease a permanent 64kbps per second (kb/s) circuit known as a Kilostream between two sites. But this was not of much immediate use for sound transmission as digital audio then required over 1 Megabits per second of bandwidth using the linear coding techniques in use. Back in 1962 the first standards for digital telephony were set and decided on the use of 64kb/s for the basic channel because at that time, using linear coding, this provided the 5.4 kHz that is adequate for telephony calls — this standard being known as ITU-T G 711.

Then, in 1987 the French firm AETA released a coder-decoder unit that allowed 2-way 7kHz transmission on a 64kb/s circuit and they called it an audio codec. The 2 to 1 data-reduction system used is known as G 722, and is an international standard that had been developed in the US to permit high-quality phone calls for such applications as audio conferencing. The BBC immediately tried out connecting a pair of coders at each end of a chain of stations between London and Sydney. It worked, and the World Service could then be fed to stations in Australia at a reasonable cost. And in the reverse direction news reports could be filed to London. It is a requirement of telecommunications, as opposed to broadcasting facilities that they are fully 2-way (duplex).

Permanent point-to-point digital circuits have some uses in the industry, but what was really wanted was a telecommunications service that could offer on-demand digital connections. This would support link-ups as and when required between any sites. It emerged that something known as the Integrated Services Digital Network (ISDN) was planned to do exactly this, but there was a lot of scepticism in the telecommunications industry that many customers would use such a service, and this led to a number of alternative names such as 'Innovations Subscribers Don't Need'. Fortunately these voices were not given credence, in 1990 some trial ISDN lines were installed in the UK, and broadcasters were among the first to try them out with the couple of makes of G 722 codec that were on the market.

So the marriage of data-reduction audio codecs and telecommunications services such as ISDN has produced the ability to dial-up audio circuits that can convey good stereo sound. But, as with all things digital, there are complex matters that have to be addressed, and systems and standards can be involved. ISDN is marketed widely and is the fastest growing telecommunications service in the UK, for example. It is fortunate that we can now get high-quality sound transmission using a general-purpose telecommunications service that is readily available. Put simplistically, ISDN can be regarded as being a conventional telephone system, but without the usual digital-to-analogue conversion that goes on at the telephone exchange for standard analogue phone lines. In this way it provides end-to-end digital calls between two ISDN subscribers, but can also support calls to standard analogue telephone lines. However, ISDN is more involved and a lot of work was required to design a sophisticated service that would yield the utmost from the old-fashioned copper cable that gives a customers premises access to the telecommunications network.

ISDN is available in two varieties — Basic Rate Access (BRA), and Primary Rate Access (PRA). The majority of use is made of BRA (known by BT as ISDN2) which is delivered on the standard pair of copper wires that is normally used for analogue telephony and it supports two channels of communication at 64kb/s each. These may be dialled-up separately as calls to different places or, if required, both to the same one and the bandwidth combined to produce 128kb/s. The 64kb/s data streams are referred to as B channels, and in addition there is another channel of >
call, and ways of ensuring a call is answered by a particular device on the line when it arrives (fax machine, PC, video-conferencing equipment or audio codec). When ordering a line the telecoms provider will ask the customer about various features and supplementary services that may be required, but, in the main, the equipment used in our applications does not require them.

The equipment a customer uses to communicate with an ISDN line (the equivalent of the telephone instrument on an analogue line) is known as a Terminal Adapter (TA), and these days they are generally built into audio codec equipment. One complication is that the protocols used by the D channel to communicate with the telecoms network vary slightly around the world. So when taking a codec abroad you need to first check that your equipment will work there. Britain in line with most of the world uses a standard called Euro-ISDN, but North America, Japan, and Australia have several systems of their own. However, you can now often request Euro-ISDN to be provided in Japan. Furthermore, the TAs in many codecs permit switching between national network protocols.

In a studio centre where many ISDN lines are needed, it may be practical and economic to use Primary Rate Access (or ISDN 30) which, as its name suggests, supports up to 30 B channels and is delivered over a single fibre optic or microwave link. Thirty is the maximum number of channels supported on such a 2Mbps line but if you need less than 30 the customer can usually specify the actual quantity required and only rent this number, thus reducing the cost. The minimum normally being six channels.

An audio codec cannot be plugged into a Primary Rate line, but there are several different ways to work with PRA.

The first is to use a small ISDN telephone exchange that provides Basic Rate Access at each extension and this can also be used to derive ordinary analogue telephone lines. The alternative is to use a special Primary Rate TA that provides variable bandwidth from 64kb/s to 1920kb/s by setting up multiple calls—a technique known as 1-Muxing.

So the basic requirements to dial up high-quality sound are to connect together an ISDN line, a Terminal Adapter and an Audio Codec. If the TA had not been built into the codec a data connection would have been required between them. There are various standards for data connections and these are usually provided by way of familiar D connectors you see at the back of computers as well as a few other multipin types. The principle data connection standards used with codecs are V.35 in North America and X.21 elsewhere. In most situations it is not necessary to know anything about these except where required to purchase the appropriate cable. Even when codecs come ready equipped with TAs they may also have a data connector at the back to allow for situations where the internal TA cannot be used. For instance when in a country using a D channel protocol not supported by your TA, in which case a locally procured stand-alone TA may be needed. It may be that ISDN is not available and some other type of digital line is provided. In all these cases it will be necessary to have an appropriate data cable.

ISDN is now operating in 55 countries around the world and is becoming commonplace. For instance, in Germany ISDN has become the normal way to provide premises with more than one telephone line. The use of ISDN to access the Internet has brought down its cost and increased its availability, but...
in some countries it is still only available in big cities. To be truly independent and transmit high-quality audio from literally anywhere on Earth there is the option to buy or rent a digital-satellite telephone that permits data calls and plug this into the data connector of the codec. The Inmarsat system allows calls to be made which then connect on to ISDN lines, and radio news broadcasters use this facility a great deal. This portability comes at a cost that is per minute call charge of about £6, but the freedom it brings and not having to arrange lines can make it worthwhile.

Sending a fax is easy, and you know you can dial up and be compatible with any other fax machine in the world. So it would have been marvellous if audio codec technology had provided the facility for any studio to dial up any other studio's number listed in an 'ISDN directory' and send high-quality stereo audio to each other. Unfortunately, there is no such universal compatibility due to there being several standards for data-reduced audio and a few other technical parameters. So you have to check out the other end first of all so you know something about the equipment they have and how it is configured.

The term 'compression' is often loosely used to describe the process used to reduce bandwidth, but this implies that like a sponge when the process is completed (and the sponge is no longer compressed) you return to having the original. But with sound you are left with a small proportion of the original that provides the human ear with the illusion of sounding like the original. A more accurate term is 'data reduction' and it is the one used here.

The first audio data reduction standard of G.722 is still commonly used for news and sports contributions, but in the field of music and for stereo, there are four principle systems.

MPEG is well known as a family of standards for audiovisual applications that was set by the Moving Picture Experts Group of the International Standards Organisation (ISO). It is the basis of digital TV and video over the Internet, but audio was the committee's first priority and MPEG1 was set in 1992 after trials hosted by Swedish Radio in Stockholm. Later augmented by MPEG2 which widened the range of data rates. MPEG audio comes in three types known as Layers. Layer 1 coding being intended for use with data rates over 48k/s, Layer 2 for 128k/s to 384k/s and Layer 3 below 128k/s. However, improvements in the technology have allowed each layer to now work with lower data rates than originally intended.

An ISDN line provides up to two 64k/s circuits so the most common requirement is for audio coding at 128k/s. This is achieved by both Layer 2 and Layer 3 coding. Layer 3 is somewhat more complex than layer 2 and thus suffers from a processing delay of about 0.25s that can be problematic in some applications such as live interviews. Layer 2's delay being about 0.1ms.

There are two proprietary coding schemes in widespread use, Dolby and apt. Dolby offers two types AC-2 and a later refinement AC-3 which like MPEG Layer 3 offers improved performance at lower data rates and gives a 15kHz audio bandwidth in mono at 64k/s. The aptX-100 scheme offers mono 15kHz bandwidth at 1.28k/s or stereo at 256k/s. The apt system is different from MPEG and Dolby in not being based on psychoacoustic data reduction principles which exploit the way it is possible to fool the human auditory system using certain tricks. For instance we cannot hear quiet sounds close in frequency to loud sounds so they may be removed and we do not perceive the difference. The apt system is based on Adaptive Differential Pulse Code Modulation (ADPCM) which is a purely engineering process. The significance of this difference in approach is in the contentious matter of signal degradation through cascading data-reduction processes. The experts disagree on this subject and practical trials seem to have produced differing results, but some claim that psychoacoustic methods are less resilient to repeated encoding.

By using multiple simultaneous ISDN calls it is possible to code at up to 384k/s (three ISDN2 lines) with the equipment on the market, and experts agree, the higher the data rate the better the quality on a single or multiple pass.

In the next article the more practical aspects of using this technology will be explored and the features to look out for in purchasing equipment.
Fritz Lang had lived to require
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The successful Munich-based
Geising-Team celebrated its
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Dan Daley visits an
all-Fairlight postproduction
facility grown on commercials
affluent denizens of the tree-lined
Rahlstedt neighbourhood.

Inside, in contrast to the exposed
ducts and concrete motif, a more famil-
ial atmosphere prevails: two or three
(no one seems quite sure how many
exactly) local, friendly mutts lay quietly
on the floor or cavort across the hard-
wood floor in play. It is not what you
might expect walking into a leading-
edge audio postproduction facility, but,
says co-owner and Chief Engineer
Thomas Froschmaier, it takes more
than it once did to stay competitive
these days.

The clients have become quite
sophisticated, he observes. So has Gei-
sing-Team, which has won a single-
spaced page's worth of national
and international commercial awards,
including a Gold at the International
Advertising Film Festival. Named after
the studio's previous location, in that
section of Munich before it moved to
its current site two years ago, the post-
production and sound design facility is
on its fifth Fairlight MFXplus worksta-
tion. Just adding its second system a few
years ago immediately underscored the
need to network the facility's audio sys-
tems, he says. a need that has grown as
the studio has. Froschmaier is awaiting
the impending release later this year of
Fairlight's MediaLink Windows-NT-
based central server system, that was
being beta-tested in two studios in
Tokyo and Los Angeles over the sum-
mer. In the meantime, he and his crew
of mixers and editors use wired access
to a central machine room (quite liter-
ally central: it stands like a blockhouse
in the centre of the facility's main floor)

October 1998 Studio Sound
which houses the Fairlight engines. Beta SP and digital Betacam video decks and a complex patch bay that allows routing of the large sound effects library the studio has built up, comprised of outside and in-house effects sources.

While there is networking between each studio for a client booking database, Geising-Team has a not-uncommon methodology for databasing audio: daily backups of each workstation and a once-a-week archiving sweep of the five Fairlight hard drives. When projects need retrieval for updating, they use what US facilities have come to call the SneakerNet: assistants hurrying between studios with files on drives for uploading. "We could always get 20Gb more hard drive space for each machine," Froeschmair remarks. "But I think we prefer to wait for the Medialink."

While individual mixer-client relationships are important, Froeschmair says they do not take on the cult status that common in Hollywood film post and New York commercial post facilities, where clients have been known to follow mixers from one house to another in mid-project. Rather, the 'team' in Geising-Team is the attraction, with the facility's five virtually identical audio suites—one of which, Studio A, is a large film mixing studio equipped for matrixed Dolby Stereo and Dolby Digital; audio suite Studio E is equipped with Dolby Surround)—allowing projects to move seamlessly between rooms, with the same acoustics and workstations in each one. Two Mackie 32-Bus consoles work with a Yamaha 02R desk, an Amek Big, and a heavily modified Soundcraft TS-24 (referring to the desk's various homemade EQ and bus mods, he jokingly calls it 'more of a Handcraft console') with PiMix modular monitor matrix, a carry-over from the previous facility.

"Some clients have a personal preference for certain mixers," he says. "But mostly they want to know that the project is proceeding steadily and properly. They don't need surprises—that's another reason we chose to have all the same workstations—we get the highest level of compatibility you can get that way." Monitoring is also common in all rooms—Genelec 1030 speakers, with JBL cinematic speakers in the film mixing studio—and it is not unusual for the same project to be undergoing multiple aspects, such as sound effects and dialogue editing, simultaneously. The Fairlight's inclusion of ORI file interchange also allows it to work with Mona Davis, a Digidesign Pro Tools-equipped commercial music production company that camps out in the building as Geising-Team's symbiotic tenant, developing music and effects which help attract new business to the post facility and which provides Geising-Team with another service to offer its clients. Says Froeschmair: "We always have additional producers or directors and others in our productions, who help the clients to realise their creative ideas, while the sound engineer is able to concentrate on technical things. Those producers are an important part of the team and the studio-client relationship. Froeschmair is quite content working in a 16-bit environment, which he says is more than sufficient for television and film commercial work. "You still have a 96dB dynamic range and you're only using 10dB of it in commercials half the time," he notes. "I don't see us switching to 20 bits when it becomes available. Though perhaps, for cinema commercials it might be something that producers will want.

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remote work. Using its apt system, Froschmaier says that increasing amounts of the studio's work are done in conjunction with other German post houses, sending celebrity voice-overs back and forth for regional spot work. There is, he adds, enough work to go around in the country that competition, while present, is more collegial than cut-throat, a reflection of how the expansion of European broadcast and cable networks has spawned a boom in post-production to meet its growing content demands. Froschmaier, who started the business 10 years ago with his partner, producer Olaf Mierau, adds that when the studio changed location two years ago, it was to accommodate what was then a 100% increase in business, a trend that has not stopped as the facility has grown to five rooms. However, Germany is not the extent of the company's ambitions: Froschmaier says they plan to found US division, GT Sound-Munich, as a joint venture together with Los Angeles company, Alexander Büthenheim, MAID Music, sometime around the end of the year.

But Froschmaier still believes that there is far more headroom in German audio post still to come, particularly from a new generation of independent filmmakers, some of whom Geising-Team works with in conjunction with a nearby school of visual arts. Film is becoming more of a factor in the German entertainment industry, particularly independent films done by young German directors, he observes. 'Combined with how much commercial work is now being done for cinema, I expect that there's a lot more growing to do for us in the future.'

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US: Hot 'Lanta

Atlanta, often overlooked as a US musical stronghold, could soon lead the way due to a new style of A&R writes Dan Daley

They don't teach geography in American schools anymore, which is why most US kids couldn't find France on a map in a well-lit room. But there are some subtleties of geography that cause understandable mistakes. Last month I wrote about how Nashville is changing, and those not familiar with the South-east region often lump Nashville and Atlanta into the same category. To set the record straight, Atlanta is a city of 3½ million people (versus Nashville's half-million or so) located 250 miles south-east of Nashville and an entire universe away. The two share nothing but relative proximity—not culture, not heritage, not music; certainly not cuisine, of which Atlanta has much and Nashville little. But they do share opposite ends of a perceptual spectrum. While Nashville labours under the misconceptions associated with country music—lack of shoes, teeth and so on—the city is at least regarded as having a music business to go with its music. Atlanta, on the other hand, has long been characterised as a city full of music, and no music business to go with it. Let us put that misconception to rest, because as Nashville's hyper-inflated star painfully condenses back to realistic proportions, Atlanta's has the opportunity to rise to its true potential.

Atlanta has always been a music city. It might just have acted as the mother ship to more than the candle over the years. When the Muscle Shoals sound waxed in the sixties, it drew as much from Atlanta as from Nashville in terms of money and music business. One of the capitals of Black cosmopolitanism in the US, Atlanta retained an African-American, geographically core even as Blacks and Black music moved north to New York, Chicago and Detroit in the sixties and seventies. It is worth noting that while New York has been the pulse centre of Hip Hop, and LA pretty much took over Rap when that genre went Hollywood in the nineties, Atlanta never stopped being the de facto capital of R&B. Boulder Brown, Whitney Houston, Jermaine Jackson and others recall there regularly, making records that define contemporary R&B.

Atlanta was also the map identifier to the world for its southern neighbour Macon back in the seventies when Southern Rock ruled the charts and airwaves. Bands like the Allmans, 38 Special and Lynyrd Skynyrd used the region as their own stomping grounds, and rock continues to be represented in the area by acts like REM and Matchbox 20. In short, Atlanta has managed to hold onto a very diverse spectrum of American musical styles for decades. Its tree trimmed with a few glinty baubles from the world of Pop, most notably Sir Reginald Dwight, who owns a football team in Manchester and lives in the Buckhead trendy neighbourhood. Now, let's get down to business. The reigning canard is that Atlanta has music but no music business. Right? The industry remains hunkered down in New York and Los Angeles, with country music running its own business operations more or less autonomously in Nashville. This is true, but only if you overlook a fundamental change that the larger music industry has undergone in recent years; there are no more A&R departments, except in name. American major record labels rarely develop their own new talent anymore; instead, they use 'check-book A&R'; find a hot producer or DJ; remixer, give him a multimillion bucks and let him bring back four or five records for you to market and distribute. If this relationship proves successful, give him his own label and fund it. Voilà—instant A&R.

Accepting this as the new model of A&R, then Atlanta is ready to lead the pack. Over recent years, it has attracted a surprisingly deep and very successful collection of the very sort of producer-auteurs, including R&B hit-makers Jermaine Jackson and Lionel Richie, as well as rockmeisters like remixer Brendan O'Brien (Foo Fighters, Pearl Jam) and Matt Serletic (Matchbox 20, Collective Soul).

What does this bode for Atlanta's studio community? One of the by-products of Atlanta's development is that it now boasts what is likely the world's largest collection of upscale personal studios, many of which are in the high-priced gated communities of Atlanta's elite and which thus like to keep low profiles.

Europe: Digital duelling

DVD Audio has encouraged discussion writes Barry Fox

When Michael Gerzon Died in May 1996, his partner on many projects, Peter Craven, predicted: 'What Michael was doing now, the world will want in 10 years time.' Gerzon worked with Craven and Peter Felgett on Ambisonics. The system was mishandled by Britain's NRDQ quango, now the British Technology Group, and is now largely forgotten. But it will be rediscovered, as DVD Audio creates the storage space needed for surround with height.

Again with Peter Craven, Gerzon wrote the theory for noise shaping. His last work was for the voluntary industry group, the Acoustic Renaissance for Audio. Recent events ensure that his theories on lossless compression will steer the future of audio in the next century.

Shortly before Gerzon died, Bob Stewart of British hi-fi specialist Meridian, and the ARA, set up a small spin-off company with Gerzon and Craven. They had filed their first patent application on lossless packing a year earlier, so luckily all the seminal work had been done when Gerzon died. Stewart had already seen the potential value of their technology.

Earlier this year, after meetings at the Stereophile show in the USA Ambisonics agreed to licence the system and Stuart suggested Meridian Lossless Packing, or MLP, to Working Group 4 of the DVD Forum. WG-4 has been setting the standard for DVD Audio.

The record companies had specified 74 minutes and 6-channel surround at 90kHz sampling with 24-bit linear coding. The only way to achieve this compression, which the prediction process in MPEG video compression. There are still some errors, so the coder also sends a signal which identifies the difference between the predicted and actual waveforms. This can save at least 50% of the bits. The saving is greater when the sampling rate is higher because there is less useful data in the bit-stream for audio over 20 kHz. MLP extends DVD Audio playing time to 89 minutes for 6 channel surround, 235 minutes for 2 channel stereo, and 125 minutes at 192kHz.

MLP could also extend the capacity of a conventional CD, to store 4 channels or 24-bit code. But this is unlikely because it would break the CD Red Book standard. The real value of MLP is for new formats, like DVD-Audio, which are not yet on the market.

We have a desire to participate in litigation, says Slusser, 'however, it becomes officer,' Dan Slusser, wrote a letter to Biree Suzuki, of JVC, in his role as Chairman of the DVD Working Group 4.

The standard as adopted now is based on clearly improper and false assumptions, procedures and information', wrote Slusser. 'We have a provisionary law firm to review the procedures... and they have concluded that DTS has substantial legal claims, including anti-trust claims, with appropriate treble damages and injunctive relief which result from the actions of WG-4 in selecting only the Dolby/Meridian system as the mandatory standard.'

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'No', answers Slusser, 'because it is less likely that we would have no real choice.'
The DTS move has gone down like another lead balloon with WG-4 and DVD Forum. Bike Suzuki very quickly replied in reasoned, but very firm tones, copying the put-downs across the audio industry, from Universal (which backs DTS) to Panasonic (which has so far supported the system), and to the RIAA in New York, along with Koji Hase, of Toshiba, Chairman of the DVD Forum. Suzuki says that WG-4 has also taken its own legal advice and is confident that the Group's standardisation procedures have been perfectly proper and there are no anti-trust problems. The DVD audio specification will 'fully accommodate DTS, as well as competing technologies, as an optional feature'. Suzuki reminds DTS that it 'voiced strong complaints' to WG-4's work only after 'objective evaluations resulted in the selection of MLP over DTS'.

WG-4 chose MLP because it met the recording industry's demand for 74 minutes in all modes and 'objective and mathematical measurements clearly showed' that MLP coding and decoding gave an exact match with uncompressed audio and 'the biggest safety margins in the format and on playing time'. DTS had rallied support from three prominent engineers. But WG-4 was not impressed. Their concerns, says Suzuki pointedly, are based on 'a misunderstanding' which may have resulted from the 'sample letter your company sent them'.

Communication between Europe and the US is about language and technology. Kevin Hilton attempts to eavesdrop.

As EVERY ONE KNOWS somebody once said that Britain and America were two nations divided by a common language. Unfortunately, everybody thinks that this was someone other than George Bernard Shaw, who, in fact, it was. In technical terms of audio technology, this prescient observation could now be reworked as Europe and America being two states divided by a common technical language. Both are moving towards digital, but with differing standards, agendas and timelines, issues that, seemingly, are enforcing differences and a lack of understanding, reinforced by existing national prejudices.

For many Europeans, the US appears introspective and unwilling to acknowledge that anything else is happening beyond its boundaries (unless those events immediately affect Americans). All countries have a tendency for this—something that was sent up years ago on a UK satire show, with a newsreader delivering details of a bomb explosion in which no Britons were involved and listing the other nationalities in order of importance—but America alone has the freehold on it. It should be pointed out, unlike many countries, that I like America (or at least bits of it), so this is not merely Yank-hashing, but the view was confirmed for me when I returned to London from the IBC in Amsterdam.

As I was leaving the plane, I found myself walking behind two men who were obviously involved in the American broadcasting business and had been at the show, with a stop-over in the UK before returning to the US. They were glad to be heading back home and were disappointed with IBC, saying that there was nothing new there, and that it was purely a local show. Matters were not helped, they said, because of the language problem. Of all the places you can go in Europe, and complain about difficulties with the language, the Netherlands is not one of them. As English comedian Eddie Izzard said, in Amsterdam they speak four languages and they take drugs, whereas in Britain (and possibly America), they speak one language—just about—and don't take drugs (at least not officially).

Over the show I had walked into the press room and seen two journos I know—one French, the other Dutch—talking. When I joined them, the local hack bemoaned the fact that he was at an exhibition in his homeland, yet he was compelled to speak English. As I said to him, this was the cost of the event organised by a British-based organisation in another country. But for the IBC to be an international show, as the name implies, and as the organisers intend, then English probably is the safest common language.

What puzzles me further about the view of the IBC as a show is that IBC is patently not a local show: it is growing, and attracts all the major manufacturers from all around the world, although, for the audio community, it is becoming less and less a viable proposition. Sure, NAB is a major showcase—and it is held in the US—but companies, even American ones, choose to make significant announcements in Amsterdam. Not everyone makes it over to NAB, and so IBC is an obvious European, if not world, launch site.

If my fellow travellers felt that IBC is parochial, then what would they make of the ITC in Montreux? The Swiss-based show has been described as 'the last event before the IBC'. It is located in local hotels, which are often in short supply, as well as other logistical difficulties. Now the event is to go annual from next year, putting the emphasis on papers, keynote lectures and discussions. Naturally there will be the exhibition, the Digital Audio Technology Management's new director, Chris Zoehelli, explained, 'This provides an enormous opportunity to create an annual exhibition for high-end broadcast solutions where emerging technologies can be unveiled to the industry as a whole. Broadcasters need such forums, how many exhibitions they need is another issue. There has to be a chance to see what the whole market is doing and how they means getting on a plane to see people outside your state line are up to. Europe is certainly aware of what the US is doing. Just before IBC, Bill Cullen, chairman of London video post house M2, commented to me, 'It will be interesting to see how many people have a sticker saying HD ready.' They did not blow out the sun, but there were certainly enough around for them to be noticeable. Europe is still not convinced about high definition, but, as America moves towards it, there is the realisation that they should at least know what is going on. It is the same with AC-3 and MPEG Surround, the latter is meant to be the European standard, but final decisions have yet to be taken, which means that people have to know about the DOLBY contender in case, like DPL, it establishes itself by default. Regardless of any nationalistic feelings, America and Europe need to know what the other is up to. Perhaps we should try to develop a kind of technological Esperanto.
DSD: where are we now?

David Walstra of European Sony explores the potential impact of Direct Stream Digital and its associated music carrier format, Super Audio Compact Disc, on the audio business.

In the early 1980s, while working as an electronics engineer at Polygram's engineering department, I was involved with the introduction of the Compact Disc. During the recording sessions in the Concertgebouw in Amsterdam, we would have an early model PCM digital-audio signal processor recording the orchestral sessions in parallel to the standard analogue machines. The engineer with the PCM recorder was allowed to be there, as long as he did not make too much noise or sit in any silly way. His supporters were in the minority—not many people believed that CD had any future at all. Let alone PCM. That came much later.

In November 1997, I found myself in a very similar situation, this time in one of the control rooms of the Henry Wood Hall in London, recording a session with an experimental DSD recorder, in parallel to the now accepted PCM equipment. I tried hard not to sit in anyone's way, and I doubt anyone but the recording engineer Tony Faulkner thought the experimental kit made any sense. That is, not until we played back some of the DSD recording to the performing musicians.

Since those early eighties, Pulse Code Modulation or PCM-based equipment, has made a tremendous contribution to the recording industry. Recently, with bit rates increasing, significant sound quality improvements could be achieved. However, the improvements are becoming smaller and smaller. For engineers at Philips and Sony, it became clear that a new encoding technique would, perhaps, offer greater improvement over a longer term. The search was for an encoding technique that would enable us to capture sound in its most pristine way, and encode it in a manner that would produce the absolute bare minimum loss of sound quality due to downstream signal processing or encoding.

The majority of today's analog-to-digital converters are based on delta-sigma conversion, operating at several MHz levels. However, in today's PCM world, after the analogue signal is encoded at this high sample rate, the signal is then down-sampled to the conventional multibit word lengths of 16-bit to 24-bit and sample rate of 44.1kHz or 48kHz. This is usually done by multistage or 'cascaded' decimation digital filters. In D-A converters, the multibit PCM signal is oversampled by using multistage interpolation filters. (On conventional 44.1kHz sampling, 'brick wall' filters must pass 20kHz audio.) Given that the signal started as a heavily oversampled digital signal, one has to ask the question why filter and decimate? Why not use the delta-sigma signal directly. This is the essence of the principle (motivation) of DSD.

An increase of sampling frequency eases the need for 'brick wall' filters, but the need for filtering remains. Simply increasing the sampling rate does not do away with the requirement for multistage decimation and interpolation. Thus, the solution was sought away from multibit PCM. DSD eliminates, to a great extent, the sound quality degradation inherent to the filtering processes in PCM by simply bypassing the multistage filtering altogether. As in conventional PCM systems, the audio signal is first converted to digital by an oversampling delta-sigma modulator. However, unlike PCM, where the delta-sigma signal is decimated, Direct Stream Digital records the 1-bit pulses directly. The resulting pulse train is resistant to distortion, noise, wow and flutter of recording media and other transmission channels.

In 1995, a number of pioneering engineers at Sony under the leadership of Ayataka Nishio, developed and hard built the world's first professional DSD recorder. (AES preprint 456-1/1-71, 1-bit, 2-channel recording system, Ayataka Nishio, et al.) The engineers took this to top recording studios in New York, New York, LA, Nashville and London to demonstrate their design and collect feedback from the industry's golden ears. It took more than two years to arrive at a design that started to show the potential sound quality of DSD. By 1997, based on the initial design of the DSD recorder, more units were built and used operationally in recording projects around the world.

Limited by a straight 2-channel recording, many recording sessions were made and experience was carefully gathered and documented. This exercise not only provided the DSD engineers with priceless feedback regarding the DSD recording system, it also demonstrated its potential during recording sessions. Numerous projects have been recorded in the studio, many of which will serve as part of the launch catalogue of SACD software titles in 1999.

Having transferred the key enabling technologies from the central R&D labs in Tokyo to the product development divisions last year, Sony is now in the process of developing a range of pre-audio equipment based on DSD. At the recent 1998 AES Convention in Amsterdam, a range of experimental DSD production tools were demonstrated at the SACD format stand, jointly manned by Philips and Sony engineers. A wide-band microphone prototype, designed to capture much of the 100kHz bandwidth, the experimental stereo DSD recorder and a DSD editor, based on a Sonic Solutions editing workstation (AES preprint).

There is a long and honourable tradition where production equipment usually is of higher calibre than the final distribution format.
Due to its reduced processing level, DSD may need less hardware and thereby less power. Downsizing of hardware design will become possible. This is attractive, especially for portable applications.

Current DSD recording systems run at 64x 44.1kHz. This sampling frequency provides very reasonable performance in terms of bandwidth and dynamic range. There is a long and honourable tradition where production equipment usually is of higher calibre than the final distribution format. In this case SACD is specified at 64x 44.1kHz, so it is quite feasible that future DSD systems may be based on higher sampling rates, in order to provide an extra reserve in dynamic range and noise floor. However, even if the sampling rate stays the same, a DSD format has potential for improvement by changing the noise-shaping filter in the recording stage. However, unlike a PCM digital format, such a change does not require changes in the replay equipment. In other words, you can play back the new recordings on older playback machines and still get an improvement in quality.

The SACD offers 6 audio channels at full 'Scarlet' book specification. It will take quite some time before we are fully able to make proper use of the capability to reproduce full quality surround with SACD. Producing 5.1 sound mixes for video is not quite the same as recording a 5.1 mix for a classical recording session in a concert hall. To put it simply, the objective of the former mix is to most efficiently support the images provided by the video, the objective of the latter is to most efficiently reproduce the orchestra's musical performance in the listener's home.

Much is to be learned and developed in multichannel recording techniques which is why it may take a few years to fully exploit the multichannel capability of the SACD.

DSD provides the advantages of analogue and digital technologies. Being digital, it allows numerical representation, which, like in the PCM world, degrades free recording on tape and disc. We also believe that DSD is the most 'analogous' representation of the original sound and allows potentially the 'cleanest' conversion back to analogue.

Over the last five years, DSD has been successfully developed from a laboratory recorder to a first-generation experimental production system allowing the music industry to experience it at first hand and further develop its inherent potential. Commercial DSD products have been introduced, which no doubt will speed up the acceptance of DSD in the professional audio domain. The future should be interesting.

The technology digital-audio mixer was demonstrated at a private showing of DSD technology at Abbey Road studios in February this year. A more mature, still experimental DSD console was demonstrated at the 105th AES Convention. This was a fully functional 8-channel DSD mixer, with 5-band parametric equalizer, 24dB/octave LF and HF filters, adjustable delay. All processing and interpolation is done at 2,822kHz. Through this development, Peter and his team have shown that DSD mixing technology is a reality.

A Sony developed conversion system called Super Bit Mapping Direct, allows for DSD recorded audio signals to be down converted to CD format by using a single stage digital-audio filter. SBM-Direct is able to convert much of the high sound quality of the DSD signal into the PCM domain. This means that having used SBM-D, a normal CD sounds notably better when played on a normal CD player. A real-time SBM-D converter was demonstrated at the 105th AES Convention in San Francisco.

Also at this show, Sony demonstrated a functional DSD authoring system, as well as DSD watermarking processors, which record the visible and invisible watermark information on to SACD key elements for an anti-piracy and copy management system.

In short, the full chain of professional audio production equipment, from wide-band microphone in the recording studio, through digital recorder, mixer and editor, to authoring tools for the mastering and replication have been demonstrated in various stages of development, leaving little doubt about Philips' and Sony's commitment to develop and introduce DSD and SACD.
Resistance is useful

Audio circuitry is only as good as the components used to construct it. This month John Watkinson looks at resistors and resistance.

Resistance is the property of electrical materials which tend to obstruct the flow of current. Fig. 1 shows that this can be good or bad, depending on the application. Where the goal is to deliver power, resistance is a nuisance to be minimised and then tolerated, but there are plenty of applications where it is essential, for example in terminating a transmission line.

Georg Simon Ohm was the first to quantify the linear relationship between current (I), voltage (V) and resistance (R). Fig. 2 shows Ohm's Law expressed in various ways. The most famous analogy with electrical resistance is the dreaded garden hose resisting the flow of water—if not taken too far, the analogy has some uses. Certainly it illustrates that the resistance increases proportionally with length and goes down in inverse proportion to area. The behaviour of the material in question is called resistivity. Fig. 2 also shows that the actual resistance is obtained by multiplying the resistivity by the length and dividing by the area. It follows that the units of resistivity must be ohm-centimetres.

The case of a thin conductive film is interesting. These are found in the construction of some types of resistor, in EMC screens and electrostatic speaker diaphragms. Here the film thickness is constant so the resistance increases with length, but goes down with width. Consequently, the resistance of a square of the material is constant and independent of the dimension. Thus film resistivity is measured in ohms per square.

Resistance in power delivery is wasteful and so the most efficient way to deliver power is to minimise the current. Fig. 3 shows that the power is the product of the current and the voltage across the load. Some power is lost in the resistance of the wiring which also causes a voltage drop. At Fig. 4a a low voltage has been used and the power loss is significant. At Fig. 4b a high voltage has been used and the power loss has been dramatically reduced. This is why electricity distribution uses such high voltages. It is cheaper to invest in the insulation needed to carry a high voltage than to thicken the conductors to carry a high current.

Sometimes the effect of a resistance is experienced when there is no physical resistor present. This will be noticed on power sources such as batteries and transformers. Fig. 1 shows a dry battery off-load so that its terminal voltage is maximum. However, as soon as a current is drawn, the terminal voltage falls. This is because of the internal resistance of the device. Dry batteries have significant internal resistance and so are not good for producing high power as much of the power will be dissipated inside the cells. Lead-acid and Nickel-Cadmium batteries have much lower internal resistance. This makes them more efficient on high power loads, but also means that more care is needed. A car battery can easily melt a spanner placed across its terminals. Putting a NICad battery for a DAT machine in the same pocket as your car keys can result in smoking trousers.

Many beginners in electronics are puzzled by the fact that resistors are only available in peculiar values rather than in a simple round number of ohms. The reasoning behind this is actually quite logical. Manufacturers can only make a finite number of different values. Suppose that these were available in from 10Ω to 100kΩ in steps of 10Ω so that there was a choice of 10Ω, 20Ω, 30Ω, and so on. If my 10Ω resistor is too small, my only choice is to double it by using a 20Ω. However, if my 90Ω resistor is too small, I can choose 100Ω which gives an increase of a few percent.

Thus uniformly spaced resistance values give steps which are too coarse at one end of the range and too fine at the other. The concept of preferred values redresses the balance. For example the E12 range would have 10Ω, 12Ω, 15Ω, 18Ω, 22Ω, 27Ω, 33Ω, 47Ω, 56Ω, 68Ω, 82Ω in a decade, making 12 values, hence E12. You can pick any adjacent pair of these values and the high one will always be 20% greater than the low one. You can also plot the logarithm of these values and find that it produces a uniform progression.

Resistors are mass produced and so will suffer from tolerances. These vary from 10% for the most unpretentious product to 0.05% for critical applications. Where there are wide tolerances, having a large number of preferred values is meaningless. For example the 10Ω down on 8Ω2 is 7Ω which is also 10% up on 6Ω8.

If the E12 scale does not give enough choice, the E21 or even higher scales can be used. E21 gives 21 preferred values per decade which are 10% spaced. The tolerance of these E21 resistors must be tightened to 5% or better.

Part of a designer’s day is to do a worst case calculation to see what will happen when all components fitted are at the least favourable extreme of tolerance. If there is a problem, a redesign will be needed with more accurate parts. Interestingly the distribution of 10% resistor values is not necessarily Gaussian. Manufacturers make loads of

Fig. 1: Resistance—Good or bad?

Fig. 2: Ohm's Law expressed
This principle is also employed in the audio volume control.

Fig.3: Power is the current and voltage across the load resistor and then test them. The ones that are close are sold as 5%, whereas the ones between 5% and 10% are sold for less. Thus the chance of a 10% resistor being less than 5% in error are slim.

Resistance can be deliberately employed to waste power. This is the principle of the electric fire—which is simply a resistor in a box. All resistors develop heat in service: what matters is whether the amount is significant. The power dissipated is simply the product of the current and the voltage, and the units are watts. In audio control circuits such as preamps and equalisers, the amount of power dissipated is negligible and the resistors concerned can be very small. In a dummy load used for testing audio amplifiers, a considerable amount of heat may be created and a suitably high dissipation resistor must be used. Dissipation can be increased by raising the surface area and the temperature. Thus a high power resistor must be one that is capable of working at high temperature. In general high temperatures are bad news. Temperature cycling causes expansion and contraction, stressing lead seals. When lead seals fail, atmospheric pollutants can penetrate the element and corrode it. There is no alternative to the dummy load, but in many electronic designs, high dissipation can be avoided by suitable technique. In a studio, every Watt of power wasted in an inefficient device will be paid for in a raised electricity bill, and then another Watt is used to power the air conditioner to remove the first watt.

This principle is also employed in the audio volume control.

Fig.4: Dry battery loading

Fig.5: Potential divider made variable with slider.
Looking for a wizard of an OS

Mac OS is user-friendly yet delicate. Windows NT robust yet arcane. Simon Croft scans the horizon for a system that gives media professionals the environment they need.

LIKE IT or not, computer operating systems have become a big issue in the audio community. As mainstream business computers become ever more powerful, but proportionately more expensive, it is a cost-effective platform for audio recording, editing and mixing hard to ignore.

Such is the range of functions you can perform—recording, editing, processing and mixing—your computer is likely to become the heart of your audio system. If you choose to go that route. But a computer is a complex device, and when you buy one you also take on some of the responsibility for configuring and maintaining it. If you happen to work in an organisation with dedicated IT support, the never-ending stream of software upgrades and new peripherals may pass you by—but someone has to install them.

Hopefully, the vast majority of your time will be spent on billable activity, so it is of paramount importance that the working environment offered by your computer is an efficient one. This involves not only the usability of the graphical interface but also how good the system is at handling and processing multiple streams of audio. Today's OS vary in their resilience, but none of them are as reliable as a dedicated hardware solution.

Arguably the most reliable is Windows NT, which broadcasters in particular tend to favour for networked media. But NT is in no way a 'plug and play' system, making it less suitable for self-op applications.

Arguably the most 'user-friendly' OS is still Apple's, soon to reach 8.5. For years, Apple users have argued that their OS is vastly superior to anything from the Wintel camp. Actually, this is not true. Apple has the easiest Graphical User Interface, but the underlying OS is ageing fast in an environment where the demands placed on it are growing exponentially.

Windows 95/98 has a similar problem, in that it was not designed with high-bandwidth and real-time throughput in mind. Because additional layers of software are needed for the old OS to support new functions, a lot of the potential power of the new processor is lost in the increasing inefficiencies of the system.

These were some of the stumbling blocks that led former president of Apple's product division, Jean-Louis Gassée to form Be Inc in 1990. The company's web site (www.be.com) makes an eloquent case for a completely new OS. Better still, it offers BeOS Release 4.1 for Intel and PowerPCs at just under $100. BeOS appears to be excellent, but sadly that's where the good news ends.

So far Be Inc has failed to attract major audio, video or graphics software developers, and even Be's web site warns, 'This release is still definitely for geeks, and is available for free.' Ironically, this is the problem Apple faced when it bought Steve Jobs' company NeXT in an attempt to get a new OS off the shelf. Apple, which had also considered buying Be Inc to the same ends, now has the building blocks. But, by all accounts, it would have taken years for developers to create new versions of their applications. This stalled the whole project.

However, Apple could still turn out to be the cavalry. By the Summer of 1999, it aims to release OSx (pronounced 10). This is a completely new OS, not just a system update. It promises enormous speed and stability improvements through features including preemptive multitasking, protected memory and a high performance kernel.

In practical terms, this should mean that when an application crashes, it won't take the whole system with it. Likewise, the system should handle the use of simultaneous programs without the kind of 'stop go' performance associated with the current co-operative multitasking environment. Apple watchres might be forgiven for an acute attack of deja vu at this point, given the number of abandoned projects and botched fixes associated with Apple OS in recent years. But this time, it looks as if things are going to be different. Crucially, Apple reckons that existing applications will need minor updates, not complete rewrites to run on the new OS. It says most applications are 90% compliant already.

It is also continuing to develop OS9, which is set to reach 8.6 by the time OSX is launched. This 'everyone comes with us' approach is light-years ahead of its previous proposal: essentially an emulator running under the new OS.

Above all, Apple is the company with the most to gain from a media-oriented OS. Microsoft already has 99% of the market and, although you can place WAV files in a spreadsheets, most corporate users are pretty underwhelmed with their new machine arrives with a pair of speakers. By contrast, nearly all Apple's business users are in some form of electronic media: publishing, graphic design, web design and, yes, running DigiDesign's Pro Tools.

Interestingly, DigiDesign has just announced that Pro Tools will now be available in Windows NT and Apple versions. For DigiDesign that makes good business sense. But before Apple users consider migrating, they could be advised to see if X hits the spot.
The most powerful, intuitive, creative digital console ever offered at any price. Twenty-four-bit definition. The superb sound quality and intuitive interface that Mackie's famous for. Expandable with cards and software plug-ins. The Digital 8*8. Well worth the wait.
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