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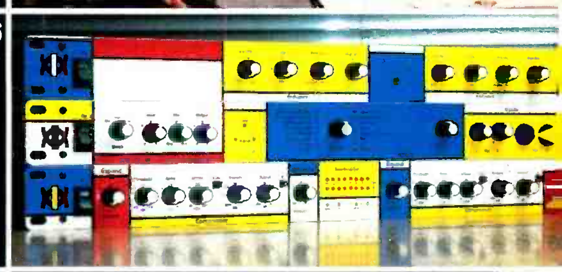
CONTENTS

SEPTEMBER 2000 Volume 42 Issue 9
POSTPRODUCTION • RECORDING • BROADCAST

78


80


44


46


16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Our world, and welcome to it

September 2000 Vol 42, No 9. ISSN 0144 5944

WHAT BEGAN AS A REGULAR MAGAZINE redesign required us to examine the business we are in and to distil the coverage that we give you. Market research conducted among readers and manufacturers confirmed that technology and market forces are causing convergence in certain areas and divergence in others. As we are committed to the key areas of postproduction, broadcast and recording, these areas have been the focus not only of an improved editorial package but a refined circulation.

The magazine you are holding this month is larger and boasts a higher quality paper stock. In addition, the redesign makes information more accessible while maintaining our industry-leading editorial standards and integrity. We've also appointed an exceptional group of consultant editors to advise the postproduction, broadcast and recording issues.

Reader feedback has applauded our independent handling of product reviews and bench tests, and supported our application features. But you will now see an increased emphasis on techniques and practical issues. Our commitment to education and the analysis and appraisal of future technologies has been vindicated. We're also doubling the amount of news and analysis to serve as a more comprehensive one-stop read for the busy sound professional. There is even a renewed levity in odd corners of the magazine. Read on, we're certain you'll enjoy it.

With great pleasure we welcome six new high-profile consultant editors. Grouped to reflect the magazine's core coverage, our new blood will maintain a regular dialogue with *Studio Sound*'s editors to offer feedback and pick up on trends from the front line and help steer our coverage. These busy individuals will even be invited to share their thoughts, opinions and experiences at the sharp end of the business.

All operate at the top of their chosen discipline and will create a body of experience and authority never before been assembled for a pro-audio magazine.

Broadcast: Florian Camerer joined ORF (the Austrian Broadcasting Corporation in Vienna), in 1990 as a sound engineer and became a staffer soon after. Working mainly in the field of production sound and postproduction, he made high-quality audio for documentaries his field of special interest and became involved in multichannel audio in 1993. He mixed the first ORF programme in Dolby Surround (Arctic Northeast) and is now responsible for all aspects of multichannel audio for the broadcaster. Today, he trains in multichannel at ORF and is a member of the AES, VDT, OeTMV (Austrian Tonmeisters) and IBS.

Chris Wolters is head of sound engineering at VTM (Vlaamse Televisie Maatschappij in Brussels), a minority language station which against seemingly insurmountable competition was able to break even after just four months operation. Chris joined at VTM's inception in 1988 graduating from chief engineer to head of sound operations when the operation was drawn together in 1993 and drew up the audio infrastructure for the broadcaster's new building. A second TV channel, Kanaal 2, was started in 1994 and more recently a network of local radio stations for which Chris is also unit manager. He still engineers and is excited by the prospect of working with *Studio Sound*.

Postproduction: Paulo Biondi, MD at International Recording in Rome has experience in film following the technologically revolutionary years to the present day. International Recording was established by Paulo's father in 1957. Today it operates in film production, TV and multimedia serving an international client base.

MD of The Tape Gallery, **Lloyd Billing** served at Columbia Pictures and Advision where he tape op'd for Geoff Wayne, Yes and ELP before mixing commercials at Leeward Sound Studios. When a client suggested they set up a studio together in 1981, The Tape Gallery was born. Now a major player on London's Soho post scene, it was the first commercial studio to synchronise sound to picture and pioneered the use of DAWs (Synclaviers), ISDN, digital pictures and the SohoNet ATM network.

The Tape Gallery Group of companies includes a radio production company, jingle and composition company, an on-line sfx database, a multimedia company, an on-line searchable voice over database and Minno Film Editors which specialises in nonlinear film editing for the advertising industry.

Recording: producer **Arthur Baker** needs little introduction, but it's worth noting his part in moving dance music from the hedonistic fringe to the mainstream. Having worked his teenage years in a friend's record shop, served a DJ's apprenticeship in Boston and worked the New York disco scene in the mid seventies Arthur entertained a film and journalistic education before returning to Boston for an engineering course at Intermediate studios. Moving to NYC in 1981, he scored multi-platinum success with New Edition's 'Candy Girl', produced Planet Rock for Afrika Bambaataa and the Rockers Revenge rework of 'Walking on Sunshine'. All have ensured his place in history as well as his authority as a music correspondent.

Trevor Fletcher's career at Miami's seminal Criteria studio complex began before school. With his mother taking bookings, Trevor was free to play in the studio and soak up both the success and the practicality of the studio that produced records of the stature of Aretha Franklin's 'Young Gifted and Black' and James Brown's 'I Feel Good'. Through over 17 years at Criteria, he has seen the studio evolve into the heady success of the seventies and eighties, through the straits of the nineties and into a brave new world under the ownership of New York's Hit Factory. Having undergone a radical refit, Trevor is welcoming a new generation of artists to a reinvigorated studio.

Zenon Schoepe & Tim Goodyer

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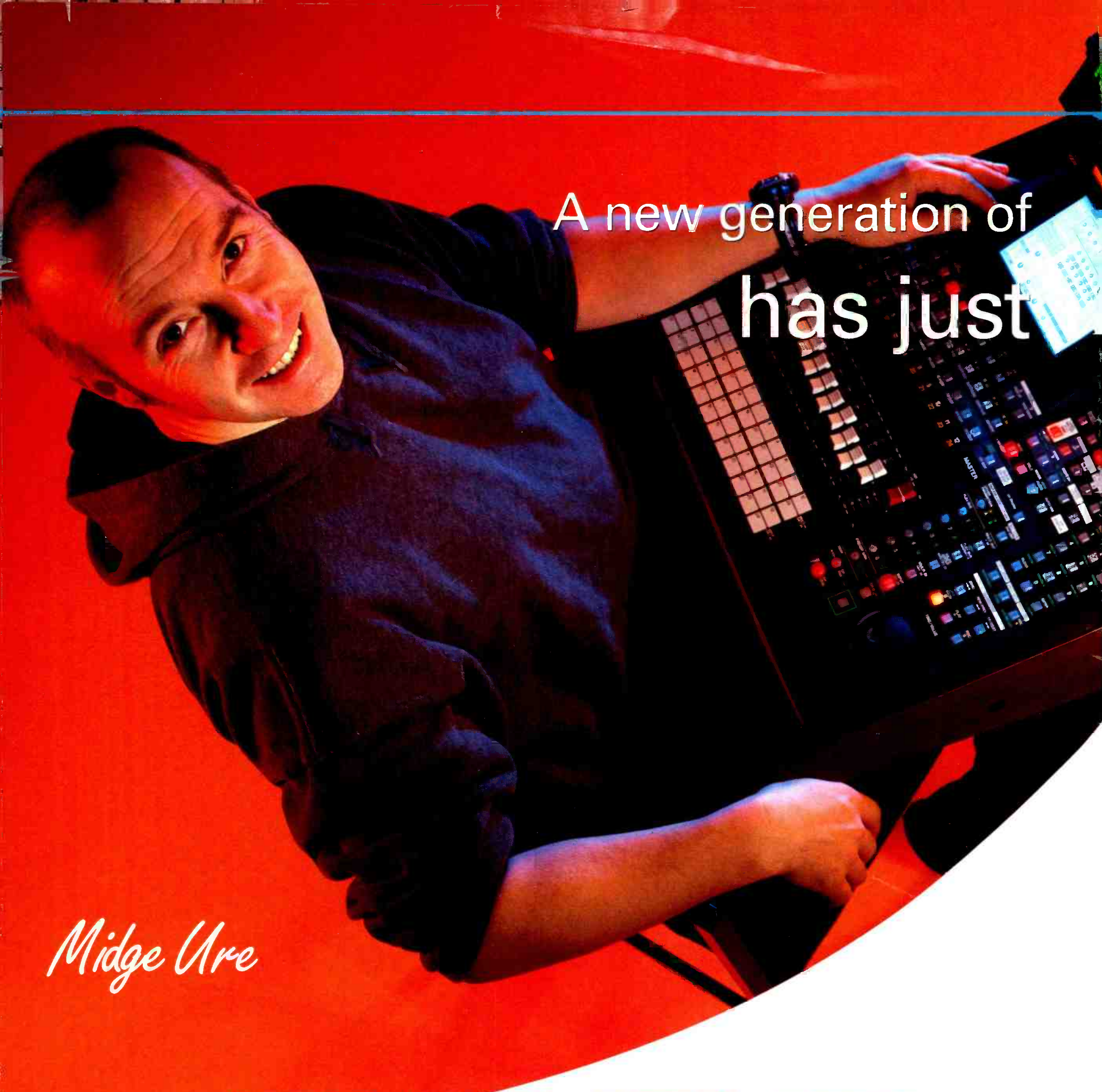

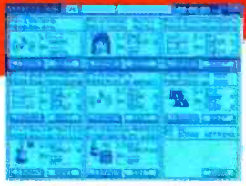

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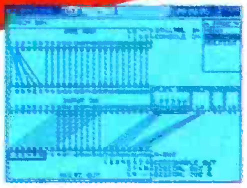
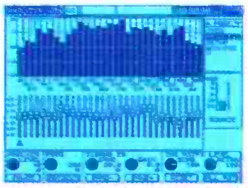


The virtual patchbay offers unrivalled flexibility. Route any of the inputs and outputs internally, with not a patch lead in sight.



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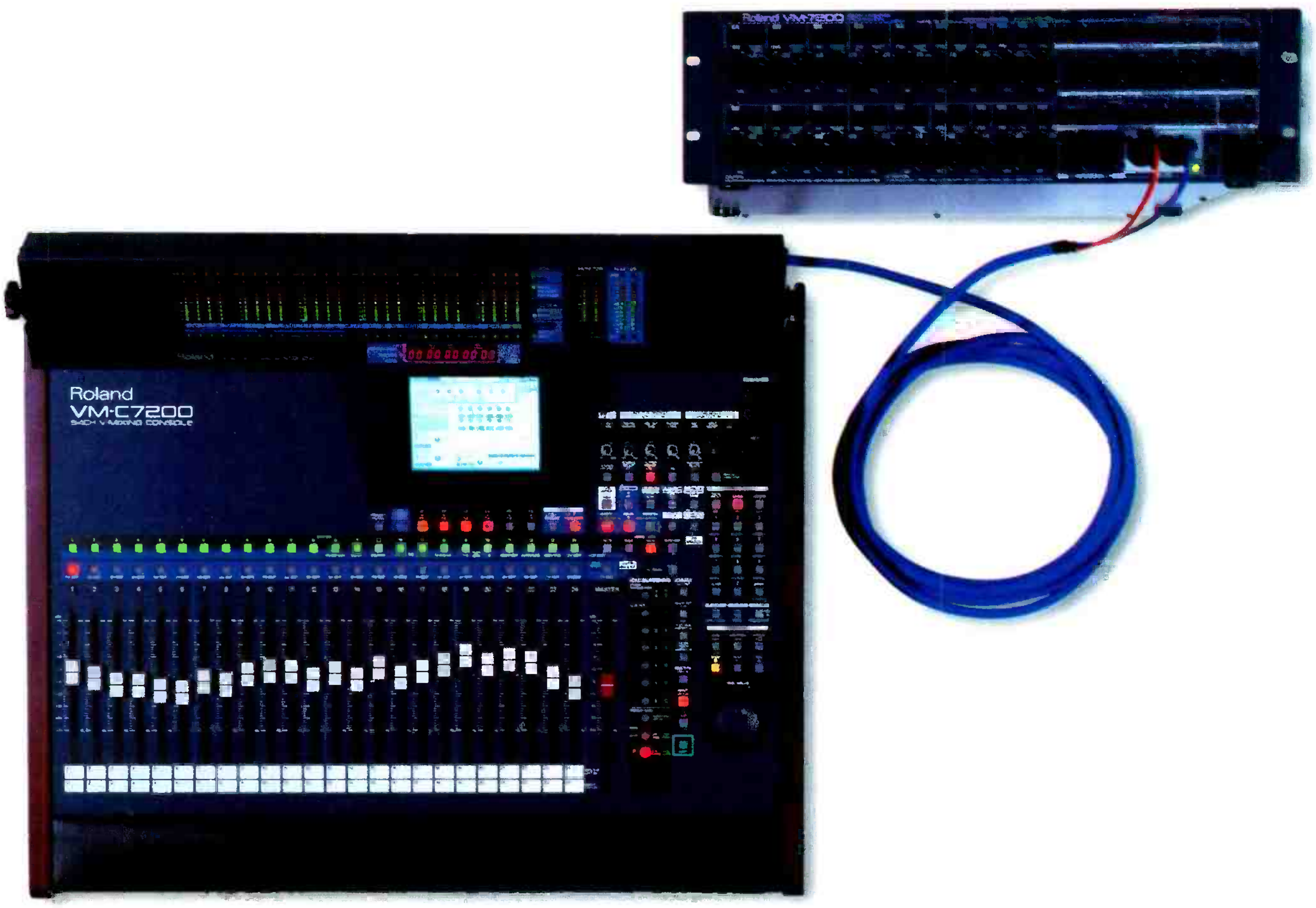

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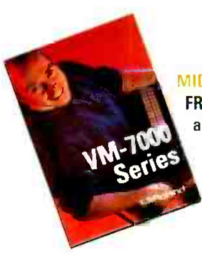

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Netherlands: Veronica Netherlands is the first broadcaster in Holland to have surround monitoring in its transmission suites following the installation of Bryston-powered PMC speakers in its new transmission suites. The largest installation is in the digital transmission studios where shielded IB1S monitors are used for LR with a matching dedicated dialogue channel. The 5.1 system is completed with TB1S surround units suspended at the rear of the room. The second installation is of free-standing IB1S in Veronica's transfer room.
Audio Connections, Netherlands. Tel: +31 226 318 800.
PMC, UK. Tel: +44 1707 393002.

Germany: Hamburg's Nemo Studios has installed a 64-fader, 160-channel Soundtracs DPC-II digital console as part of an extensive renovation. The recording facility is owned and run by

producer-composer Frank Petersen, noted for his work with Ofra Haza, Marky Mark and Enigma, and recently hosted Sarah Brightman's album *La Luna* on the DPC-II.
Soundtracs, UK. Tel: +44 1372 845600.

Egypt: Egyptian Radio & Television Union has purchased two Studer mixing consoles for its Cairo TV Studios. A 104-input, 24-fader Studer D950 digital desk will handle TV production in Studio 10, while an analogue Studer 980 will be used for on-air broadcast in Studio 7. Studio 10 will be the first fully digital TV production studio at ERTU, being used for postproduction, film dubbing and cartoon production. The studio will use Studer A5 active monitors and an A812 tape recorder as well as equipment from AKG, BSS, Eventide, Gotham Audio, Sony and Tascam. Studio 7 is an analogue on-air TV studio based around a 24-input Studer 980 also using A5 monitors and an A812 tape deck. ERTU now has eight studios equipped with D950s.
ERTU, Egypt. Tel: +202 574 6883.
Studer, Switzerland. Tel: +41 1 870 7511.

Uzbekistan: State broadcaster RTV has taken delivery of three SSL

Telarc rejects watermark

US: Leading audiophile label Telarc will not include the Verance anti-piracy watermark on its debut DVD-Audio titles due out this fall, Webnoize has reported.

Michael Bishop, a Telarc recording engineer, said the label is concerned the watermark may be audible. Moreover, he told Webnoize, there is no guarantee Verance's copyright protection scheme won't be defeated by pirates. The DVD-Video encryption scheme CSS was hacked last year.

While Telarc is not the first record label to express reservations about the watermark, it is one of the first DVD-A advocates to reject the watermark. Telarc's unilateral boycott could trip up an already shaky product launch, one that has brought forth a few high-end players and practically no software. Then again, the watermark has been licensed by all five major labels, and 4C Entity LLC—DVD-Audio codevelopers Intel, IBM, Matsushita and Toshiba —has chosen the watermark as part of the format's copy-protection system.

Telarc has not written off the watermark entirely, Bishop noted, 'if it can be shown to be transparent to the end-user and not have any long-term listening effects'.

A 5.1-channel mix of Weather Report's *Celebrating the Music of Weather Report* is to be the first DVD-Audio title from Telarc following its success as a stereo release earlier this year. The record was mixed by Doug Oberkircher and Jason Miles. 'It takes the record to the ultimate place,' Miles said of the 5.1 mix. 'It takes the listener to a new level with the amazing performances. It's a real experience for the senses.'

Rocket fuel

World: When Rocket Network first got off the ground in 1995 (as Res Rocket Surfer), the Internet had neither the prominence nor the impact on work and leisure that it has today. Originally the Rocket system was MIDI only, but the concept of networked musical collaboration via online virtual recording studios was already in place, as was the use of a text-chatting interface for online communication between musicians.

The longer term goal was to add audio to the collaboration equation, and so Rocket Network has grown into a 40plus team with a $15m venture capital funding injection from, among others, Digidesign parent Avid. Speaking at the time, Avid's President and COO David Krall, who now sits on the Rocket Network Board of Directors, commented: 'We believe the Internet has the potential to fundamentally shift the way media professionals work. We intend to be a driving force of that change, and Rocket's pioneering work is an excellent match for us. The concept of 'real-time' globally distributed postproduction will allow the creative pros to truly push the limits of their craft.'

Avid's investment in the company (a 19.7% equity stake) will see Rocket Network's collaborative technology integrated into Pro Tools, and with the high profile and large user-base of Pro Tools in all manner of professional audio production environments, Rocket Network and its online collaborative approach will reach deeply into the pro-audio industry. In addition, Euphonix is Rocket-enabling its high-end multitrack recorders and digital audio consoles, a first sign perhaps of the way that production equipment will develop to support networked working environments and methods.

Adding audio to the Rocket system has always been the key to wider application of the software, along with the integration of collaborative software functionality into established music production tools. Rocket-powered versions of Cubase VST and Emagic Logic Audio are now available which smoothly integrate the Rocket Network collaborative interface and functionality into familiar production environments—as was ably demonstrated recently by cofounder Willy Henshall on a flying visit to London. With the assistance and contributions of assorted fellow musicians at various geographical locations, Henshall showed how easy it was to put together a 30s ad spot to picture (Quicktime movie) using Logic Audio 4.5. QDesign audio compression can help keep file sizes down, and a variety of compression amounts are available, while uncompressed audio can also be sent.

But it's not only traditional audio work that can be handled in new ways by the Rocket Network collaborative approach. Rocket Network CEO Pam Miller comments that interactive audio for web sites is an emerging application, and that Rocket Network is working with Beatnik on developments in this area. The company also has more ambitious plans which develop from the ramifications of working in a networked environment. Miller talks of plans for an online 'talent brokerage' that will allow audio professionals to advertise their services and hook up or be hooked up with one another into project teams which can form and dissolve as and when required. And for musicians who want to go on and sell their tracks online, the company is, rather than offering its own service, establishing relationships with existing online sales outlets—wisely so not least because no-one knows what or if any one business model will work for online music.

Rocket will be announcing a series of third-party partnerships at AES Los Angeles this month, marking a significant shift up market for its file exchange system. DSP Media, the Australian DAW manufacturer recently expanded in Hollywood and Europe, is to add Rocket Power to its flagship Postation II; Waveframe is codeveloping its DAWs; while Euphonix is planning to modifying its digital multitrack recorders and mixing desks. The latest Euphonix mixer uses TCP-IP for internal communications, making it internet-ready.

Brazil: Leo Garrido has been awarded the Brazilian Audio Award 2000 in the Professional of the Year in Live Recording category for his work on the Paralamas Unplugged sessions recorded during the AES Brazil 2000. Garrido, owner of XEF Sound Productions, also managed the mixing of the CD and DVD formats with partner Vinicius Sa. In addition to the Audio Award 2000, the recording has been nominated for the first Latin Grammy Awards in the rock album category and will be released in multichannel DVD format. On the night, Garrido used a variety of Sennheiser and Neumann microphones including the MD509, MD441, MD431, and MD421 II dynamics; e845 and e604; ME 64/K6 and KM184 condensers; and the Neumann KU100 binaural head.

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AES will also unveil details of a Rocket Network Talent Brokerage service, which will exploit the network to circulate freelance engineers and producers, and news of a Rocket-powered Pro Tools. Digidesign recently took a 20% stake in the company.

'These new development partnerships, along with others previously announced, make Rocket Network the undisputed industry standard for online audio collaboration for everyone from high-end professionals through to project studios,' comments Rocket president and CEO Pam Miller.

London's Strongroom Studios has already become a Charter Partner with Rocket Network, and Air Lyndhurst is to follow suit this month.

Net: www.rocketnetwork.com.

Dial M for music

World: Mobile music, or more specifically music on mobile devices, may not be anything new—think Walkman. But the subject is generating a lot of interest because the mobile devices in question are networked, and the availability of streaming music is seen by some as a killer app for the broadband mobile phone technologies that will emerge over the next couple of years.

Mobile phones are already starting to metamorphose into 'information devices', networked or otherwise, and while the present generation of WAP phones just starting to emerge may not be particularly inspiring, larger screens and faster mobile connections will improve matters considerably. Also, music will be delivered digitally over an IP-based connection to the upcoming 115kbps GPRS and subsequent 2Mbps UMTS mobiles, so we're not talking about music over the narrow bandwidth of an analogue phone connection.

Meanwhile, the miniaturisation of today's processors and memory cards, power and storage can increasingly be integrated into small mobile devices. Low-power chips like Transmeta's Crusoe hold out great potential for mobile devices, while flash memory storage like the new postage stamp-sized SD memory card make mobile storage a reality (if not particularly cheap yet). Samsung's recent release of an MP3 phone may seem mere gimmickry, but it's more likely the forerunner of more ambitious musical mobile devices. Sanyo plans to market a combined mobile phone and music player that can also download music in Japan later this year. Using a low-cost mobile phone technology called Personal Handyphone System, phone users will apparently be able to download a 5-minute song in five minutes, at a cost of ¥50 in phone charges plus a charge of about ¥200 per song. The longer-term plan is to enable key-protected downloaded songs to be passed around on memory cards, allowing additional users to purchase a song key over the phone without having to download the song.

Meanwhile, streaming media technology company RealNetworks recently announced an agreement with Nokia to develop and distribute media delivery technology for future mobile devices. As part of the agreement, the two companies will implement RealNetworks' RealPlayer technology in Nokia's EPOC-based communicators and smart phones, in time allowing mobile device users to access RealAudio and RealVideo content on the Web. Speaking at the time of the agreement, Anssi Vanjoki, Executive Vice President of Nokia Mobile Phones commented. 'Streaming media is a good fit with our vision of the Mobile Information Society and introduces a new dimension to the mobile phone user experience', while RealNetworks Chairman and CEO Rob Glaser added: 'RealNetworks strongly believes that mobile devices will play a paramount role in the future of computing and information exchange—our work with Nokia is focused on bringing the compelling medium of streaming audio and video to the millions of mobile device users worldwide'. The first RealPlayer-enabled Nokia EPOC products are scheduled to be available in 2001.

Russia's new house

A PROPOSAL TO CREATE a world-class music recording and production facility in St Petersburg has been put together by a team of leading industry figures, including Anthea Norman-Taylor of management company Opal-Chant, Innate Management's Pete Dolan and leading Russian business interests. The Menshikov Music Complex will comprise audio and visual recording studios in the former riding stables of the Menshikov Palace, an historic building in the very heart of the city. Pete Dolan, managing director of the startup phase, exclusively outlined the main aims of the project to *Studio Sound*.

Q: *What are the attractions of St Petersburg?*

St Petersburg has three of the world's top orchestras and is a hub of creative energy, so this complex will be capable of handling the most demanding orchestral recordings and music projects. Not only that, it will be situated in a prestigious and spectacular location within Russia's cultural capital.

Q: *Is it solely for orchestral recording?*

Definitely not. The site will combine a 3-studio recording complex with pre- and postproduction facilities, CD and DVD mastering, and will also develop training and other initiatives for the Russian music business. The orchestral studio will be bigger than Abbey Road's Studio One, capable of accommodating a 150-piece orchestra and choir. Studio Two will be a tracking studio for contemporary music artists from both Western and Eastern Europe; and Three will be a budget-level facility catering to domestic requirements.

Q: *How do you hope to lure Western productions to the Baltic?*

First of all, the whole complex will be of the same calibre and capability as both Abbey Road and Air Lyndhurst in London. On top of this, we have secured very advantageous rates to allow us a favourable 'fixer agency margin' through our agreement with all three of St Petersburg's orchestras, and these rates are substantially less than the top London and American orchestral rates.

Then you've got St Petersburg itself, which is a beautiful environment on the verge of a real renaissance. And, of course, there is an established precedent for recording film soundtracks away from the UK-US hub. Facilities in Dublin, Prague and Budapest already take advantage of significant film market bookings, and while the Menshikov Complex will be far superior to these studios the rates will remain as competitive.

Q: *In what way is St Petersburg 'on the verge of renaissance'?*

The city was cut off from the West for over 70 years after the Russian Revolution, but western music was keenly absorbed either via The World Service and Voice of America radio stations or via smuggled cassettes. Russian followers of the jazz and pop scenes were often as educated in their field as their western counterparts, leading to a burgeoning music industry once the old system collapsed. Now, record stores selling CDs, cassettes and videos are commonplace and the demand for western and home-grown music is flourishing. Indeed, Russia's MTV service is available free to every household.

Such is the lure of the St Petersburg music scene that the long-established Frankfurt Music Fair has chosen the city for its first ever Fair in any location other than Frankfurt. The show—next June—will stimulate the market still further in St Petersburg and the rest of Russia.

Q: *Who else is involved?*

Munro Associates is on board as the acoustics consultancy; we've got Patti Nolder—who was at Air for many years—playing a key part in our marketing and PR; training and education will be organised by David Ward at Gateway; and our Russian partners are well placed within the local scene to help us navigate all of the business development. Some of them even own the best restaurants in town!

Q: *Is piracy a big problem in Russia?*

Piracy is not an issue that will affect Western clients using the studio for Western productions. But even as far as the domestic work is concerned, we expect to see a rise in the legitimate market under Vladimir Putin—his policies are very focused on fighting corruption. Innate Management. Tel: +44 20 8693 4276.

CONTRACTS

SL4000 analogue mixing desks as part of the first phase of a major refurbishment programme, with the further consoles expected to follow for the second phase. The radio facility is based in the capital, Tashkent, where the consoles will handle recording and mixing national music programmes.
RTV, Uzbekistan. Tel: +998 711 331 953.
SSL, UK. Tel: +44 1865 842300.

US: Hollywood's The Steakhouse Studio has installed 56 channels of Martinsound Flying Faders on its classic EMI Neve console. Fitted by Phoenix Audio's Geoff Tanner, the system has seen early service with Japanese recording artist, Masami Okui—known for her work in Japanese anime films. The Steakhouse's EMI Neve is reckoned to be the only vintage Neve console world-wide to incorporate a fully featured Flying Faders control panel, modified from an AMS Neve V3 module. The console is constructed from two 24-channel frames offering 56 inputs, 24 buses, 24 tape returns, and 12 effects returns, providing a total of 92 inputs. The Steakhouse also houses post, composition, and voice-over suites.
Steakhouse, US.
thesteakhouse.geo@yahoo.com
Martinsound, US. Tel: +1 626 281 3555.

France: Post house Les Studios de Saint Ouen has installed an AMS Neve Digital Film Console. Sited in Auditorium D at the Paris-based facility, the DFC will contribute to the studio's French dubbing of 'foreign' films—a record 73 completed in 1999.
Les Studios de Saint Ouen, France.
AMS Neve, UK. Tel: +44 1282 457011.

Japan: Tokyo-based post facility, Q-tec, ordered a Euphonix System 5 for installation in its MA1 Postproduction Studio where it will be used for digital broadcast, including high-definition transmissions. DVD's requirement for surround-sound and 96kHz sample rates also influenced its selection. Another Tokyo-based post house, Omnibus Japan, has ordered a Euphonix System 5 digital console for a new flagship long-form HDTV postproduction studio. With the room scheduled for completion in late August, the desk brings the total of System 5 orders to 34.
Q-tec, Japan. Tel: +81 3 3589 2373.
Omnibus, Japan. Tel: +81 3 5410 6500.
Euphonix, US. Tel: +1 650.846-1190.

UK: The BBC World Service is to install three new Audionics broadcast consoles in the Balkan region. Over

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CONTRACTS

the last ten years the BBC World Service has been developing a series of overseas offices. In the mid 1990s limited facilities, were established in Tirana (Albania), and Skopje (Macedonia). Increased activity in the region over the last few years has meant offices in Tirana (Albania) and Skopje (Macedonia) have become increasingly significant while the cessation of hostilities in Yugoslavia has prompted the opening of an office in Pristina (Kosovo). The custom ACE Mk.IV consoles standardise the facilities in their Balkan offices. The BBC has also introduced an FTP filing system and communications enhancements to the offices. The new mixers installed in Tirana and Pristina are now online.
BBC World Service, UK.
Tel: +44 20 7257 2941.
Audionics, UK. Tel: +44 114 242 2333.

Belgium: Brussels-based No Noiz has installed a Fairlight FAME system replacing the obsolescent Fostex Foundation installed at its opening five years ago. The audio facility's partnership with video, graphics and animation houses was chosen to provide ready transfer of files to four Avid systems and future expansion.
No Noize, Belgium. Tel: +32 2 241 2626.
Fairlight, Europe. Tel: +44 20 7267 3323.

Russia: Moscow's A&T Trade has bought two Sintefex Audio Replicator FX8000 digital signal processors.
Net: www.sintefex.com

Northern Ireland: Ulster Television has ordered a 48-channel chassis Calrec S2 desk as part of the Studio 1 refurbishment at its Belfast studios where it will be used for live broadcasts including the Friday night music and chat show, Kelly.
Ulster TV, UK. Tel: +44 1232 328122.
Calrec, UK. Tel: +44 1422 842159.

US: Gateway Studios has installed four dCS 954 D–A and four dCS 972 D–D convertors. Based in Portland Maine, Bob Ludwig's mastering operation will use the convertors primarily in conjunction with SADiE and Sonic Solutions workstations, on 24-bit, 96kHz and DSD projects. As well as project interchange the convertors were the only units that will convert up and down from 192kHz; and from DSD to PCM. They will also be used for bit-for-bit cloning format conversion, and converting between dual and single AES and vice versa.
Gateway, US. Tel: +1 207 828 9400.
DCS, UK. Tel: +44 1799 531999.

The most striking conception of the potential of music and mobiles perhaps comes from SSEYO, the company best known for developing the Koan interactive music system. SSEYO recently announced the SSEYO Phone, a 'concept phone' designed as a virtual example of how audio on mobile phones and other mobile devices may develop. A drum synth (based on SSEYO's Freedrum software), high-quality polyphonic ringtones, full-length polyphonic RingTrax, and Audioicons attached to emails are among the ideas that SSEYO are putting forward. More annoying noise, or a welcome improvement on the annoying simple tones of today's mobiles? These possibilities may not be so far off. According to SSEYO, its Koan Interactive Audio Engine will soon be available on mobile devices through Tao Group's media stack. Even more ambitiously, SSEYO is talking of Backtrack interactive audio backgrounds for conversations, Group jamming over mobile networks, and a Remix capability. Will the mobile phone ever be the same again?

UK: London-based postproduction and recording facility The Town House has installed a PMC monitoring system in the 5.1-capable Mastering Room 3. The system is soffit-mounted throughout and comprises BB5/XBD (LCR) and MB1 3-way monitors at the rear. All channels are powered by a custom range of PMC series Bryston amplifiers and electronic crossovers. Rather than separate sub-bass unit, the system folds back the '.1' channel into the LR main monitors.

Web mastered

US: Sterling Sound president Murat Aktar says the decision to adopt IBM's digital rights management technology is based on Sterling's conviction that electronic music distribution will become an increasingly significant part of the music and entertainment industries. The move also makes Sterling the world's first independent audio mastering house to license EMMS (Electronic Media Management System).

The tools needed to master EMMS-infused music at the renowned mastering facility were installed in late July, in anticipation of the potentially huge on-line retail music market. 'We're not doing it necessarily to start generating revenue from it immediately,' Aktar commented. 'What we want to do is be part of the process.'

The process officially began last June when the five major record companies and IBM conducted a 6-month trial of EMMS. About 1,000 participants, mostly in San Diego, were able to buy major label releases over the Internet. While several of the labels since have begun selling songs and albums via the web on their own, currently the offerings represent a tiny fraction of a label's entire catalogue.

During the same period, a bevy of competing digital rights management (DRM) products have hit the marketplace. While noting that 'there are other companies that... have compelling bits of technology,' Aktar said IBM is 'the one that has a truly integrated solution, front to back.'

With EMMS, Sterling's e-Mastering service can compress audio in multiple formats and add copyright encryption, watermarking, liner notes and artwork and metadata. Songs also can be set to control related electronic devices, like CD burners or portable Internet audio players.

Down the road, said Aktar, Sterling will offer other DRMs besides EMMS. The facility plans to arrange listening tests, allowing clients to compare compression formats, bits rates and other technical variations. He called the future of Internet music compelling, but complex.

'I think that one of the things we've seen is that it's easy to distribute music electronically, with no security features and no quality control. To distribute a high-quality product and to monitor the rights and to distribute it securely is very complex, and requires the co-operation of lots of groups.'

DAB tie-up

US-UK: NTL has taken a US$1m stake in digital broadcast specialist RadioScape. The news comes two months after Psion PLC's US$4.5m strategic investment.and represents a vote of American confidence in the UK digital radio industry.

RadioScape MD Peter Florence commented, 'This agreement will allow us to work closely with NTL to deliver unrivalled next generation digital radio solutions. Digital radio is the future of wireless broadcasts and it makes sense for us to work with trusted partners who share our enthusiasm and vision in this arena. We have been shaping the development of digital radio to date, and now both companies can work together to bring digital radio to the mass market.'

Peter Douglas, group managing director of NTL Broadcast, added, 'The decision to invest in RadioScape was a natural evolution of our relationship. We're at the beginning of a revolution in radio and by investing in the technology we expect to stay at the forefront of this industry.'

RadioScape's technology is employed at the core of Digital One, the UK's largest commercial digital radio network and partly owned by NTL as well as run on NTL's broadcast infrastructure. This system has been transmitting the digital services of Virgin Radio, Capital Radio, Classic FM and News Direct since November 1999.
Net: www.radioscape.com
www.worlddab.org

Radio-active

UK: Capital Radio, Britain's leading commercial radio group has launched its online music entertainment strategy for new media division Capital Interactive. Three narrowcast web radio stations will be launched by Autumn 2000 to provide music to valuable niche audiences. A new online music brand, kikido, has been created and will appear as a stand-alone brand as well as on all sites including the three key Capital sites: 95.8 Capital FM, Capital Gold, and Xfm, and the narrowcast stations. Deals have been signed with record companies BMG, Universal, EMI-Chrysalis-Virgin, Jive, and AIM (representing the independents) who together account for around two-thirds of UK album sales. Capital Interactive is the only UK online music provider to have such online rights with major and independent record labels. Technology deals have been

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APPOINTMENTS

Amek has named Nick Cook as sales & marketing director following a long relationship between the former head of Fairlight Europe and the OEM team behind the Fame and Prodigy consoles. Cook began his career as an engineer in the mid seventies and joined SSL in 1987 where he progressed to Head of Sales before moving to set up Fairlight Europe.
Net: www.amek.com

Studer UK has appointed Andrew Hills as managing director where he replaces David Pope. Hills joins from SSL where he headed broadcast sales and was instrumental in creating SSL's German office. Prior to this he worked in a similar capacity at Neve.
Net: www.studer.ch

Mackie Designs has announced the promotion of Jay Schlabs to the position of national sales manager. In this position, he will be in charge of American national sales for all product

lines, including Mackie, Mackie Digital Systems, and Mackie Industrial Contractor Products. Schlabs joined Mackie in 1994 as a technical sales trainer and in 1996 was promoted to western regional sales manager. This follows the promotion of Scott Garside to recording products marketing manager from director of market research. In this position, Garside is in charge of marketing 'all things recording' including the D8B and ancillary plug-in products, HDR24/96 hard-disc recorder, HR824 monitors, and the 8-Bus consoles.
Net: www.mackie.com

SPARS executive director Shirley Kaye is to stand down after 13 years in the post in favour of Larry Lipman. Lipman will be leaving the position of director of degree programs in Recording Technology and Music Business at The University of Memphis. Lipman is currently nominated for Governor of the AES as well as serving on various AES committees.
Net: www.spars.com/spars

signed with RCS, Microsoft, eHNC, IBM and Open Market. Partnership deals have been struck with Handbag.com, Microsoft and Sports.com, and further distribution deals are under discussion.

David Mansfield, chief executive of Capital Radio, commented, 'Capital Radio is in a unique position to succeed on the Intemet. We can draw on many years' experience in the radio business, our trusted brands, close relationships with artists and record labels, and cross-promotional opportunities to ensure that we have an unbeatable offer for consumers and advertisers.'

Deep water

Germany: The Fraunhofer Institute for Integrated Circuits, Applied Electronics IIS, has unveiled its advanced watermarking technology to help content providers to keep track of their content and protect their intellectual property. Fraunhofer Bitstream Watermarking technology bundles Fraunhofer's robust watermarking scheme with its suite of high-performance audio coders by allowing direct embedding of watermark data (such as digital signatures) into coded music content. In this way, material can be personalised and traced in the event of illegal proliferation. Fraunhoffer insists that this is an important step for the music industries and the secure digital distribution of audio files, and that direct embedding of digital watermarks into coded bitstreams reduces time and cost efforts for the companies while preserving optimum signal quality.

Fraunhofer IIS-A has already presented the world's first bitstream watermarking scheme for MPEG audio coders at the European AES convention (based on MPEG2), and further technology will be released at the American AES: MP3 support in the new bitstream watermarking technology now allows seamless and efficient data embedding for the *de facto* audio coding standard on the Internet

'The new bitstream watermarking systems were designed with two goals in mind: achieving the best possible performance of the combined codec-watermark system and maintaining the renowned Fraunhofer audio quality.' said Christian Neubauer, in lead of the development effort. 'Much effort has been put into rigorous listening tests and optimisation of the psychoacoustic models, taking advantage of Fraunhofer's codec expertise.'

The institute is a leading research laboratory in the area of audio coding. Since the start of its audio coding work more than 10 years ago. Fraunhofer IIS-A has participated actively in the development of audio compression algorithms. Major parts of MPEG-1 Layer-3 (MP3), the most popular audio format on the Internet, have been devised at its headquarters in Erlangen, Germany. In addition to audio coding technology, Fraunhofer IIS-A is also working on data-hiding technologies for use in watermarking and fingerprinting systems. Fraunhofer IIS-A is active in international efforts to develop methods to technically manage and protect intellectual property, including the MPEG-4 work on Intellectual Property Management & Protection, the AES' activities on Internet Audio Delivery Systems and the Secure Digital Music Initiative initiated by the RIAA, RIAJ and IFPI.
Christian Neubauer. neu @iis.fhg.de

Indian summer

India: BBC Resources has signed a deal with state broadcaster Doordarshan Television for the implementation of digital terrestrial television (DTT) in India. The contract was signed as a result of BBC Worldwide working together with BBC Resources to provide a practical solution for upgrading and modernising studio and transmission systems in order to embrace the digital age. The new service is initially planned for the four major centres of New Delhi, Mumbai, Calcutta and Chennai.

The first phase of the project will examine the business proposition and the technical facilities necessary to provide a viable digital terrestrial transmission service. BBC Resources' expertise in digital transmission will be augmented by the new consultancy business expertise and acumen. BBC Resources will provide Doordarshan with a complete blueprint for the introduction of DTT in India by the end of September 2000.

Doordarshan currently broadcasts in analogue throughout the country reaching approximately 70m homes and over 0.5bn viewers. 'We are delighted to be commissioned by Doordarshan, long established as the broadcast market leader in India,' said David Manning, Head of BBC Resources Consulting & Projects. 'They will be the largest client to date to benefit from our unique experience and expertise in the field of digital terrestrial broadcasting. This new Indian assignment follows hot on the heels of our support for the launch of Botswana Television in Africa and demonstrates the validity of developing our own consultancy practice.'

Mark Young, managing director of EMEIA (Europe, Middle East, India and Africa) for BBC Worldwide added, 'This deal marks a milestone for the BBC. Doordarshan is one of the world's largest broadcasters and tnis is the biggest deal we have done with.them. This is another step forward for both BBC Worldwide and BBC Resources business in India and emphasises the close relationships we are building in this exciting territory.'
BBC Resources: +44 20 8576 7860.

US: Sound Kitchen's Big Boy room is home to the world's largest API Legacy Plus console. Standing behind the new 80-input console in Franklin County are: (L-R) Jennifer Rose, General Manager- Sound Kitchen; Joe Martinson, Martinsound; Dino Elafante, Sound Kitchen; Paul Wolff, Director of Engineering-API; Larry Droppa, President-ATI Group; and Dan Zimbelman, Director of Console Sales-API.

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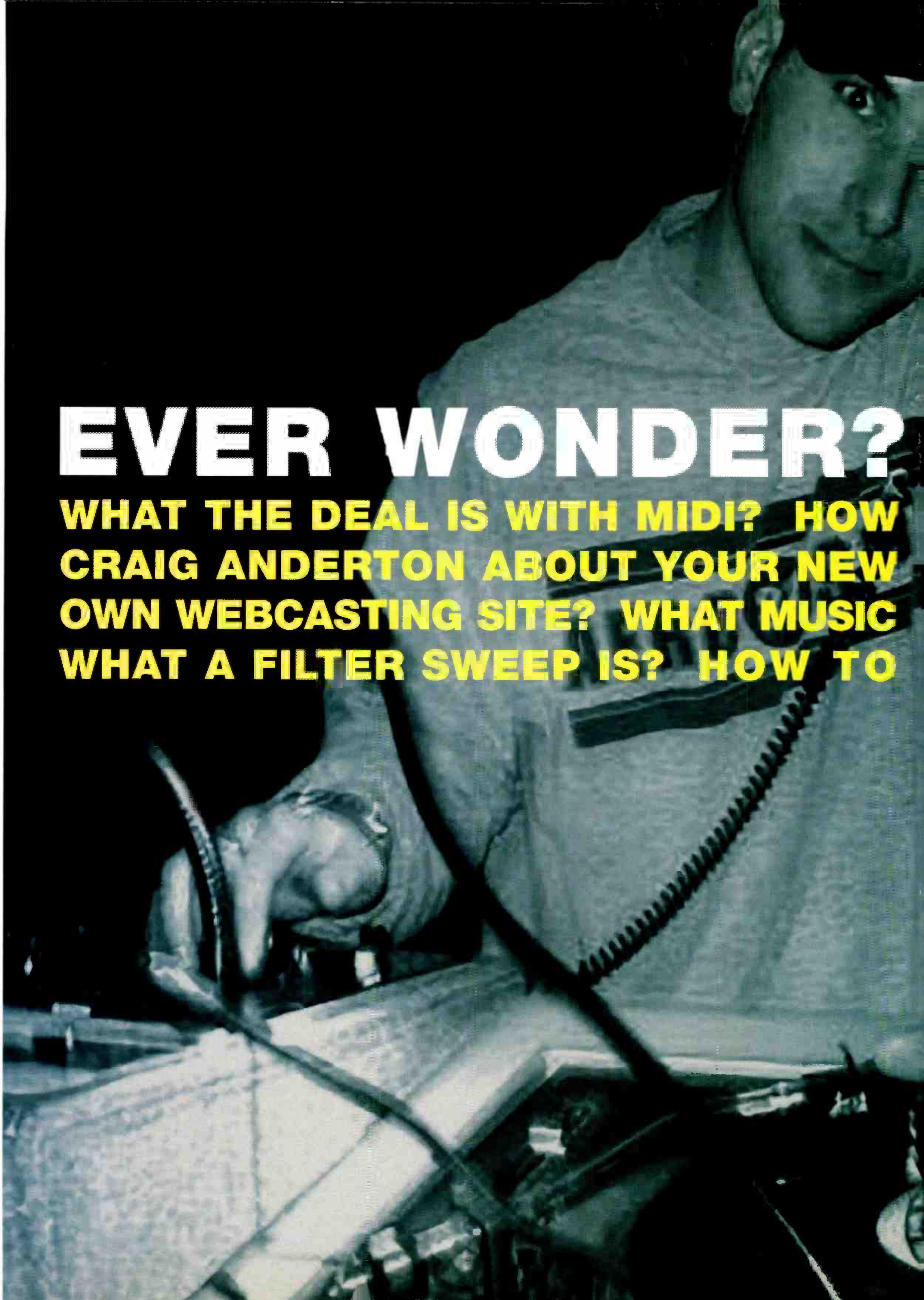

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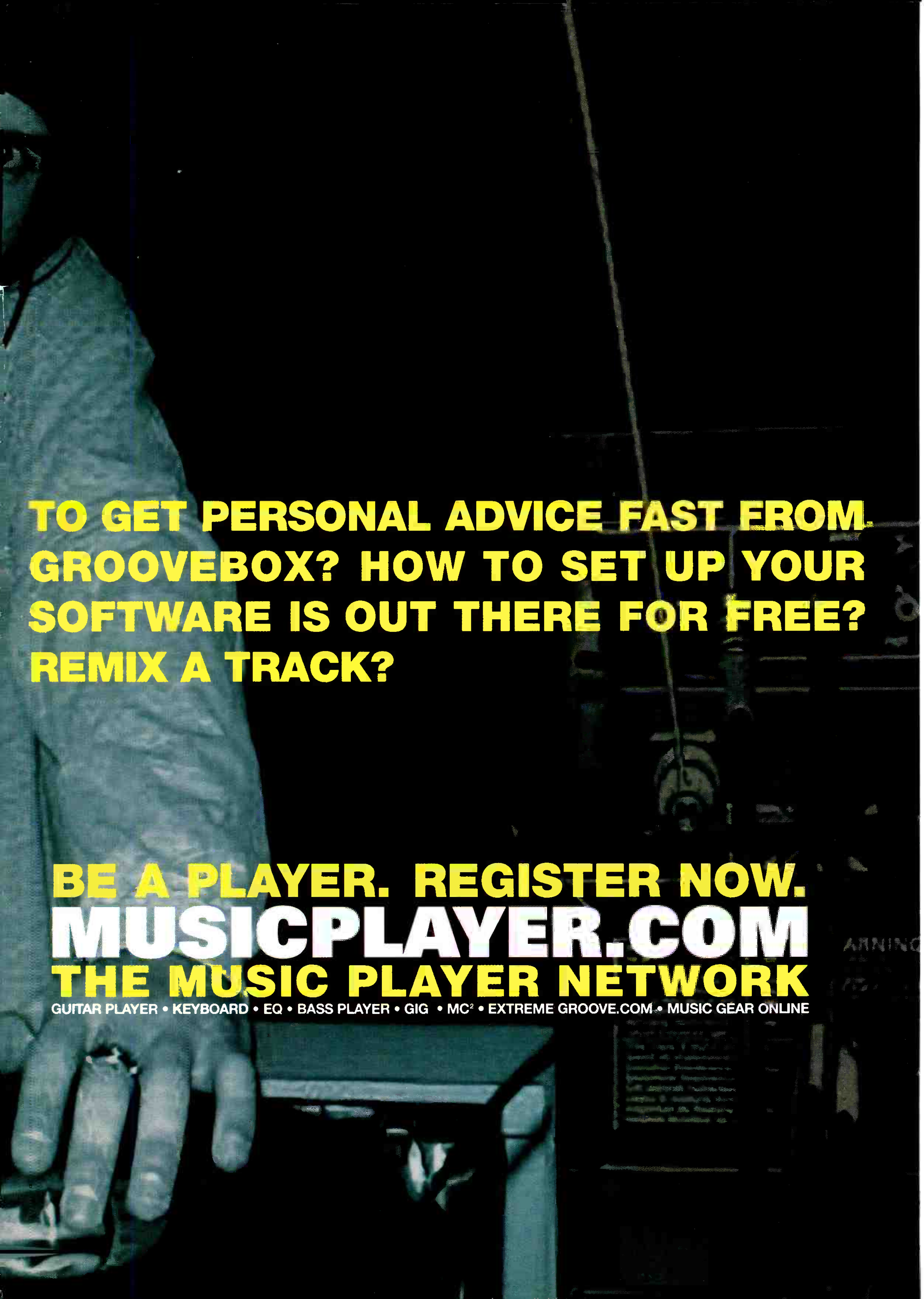

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SOUNDTRACK

One of Spain's premier post houses, Soundtrack is in hot pursuit of bigger rooms better audio and bigger clients. **Tim Goodyer** takes language lessons and visits the first Hidley-Newell collaboration in 22 years

AFTER LAYING THE FOUNDATIONS of its reputation in the mid eighties, Barcelona's Soundtrack has moved inexorably to the fore of the city's postproduction scene. The pursuit of high-quality audio begun by Josep Ferrer in 1983 was paired with Francesc Castillo's technical vision when the studios moved to new premises in the early nineties. Reopening in 1995 with a groundbreaking complement of 11 AMS AudioFiles, two Logic 3s and a Logic 2, it commissioned a flagship cinema room equipped with an AMS Neve DFC a few weeks ago, itself the result of the first collaboration between Tom Hidley and Philip Newell since they built London's Town House studios in 1978. Today Soundtrack is a busy network of rooms tailored to suit its thriving national business and ready to take it into the international arena.

'I worked for three years in a publicity company and then for three years in TV3 where I learned a lot about digital sound and new technologies,' explains the friendly Castillo. 'In 1992 Soundtrack offered me the new project, but I insisted that we change from analogue to digital, tapeless. This was new to Spain —nobody was using it and nobody really knew much about it. I found three options—Solid State with Screensound and Omnimix, Fairlight editors with Euphonix consoles, and AMS who proposed Logic 3 and Logic 2 systems with the AudioFile.

'The problem at this point was that once a recording was finished in one room I needed a simple way of moving it to another. I proposed to AMS and Solid State that they divide their systems hard drive into two—one small drive for booting the systems and the other one for sound. It was not my invention because I had seen this way of working at an AES show in New York where someone had a drive that would hold 15 minutes of audio, enough for three or four advertisements. It seemed very useful so I took the idea to Solid State and they insisted on using Soundnet, but I didn't believe that it could work in my company. AMS started to test removable drives and in summer of 1994 they tested a system at the BBC and at the end of the year they had our system ready. One of the AudioFiles was a dedicated backup station because at that time there wasn't a special system for backing up and restoring.

'We opened with ten working studios, one was the cinema theatre with the Logic 2 designed by Tom Hidley; two more were video and TV mixing studios with Logic 3s, and we had six rooms for dialogue recording; and a studio for editing music and effects but these could all be multipurpose rooms. After working with AMS technology I can tell you that it is very expensive and it's not the most flexible, but it is the only equipment that is usable in the rooms we have with over 20 different engineers. After 5,000 hours of dubbing, we have lost only one single project and that was through a human mistake. I don't want to be critical of systems like Pro Tools, but in Soundtrack I think this is the best technology.

'I think the technology we have helps a lot—we can give good sound quality and we can correct problems like when you have some incorrect dialogue, because the automatic mixing is non-destructive, it is easy to replace it. When we had the old technology we used only to keep the final mixes and they were in all different formats—video cassette, 2-track, ¼-inch, and so on. We couldn't keep all the recordings because they were on 2-inch tape and it was too expensive. Now with the AMS system we have all the recordings on Exabyte tape. We have everything from 1995 onwards.'

Soundtrack took its established clients to its new location, and the move coincided with the main Catalonian television station reallocating its work.

'TV3 held a competition between the studios in Barcelona to determine how many hours of work each studio would get,' Castillo confirms. 'The best studio would get 20% of the work, the second one 16%, the third one 14%, and so on. With the work we had done before and the new facilities, they gave us the first contract. Later, in 1996, the same thing happened, and in 1997, 1998... We have held the contract now for the last five years.

Soundtrack's technical manager, Francesc Castillo

'The population here is 35m–36m and there are 6m people in Catalonia, so we are one fifth of the audience in Spain. The biggest audience is for TV-3 in Catalonia. When Barcelona play football 2m–3m watch on television while other television series have perhaps 1m viewers. Also people from other parts of Spain watch TV-3 because its on the digital network and it's also broadcast in Valencia because their language, Valenciano, is similar to Catalan. Also in Majorca and the Baleares Island the languages have the same root as Catalan. The potential audience is perhaps 8m people.'

At the same time as moving house, Soundtrack moved into cinema work. 'It was very hard to begin with, Castillo comments. 'There were two traditional studios in Barcelona—Sonoblok and Voz de Espana and what we were doing was new. Instead of using different machines where all the recordings sounded different, and having different people for projection, sound loops, film loops, recording without the flexibility to easily check back if there was an earlier problem, we tried to use the best of the video technology in film. So we bought an Albrecht high-speed projector and synced it to a Studer A827 with Dolby SR noise reduction and it seemed incredible to the directors and the actors. Some directors were reluctant to work with something that wasn't traditional, but we persevered and finally, the rest of the studios changed their way of working to follow us.'

At the time of building the new facility, a local actors strike made finance difficult and Castillo is sure that the rooms could have been better, but remains happy with the acoustic. Particularly so after marrying it with Neumann TLM103 mics.

'We tested a lot of mics and monitors in the studios and found that the Neumann U87 is very good for video and TV productions,' Castillo recalls, 'but I was particularly impressed with the B&K 4011. I have never tested a mic like it. An actor can move all around the microphone without the sound changing. The problem is that it has a small diaphragm and more noise. For some cinema recordings the actors may whisper a metre away from the microphone and there is too much noise. Now the studio is using the Neumann TLM 103 and at 60cm or 70cm the response is still good. Good, clean, powerful sound with no noise.'

The original Hidley cinema room came about after an unhappy experience with a Spanish acoustician left the job unfinished. The introduction came via Dolby Labs finding Hidley making occasional trips to Europe from his home in the Cayman Islands.

'He wouldn't come to Barcelona, only to Lausanne,' Castillo recalls of his meeting with Hidley. He was very interested because he had never done anything in Barcelona and this was at the time of the Olympics so it had a very high international profile. He quickly agreed to present some ideas and we ended up lowering the ground by 1.5m and remounting the acoustic treatment and we got a very good room. And he never visited Barcelona...'

More recently, the proposal for a new cinema room saw Castillo track Hidley down again—this time in Florida. 'I originally tried to do the project with Philip Newell because I have read his work in *Studio Sound*, but Josep Ferrer insisted we also use Tom again,'

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Above: the AMS Neve DFC in Soundtrack's flagship room.
Below centre: the 'continuous' projector will hold 14,000ft of film

Castillo explains. He was very happy to do another room for Soundtrack because after doing the first one, he had been hearing all around the world that there was a very good-sounding room in Barcelona! In fact, the competing Spanish companies know ours is the best sounding room in the country. I think he is proud of it.'

'We lost contact for a while after The Town House,' says Newell of his collaboration with Hidley. 'Both of us got out of studio design for a couple of years between about 1980–83. Then, when Tom was doing Masterphonics, I flew to Paris to meet him and we came together to sponsor Luis Soares on one-tenth scale modelling studies because I was looking for how to get things into smaller spaces and he was working trying to get frequencies down on his infrasonic traps. But the Barcelona job was the first time we co-operated on a project.'

The resultant room has five of Newell's Reflexion Arts loudspeakers and McCauley loaded subs across the front with 16 JBLs divided into left surround, right surround and extended surround around the room. It is the first studio in Spain capable of handling mixes in SDDS and Dolby 6.1.

'I was very impressed by the honesty of the people involved in Reflexion Arts,' Castillo says of the search for an alternative to Hidley's speaker designs. 'And I was interested especially in the quality of the amplifiers.'

'Tom had done the room design, but the studio couldn't afford his monitoring,' Newell continues, 'so he recommended the Reflexion Arts as an alternative. Tom would have used slightly different monitors and I would have used a slightly different room design. But on the other hand, I haven't got any complaints about what Tom's done and the owners seem to be over the moon with it, so obviously the two things have gone together. Although the details of the designs are different the philosophies almost identical, so we're going for the same thing—that is, the most neutral monitoring possible, not making any compromise as to what something is going to sound like in a real-world situation. First of all we both believe that you've got to get the recording right, then you can control it in the playback rooms. If it's wrong in the recording, you've got no foundation to work from. Neither of us agree this thing about having to have a certain amount of reverberation because of the rooms in which you're going to listen.'

And Hidley has still yet to set foot in Barcelona...

When it came to selecting a console, the choice of an AMS Neve would seem to have been a foregone conclusion.

'I tested other new consoles,' Castillo contradicts. 'Consoles from Solid State, Otari, Harrison, Euphonix—but the DFC was the best solution. For me, the dynamics and the equalisation on the DFC are the best in the world. This is very important for cinema, especially for dialogue dubbing. When we record we never use any dynamic processing, we record direct from the microphone preamplifier to the AudioFile because we can do whatever we need with the console.'

The new room offers another unique feature in the form of a projection system capable of showing a full-length feature film with almost no disruption between reels.

'One of the complaints of the clients was that it took three or four minutes to change reels during a film,' Castillo explains. 'I looked around for a solution, but I couldn't find anything, so I spoke to a small company in Barcelona called Sistemas Profesionales de Proyeccion, and Salvadore Blanch designed a simple and cheap system for projecting a complete high-speed film that is now called the Nova continuous projector. It will handle more or less 14,000 feet of film and it will go forwards and backwards at any speed you want. At the moment we're just doing 2x normal speed, but the ballistics will handle up to 16x

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Behind-the-screen detail of the Reflexion Arts monitors in Soundtrack's Hidley-Newell cinema room

normal. It means that you can control the complete film with only 30 seconds to resynchronise the sound between reels because the time code is not continuous. We could correct this, but at the moment, the clients are not concerned about 30 seconds. It's when you have to stop, turn on the lights and the continuity of the film is lost.'

Future plans involve the building of a big dialogue recording room for cinema and a big Foley recording room to meet the absence of a large specialised Foley room anywhere in Spain to match the rooms found in Paris and London. Before this, however, the search is on to dramatically expand the client base. Castillo reckons that there are around 200 cinema films to be dubbed in Spanish every year and about 5,000 hours each year of TV films and series across all the television channels.

'We dub more TV into Catalan because we have the Catalan stations here,' he says of the local language. 'There are only perhaps 20 films every year dubbed into Catalan, but the head offices of the TV stations are in Madrid so it's easier for them to use the studios there for Castillian language dubs. In Galicia there are some good studios, but they don't have good access to dubbing actors. There are also studios in the Basque country, but they have the same problem and there are studios in Valencia where they have their own local, TV-9, and also in the south of Spain. But about 98% of the work is done in Barcelona and Madrid. About 80% of cinema is made in Barcelona—Castillian, Catalan, it doesn't matter—because we have the most famous actors here.'

But the future is bigger than the Spanish market. 'Next week Josep Ferrer is going to Los Angeles to meet some of the big Hollywood companies,' comments Castillo. 'In the past we have worked with Dreamworks, but now I think we are ready to make some important relationships with companies like Fox and UIP. At least, we are ready to try to earn these clients.

'We are working now with TV-3, Television Espana TVE-1 and TVE-2 and the main independent Spanish distributor of low-end films. We have also worked with Warner and Fox. Now Buena Vista is starting to work with us. It has been very hard work, but now all the majors are with us. The boss didn't want us to contact the Disney companies before we had the big room and I think he was right because they are impressed by its size and also the sound quality.

'What we are trying to do is to put a good root to grow in a good way,' he elaborates. 'The idea is to work slowly, we don't want to be in a rush. We are trying to take a good position in the Spanish market. We think we can get a bigger market share than we have especially in new cinema productions and film dubbing. In the future it might be good to be a centre of mixing and dubbing of different languages in the Mediterranean region—Portugal, Spain, France, Italy, perhaps Greece—because we have good people and we have good technology. We are in a good position in Barcelona because we are near the airport, the beaches are only 15 minutes away... We need to work as well as we possibly can and our clients must be happy with our work but Barcelona is also a good place to spend some time. We have the sun and the sea in summer and in winter the weather is mild and we have good skiing less than two hours away. We are lucky to be here. □

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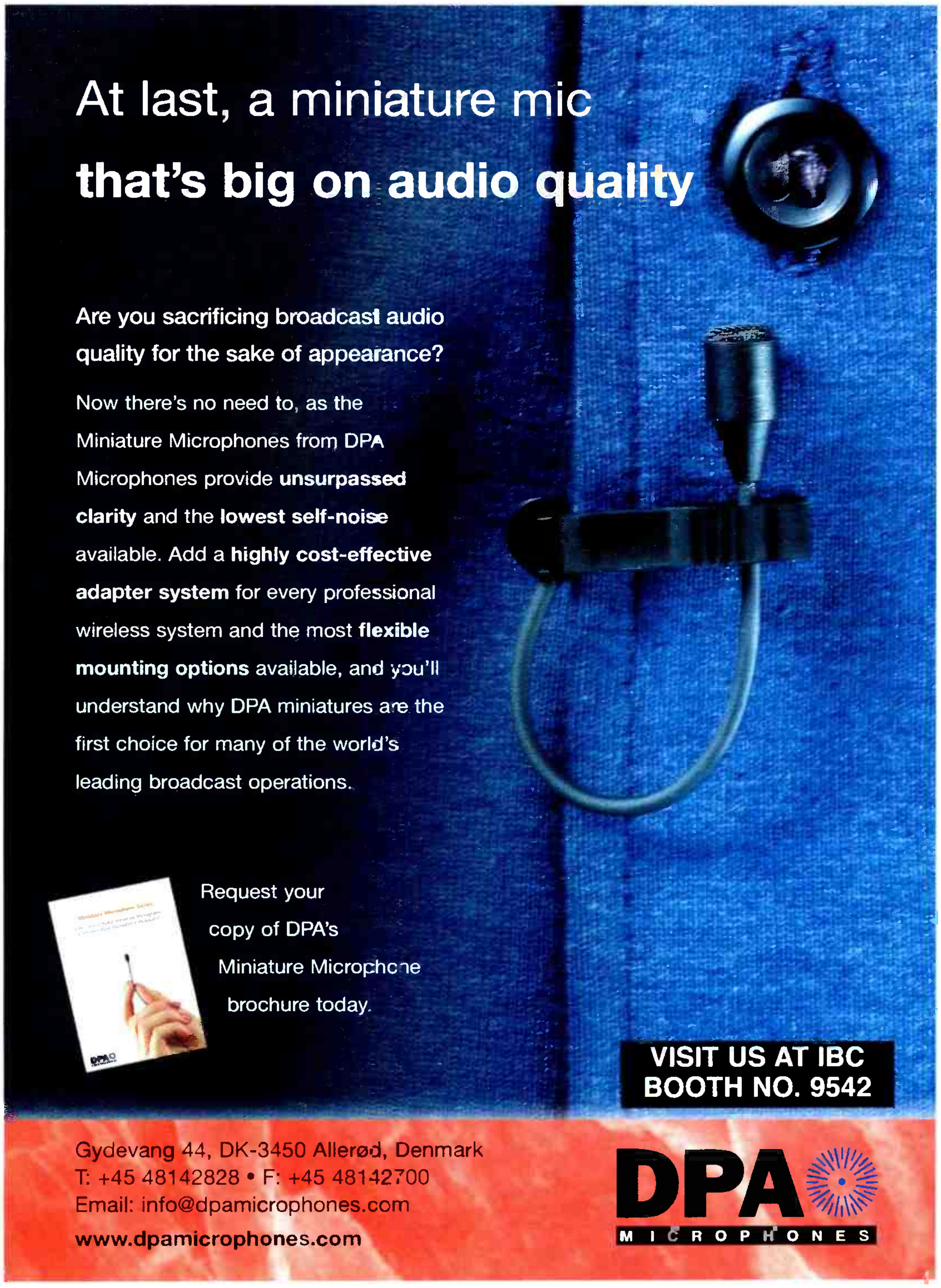

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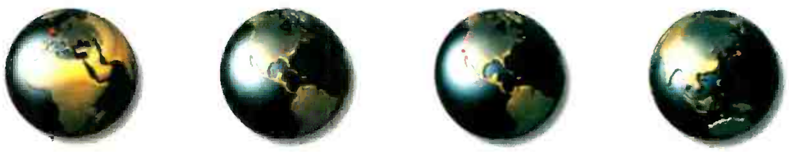

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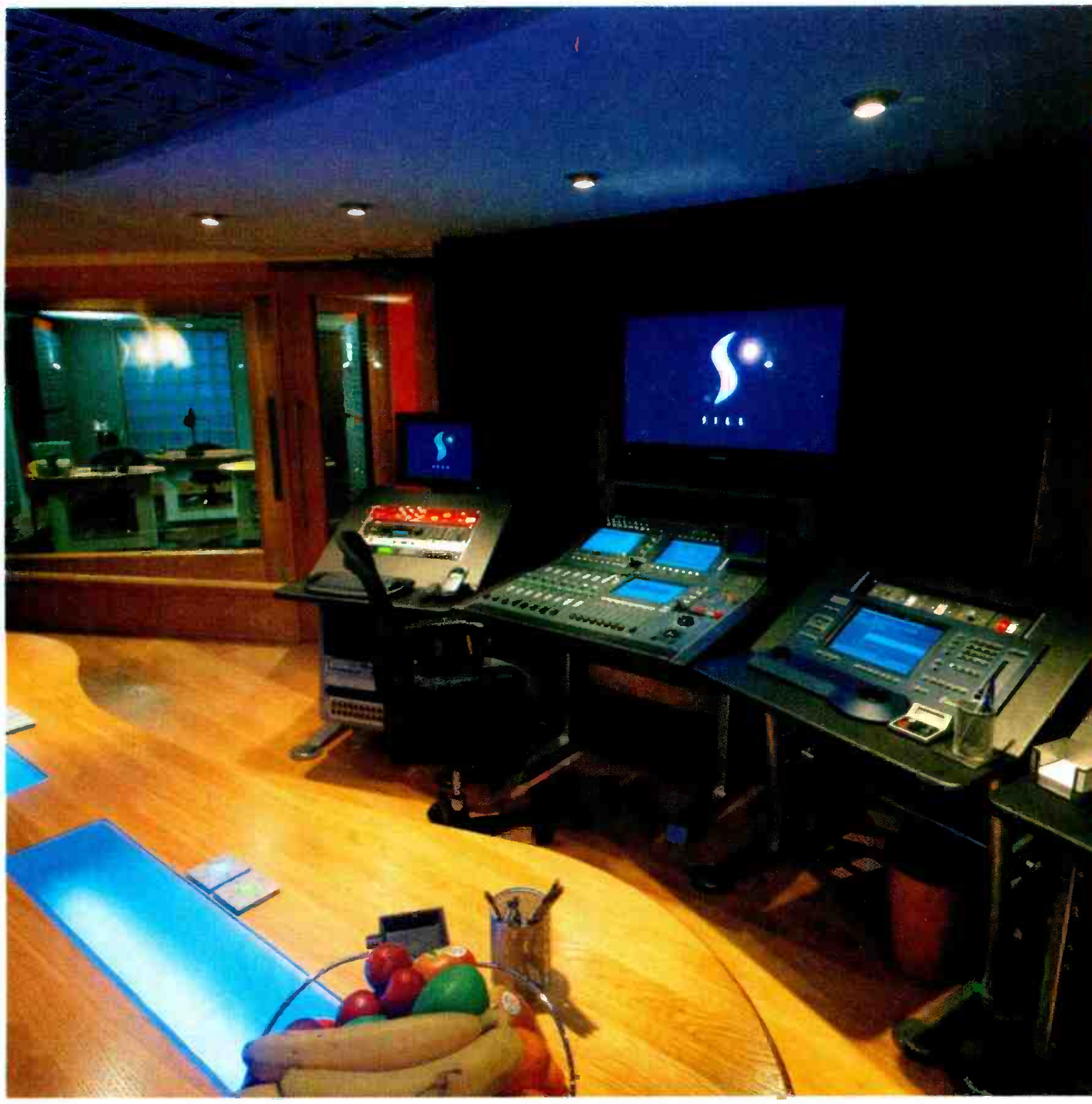

SILK SOUND

In the heart of London's post community, the capital's most famous Lexicon Opus user is refitting its four studios and the transfer bay. **Zenon Schoepe** reports on the decisions made

ONE OF THE MOST INFLUENTIAL operations on the Soho postproduction scene, Silk Sound has relished being different. With bulk purchases of the then fledgling and promising Lexicon Opus some ten years ago it became and remained an island of the technology in a sea of other brand solutions. Opuses were also installed at the holding company White Lightning's other facility The Bridge and between them they now own every Opus that was ever sold in the UK.

But it has been so for a decade and while support for the system from Lexicon officially ended in July, Silk has long been planning the refurbishment in Berwick Street.

What it has opted for in its four studios and transfer room is four DAR Storms, a couple of OMR8s and a four Soundtracs DPC-II consoles as part of an order for seven—the Silk project should serve as the model for plans for The Bridge as the next complex to be refurbished. The third facility in the White Lightning group, Space, remains an all-SSL OmniMix complex. Emotional attachment to the Lexicon systems is clearly still strong.

'It's been getting to the point that very few people at Lexicon now know what an Opus is,' says MD Robbie Weston. 'I phoned them a few months back and they thought I was talking about the Lexicon Studio PC package,' adds technical director Rick Dzendzera.

'Parts haven't been a problem because the things are built like battleships and they've been incredibly reliable,' observes Weston who adds that the systems worked out at an average cost of around £175,000. While hugely expensive for the time, he's not complaining as he's more than earned his money back on them.

The refurb has been an opportunity to start afresh at Silk and core to it all is a giant Lighthouse Digital matrix which runs 100 x 100 in analogue and in digital with machine control, time code, and video networked across the complex The refurbishment goes right down to the wiring.

'The Opus hasn't been developed for years and each time we do something new we think about what we are trying to achieve and then find the equipment that gets around it,' explains Weston. 'This installation was designed to keep everything on big servers. We didn't want removable drives and we wanted to eliminate DATs as quickly as possible. Everything had to be networked and be able to talk to everything else and we wanted to create a sort of virtual "job bag" booking system in to which you could put every element, so you could say "That's the audio, that's the mix, those are the sound effects that we used and there's the invoice." '

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Aside from the well-known DAR networking technology, Silk is the first installation for Soundtracs' net.tracs networking system for Silk's 16-fader DPC-II digital desks—'they're not very wide, but they're very deep' quips Weston. 'You could take them to 160 channels although I don't know where we'd find 160 inputs,' adds Dzendzera.

Net.tracs permits the interchange of project information between consoles in a multiroom facility and to provide a convenient facility-wide archival and restoration system. A 19-inch rackmount server controller connects via CAT5 ethernet and deals in Soundtracs session files for relocation and restoration.

'The DPC-II is a wonderful console and the best we've come across. It's relatively affordable, it's well thought out logistically, they haven't tried to reinvent things,' states Dzendzera. 'Soundtracs has been fortunate in coming in to digital desks late, watching everybody else's mistakes, coming from a mass manufacturing background and making something and getting it right,' continues Weston. 'We did think about SSL again, but what ruled them out before we even started talking about prices and facilities was that their desks are so huge. And a Silk room is a small room.'

'We like to be different,' he continues. 'A lot of people seek comfort in having what everybody else has and we like to think that there is always another way of doing it. We've been fortunate with this project that we were at a stage where we could actually ask for things we wanted and influence a product.'

Lessons have been learned from the last major build they did—the ground-up Space complex—particularly in terms of what they did right. 'The fact we used the central router worked really well and gave me the confidence to go ahead and do the same again,' says Dzendzera. 'The Lighthouse Digital has complete belt and braces, dual power supplies, backup processors and we've had it working for several months already. All the work in transfer is via the router and with the first studio on line that's going through it too.'

Studios are being completed and switched over at the rate of one a month until September after which transfer, currently running the old and new infrastructures simultaneously, will be converted.

'The Opus worked efficiently and quickly especially when you're trying to find things in it—the navigation,' he adds. 'A lot of the systems we looked at didn't come anywhere near it. To some extent the Storm did when we first looked at it but they understood what we wanted and were willing to incorporate our ideas.' Weston says that one of the advantages of DAR being a Harman company meant there were no trade secrets involved in handing it an Opus operating handbook with the good bits highlighted. They also sat DAR in front of one and pointed out the things they liked and the things that it couldn't do.

'We are not a postproduction facility we are a recording studio,' states Weston. 'Everything we do starts with recording something and that really makes a difference to how we view equipment. There are places that just put together items that were recorded somewhere else and they're concerned with autoconforming and time-code locking to this and that. It's irrelevant to what we do. We are a recording studio and it's important that we separate ourselves from the room at the end of the corridor in the video facility.'

They praise the openness of DAR which contrasted to the attitude of other manufacturers who failed to respond to questions or by modifying software. 'DAR's D-Net is the most advanced networking system from any of the DAW manufacturers that is actually working and can be delivered,' states Dzendzera. Weston highlights the sound effects handling of the system for special mention as it avoids cumbersome combination systems. 'You can describe the effect, audition in place, scrub in place, without having to transfer anything. It's in and it works and we have around 200Gb of storage just for sound effects. CDAdvance was the other thing that impressed us and that is just so brilliant,' says Dzendzera.

Weston calculates that he's paid around £40,000 less on the editor-mixer combinations than he did for his Opuses, but adds that he's also getting much more for that money. 'But that's a long time ago now and you would expect the technology to have got cheaper,' he says. 'Some of our corporate clients send us letters saying that "in keeping with our determination to drive down costs 15% per annum please present your pricing proposals." We ignore it. With a facility, how are you going to drive the price down? Prices are determined by interest rates, rents, rates and salaries. Interest rates may have gone down slightly, rents in Soho have gone through the roof—they've more than doubled in the last five years—Westminster council is not known for cutting the rates, and if you want highly skilled editors and engineers you have to pay for them.

'If we said "Work for Silk Sound, we have a commitment to driving down wages 15% a year" we're not going to get many good people. I can't understand how anyone can think that our services can become cheaper. Instead there are lots of enthusiastic chaps who haven't quite caught on to the realities of life—of mortgages, getting married, having children, and living a bit—who work for nothing in all these funny little rooms. There are people running business with two small rooms earning way less than they could earn working for someone else. Why do they do it? The only way you can make it cheaper is to do it for less.'

Silk's acoustic design was by Recording Architecture with Miller & Kreisel monitoring throughout, 5.1 in the bigger rooms. They have also taken up the option on the building next door with incomplete plans to build 'a very special studio' there.

And what of the fate of the Opuses? 'We'll put them in storage, see if anyone wants them, keep the spare parts,' answers Weston. 'It's sad but they've earned their keep now. The reality is that unless you work the way we do you can buy a Lexicon card that plugs in to your PC now that appears to do far more for a fraction of the cost and without all the difficulty of administering a rack the size of a phone box.'

'There is an enormous difference between the type of system we need for what we do and a PC or Mac based system,' he continues. 'If somebody giving you a load of elements and you have to produce a fantastic mix a week on Thursday and you're not doing it under any great pressure, then these systems can work. But for the guy who is under pressure to perform with his clients sitting around him, grumpy actors in the booth, it is a nightmare using do-it-all PC systems.' □

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Amek Media 51

It aims to bring a package of real multichannel capability with an affordable and automated analogue console with a big desk feel. **Zenon Schoepe** meets the successor to Big

A LONG TIME AGO, IN A GALAXY FAR away, when the talk and prospects of console design was centred on the promises of digitally controlled analogue and the idea of affordable digital desks was as far fetched as desktop email and Internet access, Amek hoped to supersede the flagship of its APC1000 digitally controlled analogue desk. Known only as the 'Media Console', strips were shown at shows and explained to potential customers who marvelled at the pot and switch density but glazed over when they were told that it would work in all formats known to man and those that as yet weren't. Here was the future they were told yet the price, size, ambition and a user base solidly embroiled in stereo proved otherwise and it never came to pass.

The best part of a decade later, part of the name has resurfaced and with it many of those original sentiments but it's interesting that the Media 51 should be so affordable. But then it is derived from the Big by Langley which it effectively replaces and things have moved on. For those who have forgotten, Big was revolutionary in pricing terms by offering a high density of I-Os in a compact frame that included Supertrue automation, Virtual Dynamics and Recall and the new console serves to continue this tradition.

Key things that need to be noted are that the motherboard system is exactly the same as Big so it's tried and tested but it has useful additions up top of a Rupert Neve mic amp and EQ. Big was becoming a little dated in this respect but, perhaps most significantly, it includes 5.1 capability to at last make these facilities available at a lower end than was previously available on a real analogue desk.

In terms of available channel numbers, they are the same as the Big. Minimum size is 28 inputs in a 44 frame with a 60-channel frame being the biggest size likely. In terms of automation it offers the same as Big in VCA faders, the aux mute and input mutes with recall available across all controls and switches.

Auxes numbers stand at eight arranged in pairs on dual concentrics and available from either path with four switchable pre-post.

Routeing to the 12 buses is accessed on individual switches and can be sourced from the mix or channel paths as well as from the output of Auxes 3&4. The channel path section contains the gain controls with phantom power, mic-line selector, level and AFL while the Mix path gets a gain with tape selector, phase reverse, input flip and an insert that can be flicked to the channel path.

Following the EQ, which lives in the mix path but can split the HF and LF to the channel path, we're in to the stereo and front-rear pan controls, automation select switches, AFL and mute.

Stereo return modules sport a stereo effects return and a stereo line input and four come with the desk as standard. These are, with cosmetic changes, identical to those on the Big. These get access to all the auxes and a more rudimentary 4-band stereo EQ but still have enough features to make them integrate well with the rest of the desk and offer a convenient means of bringing stereo submixes in to the board.

As befits its title, it is when we get to the panning that things get interesting although it is worth pointing out that you can ignore the multichannel abilities of this desk very easily and drive it as pure stereo board without any hardship or inconvenience. At any point you can steal Buses 1–6 either as LCRS buses

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with a press of the LCRS button or 5.1 buses by also pressing the SS(stereo surround) button in which case bus 4 becomes the LFE channel and is sourced post fader. A corresponding WIDE (divergence) button next to the Mix pan permits discrete panning and a front-rear pan does exactly what it says on the label. The remaining multitrack buses are free to be sourced from the aux sends or the channel paths.

An interesting twist is afforded by the 1:1 switch. Playing back a 5.1 mix and reassigning it to Buses 7–12 and pressing the 1:1 switch provides a means of transferring the inputs without any gain control to the buses. And before anyone wonders PEC/direct fashion switching is supported.

Much of the Monitor module also shares functionality with the Big but moves in to a completely different games level when it comes to multiformat use. The monitoring section is actually extremely comprehensive although its arrangement, presentation and the relatively small amount of real estate that it takes up on the board means it doesn't look it. The pertinent monitoring functions may yet emerge as a stand-alone bolt-on product for other desks.

This monitor module handles all the traditional things like mix fader assignment, metering, aux masters (with blend—an Amek invention), bus trim tweakers, oscillator, talkback and solo control just as the Big did before it. Although the desk is tagged up for 5.1 work the six buses and the main stereo can also be combined for 7.1 operation—perhaps a little far fetched given price and likely target users but it is possible anyway—and this is bolstered by the existence of a switchable 8-wide insert and the pressing into service of the LR bargraphs to follow Lc and Rc when operating in this manner. Metering source selection is performed by buttons that activate input, playback, direct and the speaker outputs.

Things get clever around the monitor source selection section of the panel which permits the selection of five stereo and five wide-format sources and it's here that you'll also find master PLAYBACK, DIRECT and DIRECT-PLAYBACK switches. Dedicated switches are also provided for encoder and decoder inserts.

Higher up the chain of command are mode switches that add functions to the speaker mute and format switches. It's here that you can define solos for playback-direct monitors and the speakers, mute each playback-direct monitor and mute inputs to the monitor system after the decoder insert and before the metering, and realise individual direct-playback switching for the six monitor input paths. These work in conjunction with two rows of six speaker format switches which follow the aforementioned modes and where monitoring or creation of downmixes is performed to mono, stereo, LCRS, 5.1 and 7.1 the last introducing an additional pair of close-fields. Speaker levels can be preset or calibrated (bypasses monitor VCAs and allows calibration on trimmers) and recalled on switches. An 8-wide joystick will also be available as an option.

There is a good logic to this section in the way that the solos and speaker mutes work, for example, throughout the various modes. As already said it is all here in an extremely comprehensive section although the layout is not ideal. The layout is dictated by the width of the monitor panel rather than any overriding ergonomic considerations, consequently everything is crammed in to where there is space as opposed to where it might like to be found in a more ideal world. Many Media 51 users will be new to the idea of working in multichannel and will not understand the history and origins of multichannel operation, why things are presented in the manner that they are and why there are

The Rupert Neve Effect

THE DESIGNER'S INPUT has been limited in the Media 51 to the mic preamp and the EQ section of the desk, the former being a new padless arrangement offering up to 66dB (-10dB to +15dB on line). EQ is described in the literature as a refinement of that on the Big but in reality it is a light year away from the original's swept mids and switched frequency HF and LF with a single switched high pass. The new desk has swept frequencies on all bands covering, from the top, 2kHz–20kHz, 500Hz–5kHz, 100Hz–1kHz and 30Hz–300Hz. It's all ±18dB and with 25Hz–500Hz and 2kHz–30kHz filters stuck on it amounts to one mighty section. The HF and LF can additionally be switched individually to peaking characteristics.

I'll admit to being taken back at how modern sounding this EQ is and it's quite a bit more drastic than less recent designs by the man which I would describe as more delicately geared. A twist results in immediate aural action which can be quite dramatic but it's still good powerful stuff. The filters are an enormous bonus on a desk in this price range.

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Supertrue

NOW RUNNING VERSION 3.7, which is long time since the switch to PC from Atari that coincided with the arrival of Big, Supertrue remains a simple and approachable system. It's important to remind that it has the largest installed user-base of any automation system ever. That spans desks of the last decade and is taken through, albeit in a far more automatable bits and pieces form, to the flagship 9098i.

On the Media 51 aside from the faders, automated control extends to channel and mix path mutes, plus their equivalents on the stereo return module and Aux 1&2 mute on the channel strip and Aux 3&4 on the stereo return.

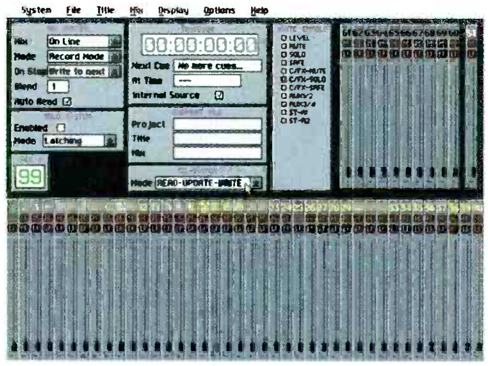
This system has been covered on numerous occasions not least by me although long ago enough for me to have forgotten that the system does not present a horizontally scrolling fader position display as part of its editing portfolio.

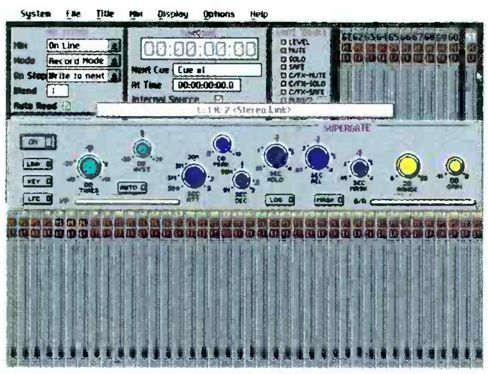
Fundamental operational modes are the familiar read, write, and update selectable from the Select switches on the desk or from the screen. They are augmented by the popular concepts of autotakeover, match and solo—the last of these allowing solos to be written in the traditional way.

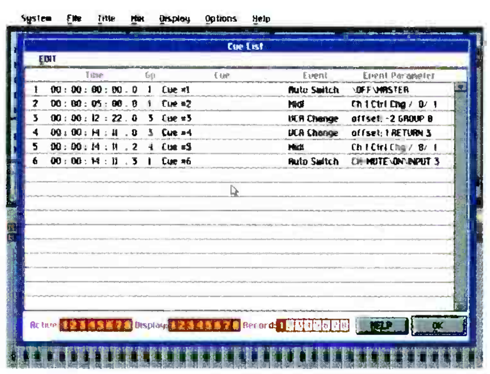
Off-line editing amounts to a mix processing page for the likes of off-line trim, which involves the defining of in and out points for the processing to occur over selected channels. There's also repeat, shift (slip), erase, copy and swap for moving channels around. This process can be applied to definable automatable event elements.

Recall is pretty standard stuff and includes the autosearch function which scans the desk from a given snapshot and identifies controls that have been altered. The screen is clear and simple. Of course there's also Virtual Dynamics which employs the channel VCA as the tool with which to insert dynamics processing in to the signal path. Virtual FX meanwhile serves as a screen front end to control and adjust MIDI equipped outboard gear parameters and presets.

Together this constitutes one hell of an automation package at a time when you can no longer buy an 'affordable' third party automation system anymore. Look at it as buying the desk and getting the automation thrown in for free.







a variety of subtlely different ways of achieving what are to an inexperienced eye very similar functions. The shoe-horned layout may not help matters but anyone who masters the presentation of principles on this desk will be able to transfer those skills further up the technology ladder with ease. Amek has also produced useful supporting literature on multichannel principles which should go some way to speeding the process.

I like the way that operational principles follow through on this desk and the way that the simpler business of stereo working, or in fact simple multichannel working, requires decisions to be made further up the top of the board. You only have to delve in to the monitor section with a vengeance for the really fancy stuff and the 'ordinary' user is to a great extent protected from this.

In line with most potential user's requirements this is a stereo console that you can do 5.1 work on as opposed to a desk that is optimised for multichannel work that you can do stereo work on. It's a subtle distinction but an important one to make. It's probably the correct weighting of the issue and a good reading of demand. It's a desk that is simple to operate on one level with enormous power and capability beneath that is likely to be beyond the requirements of occasional multichannel users but well up to the job for the regular user.

As such it represents by far and away the most sophisticated analogue desk of its price and type and has approached the multichannel aspect in a manner that puts the tokenism of many affordable digital desks to shame. Too frequently we encounter consoles that have multichannel capability almost bolted on as an afterthought, with the Media 51 we are looking at a desk that has struck a modern and extremely accessible balance.

Proper multichannel tools in a proper and well equipped stereo analogue board. The package is irresistible and it only remains to be seen whether Amek's reading of the market is correct and apt. I've got to admit that I think that it probably is.

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Tascam MX-2424

Getting to grips with Tascam's new hard-disk recorder-editor reveals why it may change the recording market. **Rob James** looks at the operational aspects of the machine and courts revolution

HAVING LOOKED at the technicalities of the Tascam MX-2424 last month (*Studio Sound*, August, page 14), it's time to get something of the feel of it, its implications and applications. I have long found the appearance of new equipment in my life to be remarkably similar to embarking on a new relationship. If there is sufficient initial attraction, any perception of obvious defects tends to be submerged in the rush of excitement. This is followed by a period of exploration and discovery while the euphoria slowly wears off. After this, if the balance of advantages and disadvantages is positive, I settle into comfortable familiarity. This will usually last until some catastrophic fault occurs or general unreliability creeps in. Eventually though, a newer model will appear and may be seductive enough to supplant the 'old faithful'. Even then, parting is likely to be protracted. There is always a reluctance to let go of the familiar and safe.

The last time this happened in a big way, it was goodbye to multitrack analogue tape and magnetic film and hello DTRS and ADAT. In the meantime several alternative temptations have appeared, based around hard-disk technology. So far none has really succeeded in supplanting rotary head tape. This time, I think it really has something to worry about.

First impressions of the MX-2424 are promising, transport control feels positive, the quality of scrub audio is more than adequate and, more remarkably, operates across all 24 tracks. I still miss 'chatter' audio at up to 4x speed, but that's probably just my age showing. There are no solos on the machine or Viewnet, but since the MX-2424 will almost invariably be used with a console this is no loss. For the same reason I didn't miss meters on the remote, although the peak LEDs are useful.

Until the TI-SYNC synchroniser arrives, transport sync to other devices is limited to other MX-2424's, TL-bus machines or MIDI. I hope it won't be a long wait. TL-SYNC will add four Sony 9-pin outputs and one input plus SMPTE LTC and DTRS-ADAT sync among other things. The Timeline heritage should ensure it performs well.

There are a few signs of the machine and remote having been built 'to a price'. The rubber keys on both units are particularly 'squidgy'. For the lesser keys this is of little importance, but the transport keys do not inspire confidence. The lack of internal illumination is another minus. This is less forgivable on the RC-2424 remote that is a relatively expensive item. It isn't as if Tascam doesn't have suitable parts in the bin. The transport keys on my DA-60 MkII DAT recorder are shining examples. On the other hand, since most people will use a remote and the Viewnet software, the keys on the front panel could be said to be largely irrelevant. In fact for many applications the MX-2424 could just as well be a black box with nothing but a mains switch on the front.

I was immediately impressed by Viewnet which first appeared for the MMR-8 Digital Dubber. It could be made sexier and some of the ways it goes about things are less than intuitive, but it works well.

Network newbies would be well advised to get their dealer to set up the computer's network address to work with Viewnet. For anyone who has a nodding acquaintance with IP addresses, network masks and Gateway addresses it should be a doddle.

Booting the application takes a minute or so, while the networking sorts itself out. From then on it is robust and co-exists happily with several other applications I tried—such as Vegas Pro, Nuendo and Samplitude. The only real source of frustration is the realisation of how much more Viewnet could do. Waveform editing is slated for later this year so we will have to wait and see what this brings. If Tascam gets this right it will have a serious multitrack editor on its hands. This could provide much more powerful ways of dealing with the problems which arise during mixdowns where moving all the elements to a workstation for a fix would be extremely cumbersome, if not impossible. As it stands the editing facilities are welcome but fairly rudimentary.

At present the networking is used purely for control of the machine or machines. However, 100 Base T is perfectly capable of moving audio about at reasonable speeds, even in real time for a limited number of tracks. The MX-2424's appeal would be greatly enhanced if moving audio between machines was possible and I cannot see any good reason why this shouldn't be implemented in the future.

The convenience and speed of operation of a dedicated hardware solution often greatly outweighs the lack of a lengthy feature list. It is all about designing tools for specific purposes and attention to detail. For example, Tascam has elected to use the slightly more expensive SCSI drives rather than EIDE. Although DMA66 (and recently DMA100) hard drives are now capable of giving good performance in audio and video applications they do not lend themselves well to simple interchange between different manufacturers machines. SCSI drives also still just have the edge in performance and may be moved from the machine to PCs and or Macs with relatively little drama. Where it is necessary to connect multiple drives, this too is more easily accomplished with SCSI.

The backup utility demonstrates the same sort of philosophy. Once a project has been backed up, subsequent backups only record new material and changes. This can save a great deal of valuable time and disk or tape space.

On the editing front, the MX-2424 is adequate rather than spectacular. In my opinion, front panel editing is not really viable for intensive work on any machine seen to date. Editing from the RC-2424 remote is considerably easier, but I don't believe this

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machine is intended to replace a serious editor. A bigger limitation is orphaned cues. In non-destructive mode a cue that is completely overwritten is no longer accessible, although it still exists and uses disk space. The same applies when editing. If a cue is placed over another, completely hiding it, the underlying cue 'disappears'. Moving the surface cue away does not reveal the underlying one. When editing this is less of a problem since the multilevel Undo will restore everything to normal, but I feel there should be a way of getting at the orphans.

The MX-2424 will already prove highly attractive to several groups of users, leaving aside what might or might not be possible in the future. For anyone looking for a direct replacement for linear tape it offers 24 tracks of 24-bit recording at a quite remarkable price point. The audio I-O options are well thought out and the analogue convertors sound fine. More importantly, all the I-O is as reasonably priced as the machine itself. Thanks to this and the overall system design it is not only possible but economically viable to put together systems with huge numbers of tracks. For music recording there is no longer any need to compromise the number of tracks on cost grounds. If you feel the need to record every individual feed from a massive mic rig it won't be the cost of the recorders which prevents you. Even at 96kHz four machines will give 48 tracks. Once FAT32 arrives later this year, record times shouldn't be an issue either. With hard disk costs continuing to fall and low cost backup to recordable DVD, stock costs won't pose a problem. With tape, and especially rotary head machines, the purchase price pales into insignificance beside the maintenance bill. By comparison MX-2424 ownership will be thoroughly reasonable, hard disk recorders are already prized for low maintenance and high reliability and the MX-2424 should be no exception.

Film and TV post took to DTRS in a big way. Once the synchroniser appears the MX-2424 will be very much at home. There is no biphase, but this is not a show stopper in the vast majority of applications. Lack of broad file compatibility might be seen as a problem, but in many current production process models the SDII and .WAV capabilities will be sufficient. As Tascam's Open TL (Open Track List) EDL format is taken up by workstation manufacturers, moving projects will become easier still. Steinberg and SADiE are already signed up.

I can also see the machine finding applications in theme parks and similar installations which need a reliable and compact source of multichannel audio.

All this would count for little if the feel of the machine didn't appeal. Again, Tascam has paid attention to the details that count. The transport dynamics, jog-shuttle and ramping of rewind and fast forward all contribute to a sense of comfort. The destructive tape mode is ideal for acquisition since it produces single files for each track and the running time is a known quantity. With any machine of this type there are going to be a lot of setup menus that can initially seem a little daunting. I did read the manual pretty assiduously, but I didn't have to keep going back to it. Familiarity grows quickly and you soon forget about the machine and concentrate on the mix —just as it should be.

While perfectly capable as a stand-alone machine, the MX-2424 really comes into its own when used with Viewnet. This makes setting up the machine and editing far easier and clearer. The moving track display together with the overview gives the operator a great deal of information without confusion. Viewnet is even more useful when multiple machines are involved. Since I only had one machine I couldn't try this in anger, but a number of highly useful things should be possible. For example, to facilitate quick changes, machine setups can be saved and reloaded to many machines in one hit. Editing operations may be applied to selected tracks on several machines at one time. This will prove invaluable for making global changes to big projects.

Tascam will not have this market to itself for long. There may well be other manufacturers waiting in the wings and Mackie, in particular, has been promising their own offering, the HDR24/96, for some time now. Meanwhile Tascam has come up with a highly significant machine. I think with a bit of luck it should set the agenda until the day when the whole paradigm shifts again.

The quality of recordings is equal to those made on machines at several times the cost. There are minor compromises such as the rubber keys, but these in no way impinge on the core functionality and scalability of the system. Most importantly, the MX-2424 feels good in use. If you really need more bells and whistles it is going to cost a great deal to get them. I think for many applications the limitations are trivial and I for one would rather have 96 sample accurate tracks than 24 for the same money.

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Waves L2 Ultramaximizer

Breaking out of the virtual and into the physical domain, Waves digital Ultramaximiser is an impressive piece of hardware. **Dave Foister** gets hands on

THE VIRTUAL STUDIO has still not taken over the world. Many would question whether it ever will or should, and the biggest factor fighting it is the fact that we love hardware. Working on a computer screen may be space efficient and cost-effective, but it's never a substitute for grabbing real controls and adjusting them, as shown by the hardware interfaces that always follow in the wake of the latest DAW. Of course, while the existing hardware manufacturers have all been keen to get plug-in versions of their products into the software domain, there have been newer companies only producing software who have perhaps realised that there's a whole chunk of the market they're missing.

However good their processors are, there are many studios who will never buy them unless they come in a box. This is why we have, as an example, the Antares Auto-Tune in a box, and now it's joined by the Waves L2 Ultramaximizer.

Waves has been doing plug-ins as long as anybody who still has their faculties can remember, and the standard package of dynamics, EQ and other tools and effects is to be found in almost every workstation. A particularly powerful element is the L1 limiter, a mastering processor that does a little more than just limit, and it's this component that has found its way into a 2U-high rackmount box with enough flashing lights to satisfy the most demanding hardware die-hard.

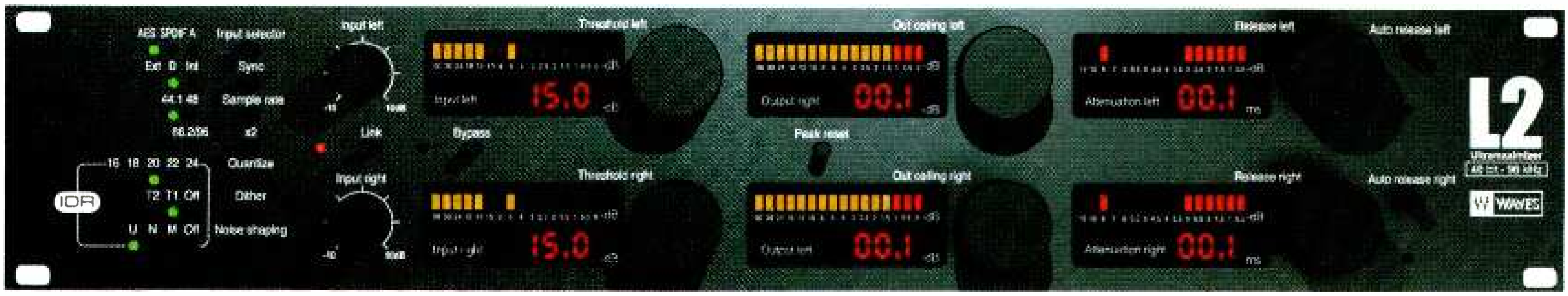
If the appearance of the L2 looks a little extravagant for what's basically a digital brickwall limiter, that's because of the extra bits and pieces that have uses of their own even when the limiter is not in use. Admittedly its functions could have been confined to one display screen, some softkeys and a single encoder, but if you're going to turn a software processor into a physical metal box you might as well eat the whole hog. For this reason there are no less than six main controls, each with its own meter and numeric display. For stereo mastering use, only the top controls do anything, but it's as well to know that the L2 is equally happy with two independent mono signals.

Despite appearances, operation of the L2 is kept extremely simple. For most uses only one of the parameters is important—the Threshold of the limiter. This is adjusted in 0.1dB increments for the benefit of the fastidious, and compensating gain is automatically added. Thus setting the threshold at -6dBFS adds 6dB of gain so that peaks still hit zero. This can be modified by setting the output ceiling below zero, and if this is done the threshold setting relates to the ceiling value, not full scale. Why the ceiling should ever be reduced is not clear, but it does make it possible to compare raw and treated signals at the same subjective level—simple Bypass obviously removes the added gain, making comparison difficult. Reducing the ceiling by the same number of dB as the threshold setting effectively removes the gain, bringing the output level down to the same as that of the bypassed signal. In this way any side effects of the limiting itself can be heard.

The third control is for setting the release time of the limiter, but again for most purposes this is happily dealt with by the Auto Release Control circuitry, engaged by a single button marked ARC. Its associated meter shows gain reduction, while the other two show input and output levels, and all three have peak hold by default, reset globally by a single button. Attack times are not adjustable by the user, as the L2 can do better than that; it incorporates enough delay to allow it to look ahead and deal with peaks before they arrive, shaping them in the most unobtrusive way possible.

There are in fact two more knobs, and they adjust the analogue levels coming into the L2. I was surprised to find that in their centre detent position the gain did not seem to match any standard level I was expecting; in order to get a 0vu input to deliver a digital output at -18dBFS I had to increase the input level significantly, and for the often-used -12 a lot of extra gain was required. The centre detent corresponded to something like -19 or 20, and didn't even seem to be the same on both channels. Not a real problem of course, as the meters on the unit allow precise calibration, yet puzzling nonetheless. The output level meters corresponded very well with an outboard digital reference meter.

The left-hand end of the front panel has the push-button selector switches for the various parameters relating to the digital output format, and this is where the added features come in. Waves had a close and productive relationship with the late Michael Gerzon, and this resulted in several Waves products featuring Gerzon's ideas. Some were to do with stereo image manipulation, but what concerns us here is Gerzon's noise-shaping ideas, known in their Waves applications as Increased Digital Resolution or IDR. The analogue to digital conversion in the L2 is 24-bit at up to 96kHz, and all normal rates and lengths are provided at the outputs. There are three push buttons to deal with reducing the word length from the standard 24, and when these parameters are as accessible as this it's a useful reminder as to what's going on here. One switch cycles through the word lengths from 16 to 24 bits in 2dB increments; one adds a choice of two types of dither; and the other applies various flavours of IDR noise shaping to the end result. Putting a signal through the L2 at about 60dB below the usual level shows all too clearly why such things are necessary, and allows ready comparison of the various settings.



Straight truncation to 16 bits makes such a mess of a low signal that it's immediately obvious why a more sophisticated approach is called for. The 'grainy graunch' sounds like a faulty gate or a loose connection, and the implications for reverberant tails and other subtle details are all too clear. The L2 begins to deal with this with two types of dither, although no details are given as to what they comprise. According to Waves, Type 1 provides no non-linear distortion, while Type 2 exhibits a lower hiss level. Certainly Type 2 sounded much quieter while smoothing out the unpleasant truncation effects just as well as Type 1 on the material I was using. The noise shaping can then be applied, and three shapes are available, simply labelled Moderate, Normal and Ultra. The difference is the degree to which the process moves the noise into the upper end of the spectrum. Ultra leaves a very clear mid range while producing the most extreme HF noise, which in many cases will be very successfully masked by the musical material. If the masking is insufficient, a lower setting can be used at the expense of the amount of mid-band noise.

I was working with some quite aggressive, bright material, and there was no question that the most extreme settings of these controls worked subjectively the best. If it hadn't been for the prominent cymbals this might well not have been the case, and it is therefore reassuring to know that there are more gentle settings, all of which are very effective at delivering an improvement over an untreated signal. Waves claims that IDR can yield a subjective resolution of 19 bits on a 16 bit medium, and 23 bits on a 20 bit format. A claim like that is hard to verify, but there is no doubt that IDR on these low-level signals revealed detail

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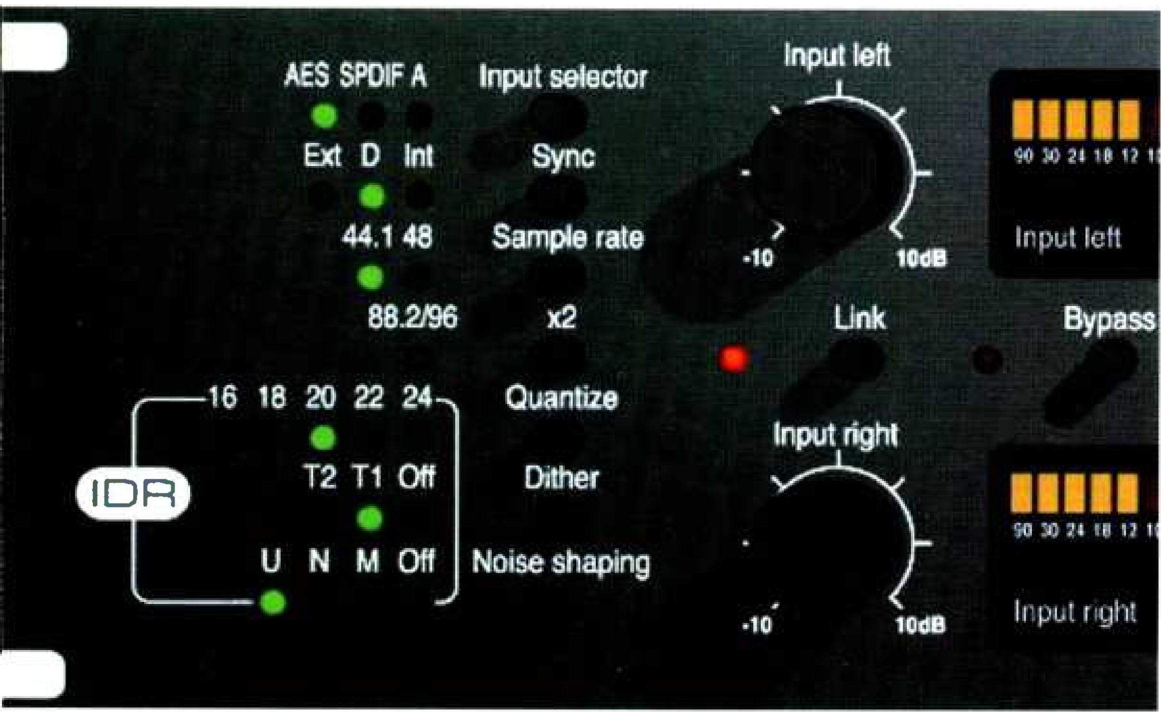

Noise-shaping with Gerzon

THE LEFT-HAND END OF THE FRONT PANEL has the push-button selector switches for the various parameters relating to the digital output format, and this is where the added features come in. Waves had a close and productive relationship with the late Michael Gerzon, and this resulted in several Waves products featuring Gerzon's ideas. Some were to do with stereo image manipulation, but what concerns us here is Gerzon's noise-shaping ideas, known in their Waves applications as Increased Digital Resolution or IDR. The analogue to digital conversion in the L2 is 24-bit at up to 96kHz, and all normal rates and lengths are provided at the outputs. There are three push buttons to deal with reducing the word length from the standard 24, and when these parameters are as accessible as this it's a useful reminder as to what's going on here. One switch cycles through the word lengths from 16 to 24 bits in 2dB increments; one adds a choice of two types of dither; and the other applies various flavours of IDR noise shaping to the end result.

that the 16 bits did not appear to contain. Of course there are many other ways of achieving this, but the Gerzon IDR seems to have found favour in its software guise; its availability now in a rackmount box for mastering may well see it gain further exposure.

The back of the L2 reveals a good selection of analogue and digital interfaces, from phono unbalanced analogue ins and outs to AES-EBU, and there is a wordclock input on BNC. If I have a reservation about the physical box it's the apparent standard of construction, which does not inspire total confidence. It doesn't look or feel expensive, which may not matter, and the push buttons are wobbly and sometimes poke out of their panel holes at an angle, which may. On the other hand, the multiple meters and read-outs, which at first look gaudy and unnecessarily flash, turn out to be extremely useful, with good use of simple yellow and red bars to show when levels are nearing the top, half-dB resolution in the critical areas, and the always-active peak hold that can be seen from half way across the room. The tiny increments in the threshold adjustment may have their uses for some people, but their main effect is to make adjustment take rather longer than strictly necessary as the encoder is cranked round and round several times just to drop the threshold by a few dB.

But at the end of the day the L2 undoubtedly delivers what it sets out to deliver. It seems to be possible to make any signal louder than it started out, with or without obvious side effects as required. Importantly, the limiter itself can be made to work quite hard without any more noticeable indication than the gain reduction meter's quick flashes. It can do this on a digital source as readily as on analogue material, and if presented with analogue provides mastering quality conversion with all the flexibility you could need. The IDR dithering and noise shaping, whether used on a high-bit digital source or following its own convertors, is no mere gimmick, but a significant and worthwhile contender in this important field. Set up with care (which is not hard to do) the L2 can make a worthwhile contribution to the final stages of anybody's signal chain. Are there more physical manifestations of Waves' expertise in the pipeline? Watch this space...

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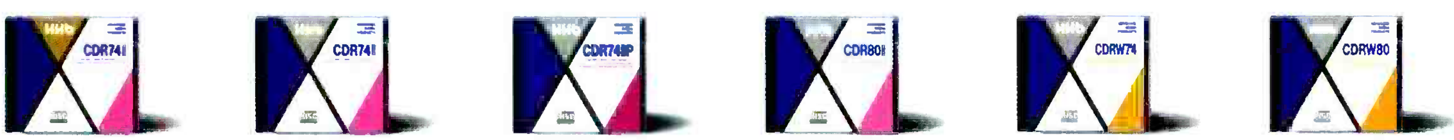

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JBL LSR25P

Studio Sound's 'bench test' loudspeaker reviews continue with the LSR25P. **Keith Holland** reports

THE JBL LSR25P is a compact, 2-way active loudspeaker comprising a 130mm woofer with a carbon-fibre composite cone, and a 25mm titanium composite dome tweeter that radiates through an 'elliptical oblate spheroidal' horn; both drivers are magnetically shielded. The power amplifier and crossover electronics are housed in a diecast alu-

minium cabinet which has external dimensions of 175mm wide by 270mm high by 240mm deep. Each loudspeaker weighs 7.7kg. The drivers are arranged vertically on the front panel along with two bass reflex ports, an on-off button with LED and an input sensitivity control. The signal and mains input sockets are mounted in a recess in the back panel, such that the plugs insert vertically, permitting flush-mounting of the loudspeaker against a back wall. Also on the back panel are a set of DIP switches which control HF level (±1.5dB), a high-pass filter for use with subwoofers (80Hz) and 'workstation LF boundary compensation'. The measurements presented in this review were carried out with all controls in their flat (default) position. JBL claims amplifier power outputs of 100W for the woofer and 50W for the tweeter giving a maximum peak SPL of 109dB at 1m distance under anechoic conditions. The crossover is specified as a 4th-order electroacoustic Linkwitz-Riley at 2.3kHz.

Fig.1 shows the on-axis frequency response and harmonic distortion for the LSR25P. The response is seen to lie within ±2dB limits from 90Hz to 20kHz; a remarkable result. The low-frequency roll-off is approximately 6th-order with -10dB at about 50Hz. The harmonic distortion performance is also remarkable, with the 2nd harmonic reaching a maximum of -47dB (0.4%) at 75Hz, with all other levels better than -50dB. These are very low levels of distortion for such a small loudspeaker enclosure.

The horizontal and vertical off-axis responses are shown in Figs 5 and 6, respectively. The response is seen to fall smoothly with increasing frequency and off-axis angle, with no evidence of side-lobes or mid-frequency narrowing. A driver interference notch is evident in the vertical plane at the crossover frequency of 2.3kHz; this is characteristic of most non-coaxial designs.

The time-domain response for the LSR25P is depicted in Figs 3 to 7. The step response (Fig.3) shows good driver time-alignment with very little delay between the high and mid frequencies, but the acoustic source position is seen to shift to over 3m behind the loudspeaker at low frequencies; a consequence of the rapid, 6th-order roll-off (Fig.2). The power cepstrum (Fig.4) shows very little activity as the on-axis response, from which it is derived, is so smooth and flat. The small, sharp dip in the on-axis response at 1100Hz is seen to coincide with some ringing in the waterfall plot (Fig.7), suggesting the presence of a low-level, high-Q resonance rather than an interference effect, but the low frequencies are seen to decay very rapidly considering the 6th-order roll-off.

Overall the JBL LSR25P is an excellent small loudspeaker. The frequency response, both on- and off-axis, is among the smoothest and flattest of any of the loudspeakers tested to date. Low frequency extension is admirable for such a small loudspeaker and the harmonic distortion is exceptionally low. Driver time alignment and low-frequency decay are both noteworthy, although the low-frequency phase response, represented by the acoustic source position plot, does suffer due to the 6th-order roll-off. The high performance and small physical size of the LSR25P should ensure that it wins many friends as an accurate close-field reference monitor. ☐

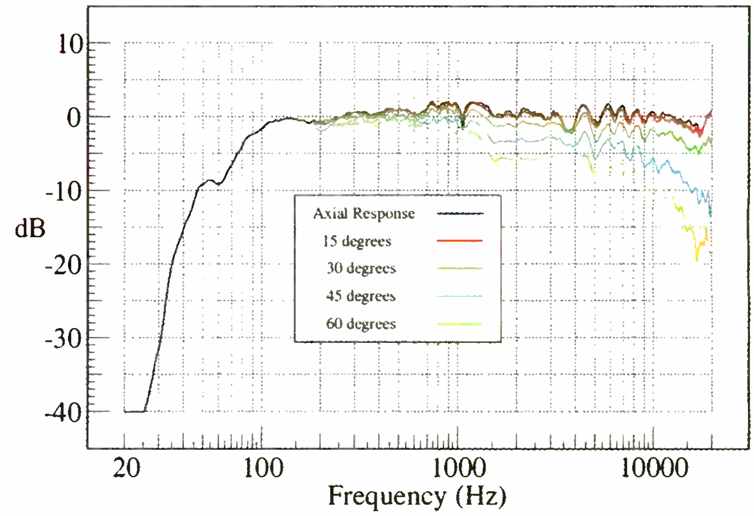

Fig.5: Horizontal Directivity

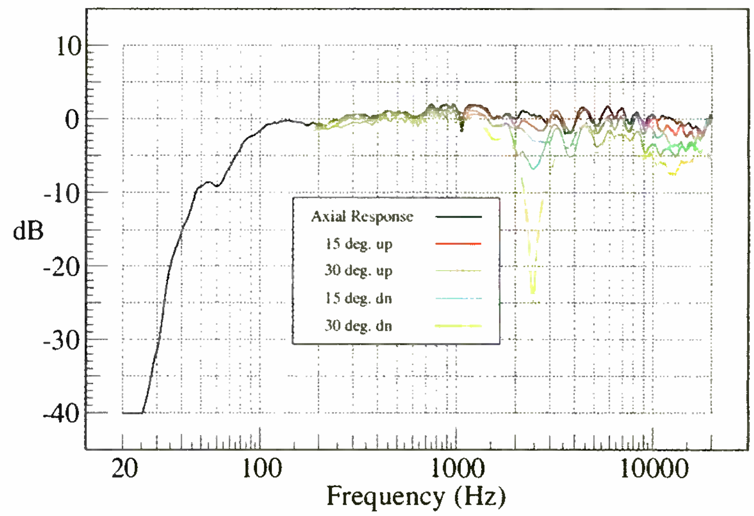

Fig.6: Vertical Directivity

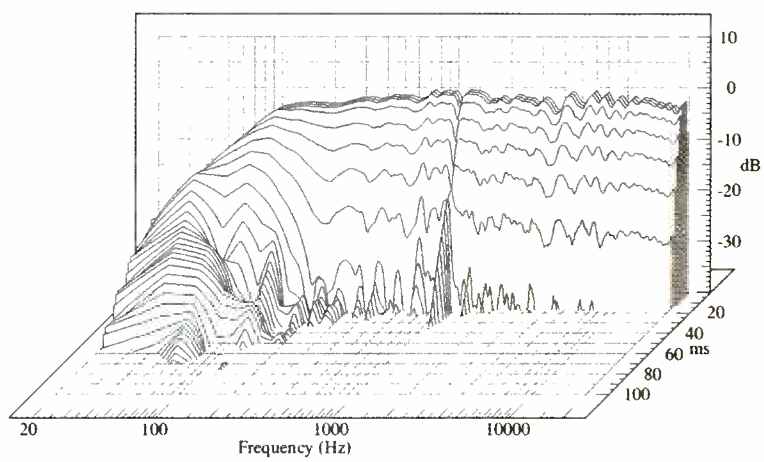

Fig.7: Waterfall

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Methodology

Studio Sound, April 1998, page 14
Net: www.prostudio.com/studiosound/aprl98/r–tannoy.html

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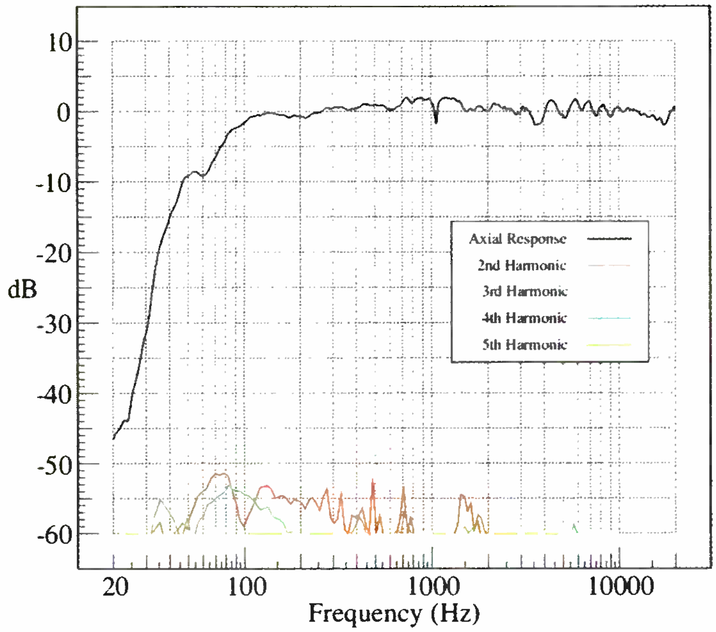

Fig 1: On-axis Frequency Response and Distortion

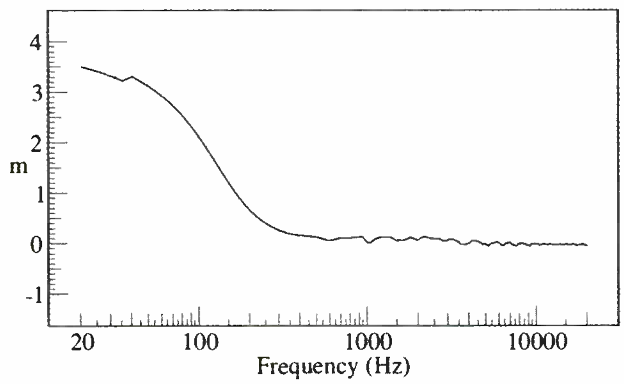

Fig.2: Acoustic Source

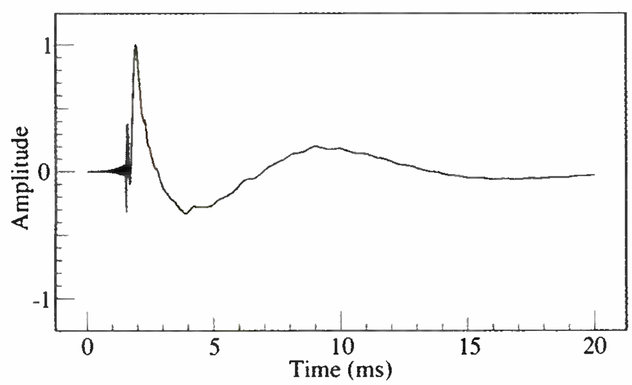

Fig.3: Step Response

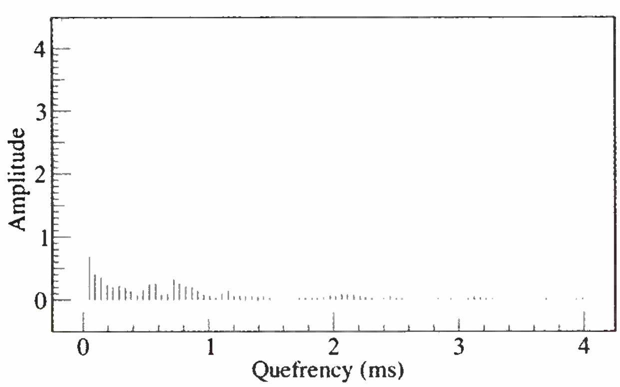

Fig.4: Power Cepstrum

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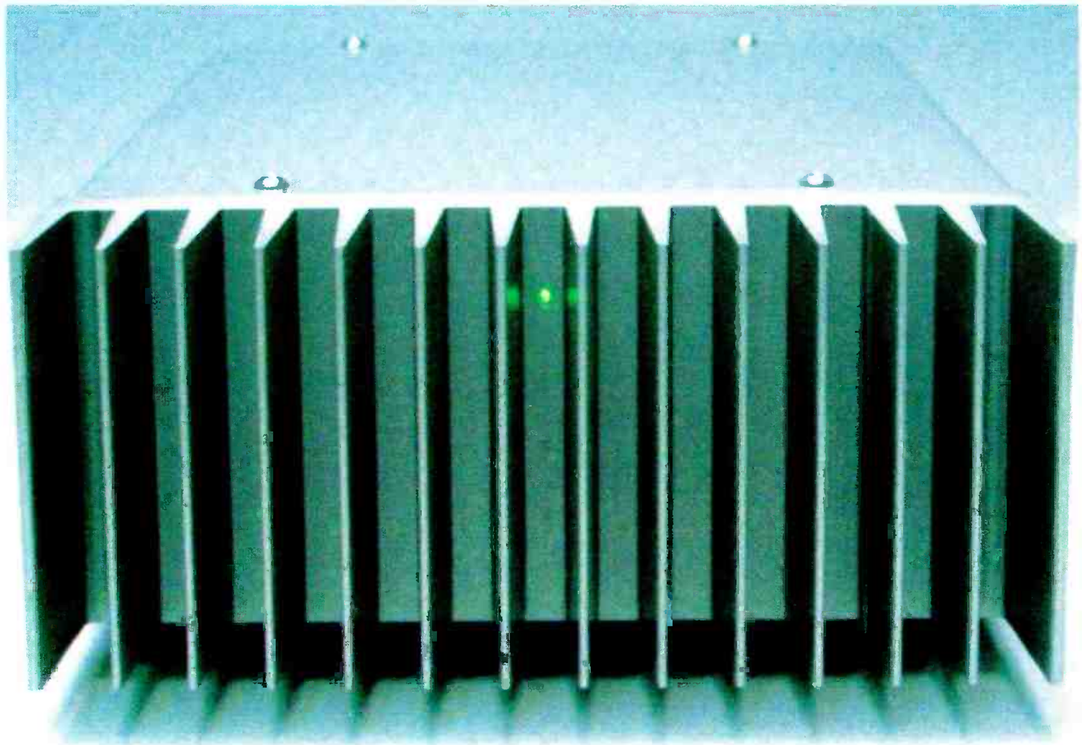
Stellavox PW1

Studio Sound's 'bench test' amplifier reviews continue with the PW1. **Paul Miller** reports



IN MANY KEY RESPECTS, the Stellavox PW1 is the antithesis of Manley's valve amplifier tested last month. Both make claims for high sound quality and both are targeted across domestic and professional markets, but the routes taken to achieve their common goals are as far apart as imaginable. And imagination certainly plays a role in this review, for the Stellavox arrived from Switzerland quite unannounced, bereft of anything but the simplest specification.

At first sight, the compact (150mm x 270mm x 60mm) dimensions and light 4.2kg weight of this monoblock amplifier, coupled with its high 200W rating and 'PW1' model name, suggest some sort of switch-mode or PWM design. Instead, the PW1 *looks* like a linear class A/B amp running with a very high (~70V) rail voltage. Two pairs of Hitachi J162/K1058 audio MOSFETs are bolted to the back of the extruded heatsinking, devices that can be married to a relatively simple drive circuit that needs no significant current capability of its own.

This ties-in with what appears to be an encapsulated driver stage with built-in heatsinking. Indeed, the entire PCB measures no more than 75mm x 95mm, and this includes a pair of small 100V/1000μF electrolytics. RCA and XLR inputs are provided, though both are single-ended and connected, unswitched, in parallel. Do not, under any circumstances, connect two sources to the PW1 at the same time. Stellavox's 3k–10kΩ input specification for the PW1 suggests that, at one time, these inputs were directly loaded by its stepped attenuator while, in practice, the output of preamps, consoles and the like are actually faced with a friendlier 25kΩ resistor. The attenuator, meanwhile, offers adjustment over a 10.3dB range (gain from +19.7dB to +30.0dB).

Because power supply energy is a function of volts squared and *not* capacitance, the PW1 can get away with a small reservoir at mid and high frequencies. Nevertheless, the low capacitance supply is less able to sustain a continuous output at and below 50Hz, as Fig.1 demonstrates, though the MOSFET output stage confers a good power bandwidth, extending to a full 240W at 30kHz.

Load tolerance is also affected. So the PW1 may achieve 240W into 8Ω, but only manages 245W into

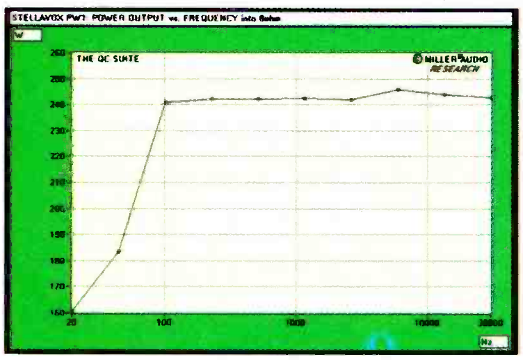


Fig.1

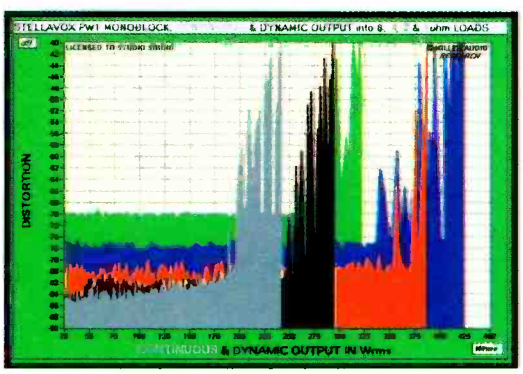


Fig.2

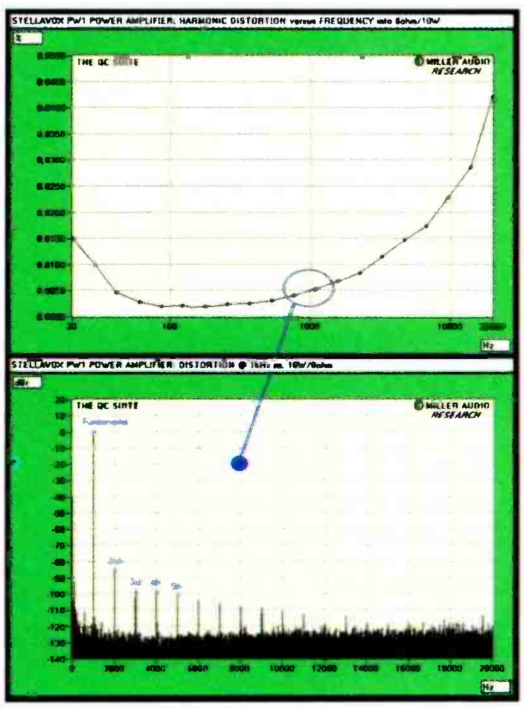


Fig.3

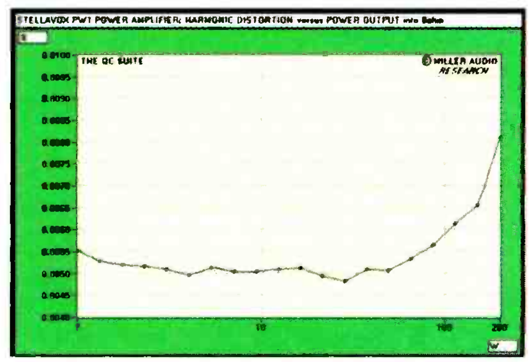


Fig.4

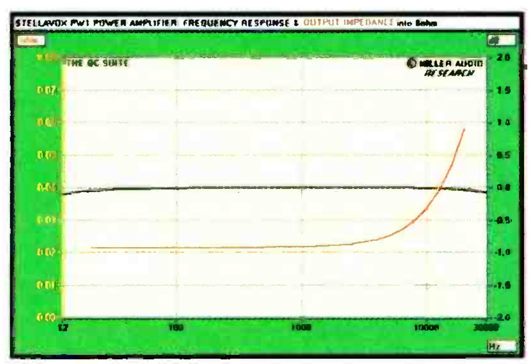


Fig.5

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4Ω under continuous output conditions. Under momentary or dynamic conditions, this increases to 292W, 386W, 417W (14.4A) and 318W (17.8A) into 8Ω, 4Ω, 2Ω and 1Ω loads, respectively. These are the black, red, blue and green traces, respectively, on Fig.2 with the continuous 8Ω output represented in grey. These profiles are very revealing, for while distortion remains both low and controlled up to 80-90% of full output, there are obvious bursts of distortion, possibly caused by a mild instability, prior to true clipping. Furthermore, the maximum 17.8A current capability is certainly not excessive for a 200W amplifier (around 30A would be more typical). There's no such thing as a free lunch: the amplifier is powerful and very capable provided we steer clear of LF reinforcement or low and reactive loads.

Assessed within these limits, the amplifier is very impressive indeed. At 10W, distortion falls as low as 0.0015% (around 200Hz) and increases to just 0.015% at 20Hz and 0.04% at 20kHz. This is reflected in Fig.3 which also includes a high-resolution spectrogram of the midband THD, showing a low but extended pattern of harmonics.

As reflected in the dynamic profile (Fig.2), distortion is also well maintained over the amplifier's specified 200W range, showing no obvious crossover effects (THD = 0.0038% from 1-10W) and a gentle increase in midband distortion levels to just 0.008% at the rated 8Ω output level. Residual noise is very low indeed at just -73.7dBV (0.21mV) and the A-wtd S–N ratio equally impressive at 90.9dB (re. 1W/8Ω). This is some 5dB better than 'average' for a product of this rating.

The exceptionally wide response of the PW1 is demonstrated by Fig.5 (black trace), though whether Stellavox's ±3dB limits of 3Hz–1MHz are strictly necessary is another point altogether, bearing in mind the stability implications of unusual speaker-cable combinations. The output Zobel network (or LR to be precise) will have a damping influence and limits the output impedance to a minimum of 0.02Ω (see red trace, Fig.5). Play it safe and there's no reason why the PW1 should turn into an RF amplifier.

Overall, the Stellavox PW1 looks to be an interesting product and certainly a departure for its parent company, Goldmund, who are—or at least were—best known in domestic audiophile circles for its high-end and highly expensive hi-fi gear. How competitive the PW1 will prove depends on its price in the UK which, with any luck, will not be on a par with previous Goldmund product. This being so, the measurements suggest that the PW1 is not best placed for driving long, reactive cable loads or bass bins with crude low-pass crossovers. Compact, close-field monitors, however, will suit the PW1 famously. □

Contact

Digital Audio Trading, 2 Chemin de la Graviere, 1227, Geneva, Switzerland
Net: www.stellavox.com

Methodology

Studio Sound, June 1999, page 27
Net: www.prostudio.com/studiosound/index.html

Power Amplifier: Stellavox PW1
(Rated Specification, in brackets where given):

	20Hz	1kHz	20kHz
Max Continuous Power Output,			
0.5% THD into 8Ω (one channel)	160W	240W (200W)	240W
1% THD into 4Ω (one channel)	-	245W (200W)	-
Frequency Response @ 0dBW	-0.05dB	0.0dB	-0.04dB
Dynamic Headroom (IHF)		+0.8dB (290W)	
Maximum Current (10ms, 1% THD)		17.8A	
Output Impedance	0.02Ω		
Damping Factor	400		
Unbalanced Input			
Total Harmonic Distortion			
(0dBW, 1kHz)	-89dB (<-80dB)		
(2/3 power, 1kHz)	-83dB (<-80dB)		
Noise (A wtd, re. 0dBW)	-90.9dB		
(re. 2/3 power)	-111.3dB (<-105dB)		
Residual noise (unwtd)	-73.7dBV		
Input Sensitivity (for 0dBW)	49mV		
(for full output)	760mV (1400mV)		
Input loading	25kΩ (3-10kΩ)		
DC offset	+16mV		
Serial Number	60-05-01		
Retail Price	£ ??		

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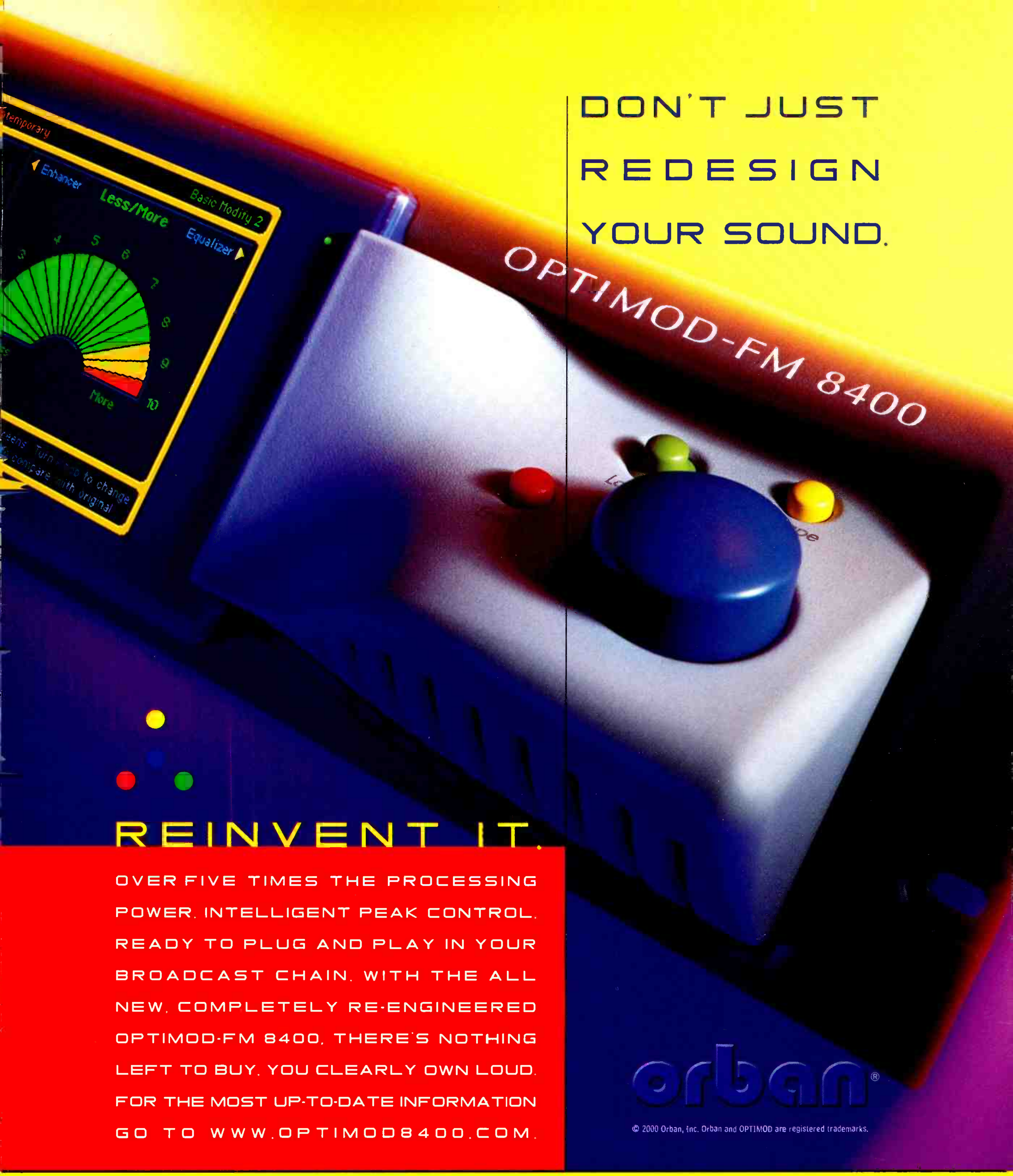

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Trident Audio S80

The emergent trend of near-identical vintage equipment reissues continues with a strip from the Trident Series 80 console. **Dave Foister** casts a connoisseur's eye and ear.

JOHN ORAM's presence at Trident during the golden years makes him well placed to understand what it is about certain consoles that made them special. So far he has applied that experience to producing new processors under the Oram Sonics banner, but now he has bought the Trident Audio name and produced a replica of the channel strip that made the Trident Series 80 the console it was.

The classic channel strip, and particularly console EQ, in a box is nothing new. If you want the original sound of a console by Neve, API, Focusrite, Amek, or one or two others, they're there to be bought off the shelf, with the added reassurance that new components and a warranty can bring. How close a link the replicas can claim with the original is very much a variable, but not with the Trident Audio S80 Producer Box.

From the metal knobs to the case made of the same distinctive wood as the original Trident trim, the S80 is a painstaking reproduction, and this runs right through the whole design, as it does on many of the other replicas already mentioned. No attempt has been made to sanitise, update or 'improve' the original design; the aim was to produce exactly the same sonic character as on the original, and that meant cloning it as closely as possible. As is often the case, the most problematic component was the transformer, and Sowter was commissioned to replicate one of the few original transformers available. The front panels were printed using the original anodising process, and the engraved push-buttons are the same; the only obvious difference is the mirror-image placement of the controls, so that the EQ has the frequency selection on the left and boost-cut on the right.

The circuitry itself comprises the microphone preamp and the equaliser, and all the controls for these elements are on a short strip. The box contains two of these strips, and a third panel that constitutes a simple mixer for the two channels, with pans, mutes and grown-up faders. Each channel has its own XLR inputs and outputs plus insert points, and the mixer has separate stereo outputs.

The controls themselves are relatively few and simple, and will be familiar to those who remember the original console. The preamp has separate gain trims for the mic and line inputs, a switch to select between them, phantom power and phase reverse. The EQ itself has four bands, the outer two shelving and the inner two switched-frequency mids. This is where the real simplicity is apparent, as there is no bandwidth adjustment, no great overlap between bands, and fairly coarse switching of the frequencies. HF and LF have only two frequencies, selected with push-buttons, while the mid bands have seven each on rotary switches. There is a separate low-cut filter at 50Hz and a button to switch the whole EQ in and out, shown by the only LED on the strip.

But features and complexity are not what an EQ like this is all about. The Series 80 console had a sound that people wanted, and the success of this Producer Box depends solely on how well it duplicates that sound. Oram's studio had an original Series 80 channel strip hooked up alongside the new box via an A-B switch, and I was left alone to explore the two.

The character of the EQ was immediately obvious. There is a great big warmth in the lows and mids, and sparkle and edge in the upper end. The HF shelf set at 8kHz is quite aggressive and easy to overdo, while at 12kHz it adds a beautiful smooth sheen. Switching around the mid frequencies with lots of boost showed the Q to be narrow enough to be able to hear the centre frequency while wide enough to give broad control without worrying about how few frequencies there are to choose from. The most remarkable thing though was the fact that Oram has achieved the aim of duplicating the original; any setting I created on the original could be cloned exactly on the S80 with no audible difference. The one variation is that the new boost-cut pots have a slightly different law from the old ones, and have to be moved a little further away from the centre detent to get the same result, although the settings at the extremes are identical.

This opportunity to do such a direct comparison is a rare thing for a manufacturer to offer to a reviewer, and Oram's confidence in the S80 design was vindicated. This is truly a replica of a desirable original, and if that particular colour is missing from your palette the S80 will fill the gap. ☐

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NEW TECHNOLOGIES

CD-R for glassmasters

Marantz has released the CDR500 combination CD-R/CD as its first product with a new Automaster process for full Red Book Disc at Once recording. According to the manufacturer, this replicates mastering plant procedures and negates the need for

transfer of recorded material to exabyte and produces a CD-R from which Glassmasters can be produced directly. Features include a DSP audio buffer, CD text writing, normal and x2 duplication, full SCMS manipulation, balanced XLR inputs, SPDIF I-O and CD/CD-RW playback. Other features include alternate and parallel playback, 12–56kHz and 96kHz SRC with bypass, digital recording level attenuation, smart auto and manual stop, and serial and IR remotes.
Marantz, UK. Tel: +44 1753 686080.

CEDAR 'zero latency' suppressor

CEDAR Audio is to launch the DNS1000 Dynamic Noise Suppressor at the AES Convention. The DNS1000 is a free-standing unit designed for audio post, live broadcast, and forensic audio. With virtually zero latency, its 40-bit multiband processing is suited to removing unwanted noise from location sound, sound effects and dialogue. Described as a 'breakthrough' in noise suppression, it will remove rumble, hiss, whistles, broadband noise, and the 'shot' noise that often makes audio unusable. The user-interface is simple yet powerful in employing a few faders and switches and should make it extremely attractive in film mixing applications.
CEDAR, UK. Tel: +44 1223 414117.

Avant v4

Version 4 software for SSL's Avant postproduction and film console adds new grouping options that enable controls such as panning and EQ to be linked. The addition of PenPoint panning also gives operators the option of screen-based surround panning using simple pen and tablet operation in addition to operation from channel pan controls and physical joysticks. The Virtual Paddles feature available with v4 provides additional monitoring and recorder control from a smaller number of physical paddle switches. Aysis Air Mobile is a compact-format console for OB vehicles and space-restricted studios. Using the standard Aysis Air software, the console's channel layering function enables a fully specified 96 channel console to be fitted in a 48-fader frame less than 2330mm wide.
SSL, UK. Tel:+44 1865 842300.

DSP file transfer

DSP Media has introduced AVtransfer, an optional self-contained software program for file translation and conversion between workstations. AVtransfer reads and writes all the common professional audio media formats including 'difficult' and obscure files as well as

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TL Audio Fat One

Whether the Fat Man takes the drudgery out of setting up a compressor or limits your options is food for thought. **George Shilling** tries a new valve diet

I WAS JUST a few days into my health diet when the Fat One arrived, threatening to spoil things. The novelty of receiving a cubic container was quite irresistible—apart from mixing desks and microwave ovens (CD changers) I rarely review something that cannot be bolted into a rack. Except that despite the Fat One's chubby looks, by the time you read this there will be available a matching red rack tray, enabling one central or two adjacent Fat Ones to be held in a 3U-high rack space. For the time being, though, I had to balance the One on top of my rack—those Fat blokes had kindly supplied some stick-on feet for the purpose.

The Fat One is a stereo compressor, with two inextricably linked channels. The front panel therefore quite reasonably features only one set of controls. It is, like many other TL units, a hybrid valve/solid-state unit, the single dual triode serving both channels, providing a preamp stage before compression takes place.

The Fat One has certainly not been dieting—it is quite weighty. The sturdy case features a mesh top, that rakes down backwards from the front panel, providing ventilation for the circuitry—even just one valve generates plenty of physical warmth. The simple, small rear panel features an IEC mains socket and input and output connections on balanced jack sockets (which will also happily accept unbalanced plugs) with a GAIN switch to select -10dB or +4dB operation.

The front panel features a single cute illuminated vu meter, that works well, although a larger one would of course be preferable from a pro-user's perspective. A button allows display of Output Level or Gain Reduction. There is a THRESHOLD knob with a -20dB to +10dB range, and a RATIO knob with a range of 1.15:1 to 1:30, with 1:3 coming about a third of the way up. These ranges are well chosen and allowed for any settings from very subtle 'tickling' to full-on limiting.

There are also two separate buttons to switch fast-slow for Attack and Release settings, which give enough variation for many situations. There is also a hard-soft KNEE button, which also gives the unit some unusual flexibility for a budget unit, although the difference can be subtle on many signals. Input and output gains feature useful ±20dB ranges, with centre detent. Turning up the INPUT GAIN will drive the valve harder. A GAIN MAKE-UP knob provides up to 20dB further gain when the COMPRESSOR ON button is activated, usefully allowing one to roughly match the uncompressed and compressed signal levels. An LED near the meter glows when COMPRESSOR ON is pressed. Sensibly, a front panel power rocker switch is included, with an LED indicating power on. Unusually, for a completely analogue compressor, there is a big rotary knob that clicks into each of 16 positions: 15 presets and a Manual setting. These presets cover settings for Threshold, Ratio, Knee, Attack and Release, disabling those controls. The presets are usefully described by the application the designers recommend them for. So there are settings for Vocal (3 positions), Keyboards, Bass (2), Acoustic Guitar, Electric Guitar (2), Snare, Kick, Kit and Mix (3 settings).

Perhaps slightly to my surprise, these worked rather well, with minimal tweaking of the Input and Output Gains. They were always a good starting point, and if not quite right it was very easy to try something else. Unless you refer to the manual, you might not guess their exact settings, which is not always a bad thing. Inevitably, with such few variables, a number of presets are similar, for example the Vocal 2 setting is the same as the Mix 2 with a slightly different threshold setting.

But it is a fun new way to work, which is very quick in practice. If you want to get fussy, there are a couple of explanatory charts in the manual which explain the settings and show exactly where the controls should be to recreate the presets, allowing you to understand and modify them.

The unit performed particularly well across stereo mixes, the valve circuitry injecting a vibrancy, which brought my pop-rock track alive in the midrange, but always retaining a solid, indeed 'fat' low-end. There is not the 'zinginess' I would associate with a Focusrite, but instead a fat, warm, rockin' 'British' sound, especially good with lively mixes and rock guitars. The use of a transconductance amplifier, rather than a VCA, for gain control, is credited with giving the unit its characterful sound.

This is also used in more expensive TL units. An added bonus is a pair of Instrument jacks on the front panel, which gives the unit a useful DI box role. These sound terrific with an electric guitar, and can be used simultaneously with the rear inputs with no loss of level.

The Fat One is cheap for a unit featuring valve circuitry, albeit featuring just one valve. But even forgetting about the valve it is a funky, friendly, good value compressor. Wherever you set the controls, you are guaranteed a truly fat sound. I knew the diet wouldn't last...

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NEW TECHNOLOGIES

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Studer D950 M2

The Studer D950 M2 represents the evolution of the Studer D950 and has an all-new look that exemplifies the enhancements to the software-based feature set. It combines enhancements to the software-based feature set with a newly designed control surface.

It comes standard with a new Central Assign Section, colour 8-channel surround meter, as well as a larger 15-inch TFT colour display monitor. The knob sections contain new rotary encoders with an integral 21-LED ring for display of knob values and each knob is flanked by an alphanumeric read-out. Almost all the channel circuitry has been redesigned and enhancements have been made to touch sensors, power distribution, and the moving fader servo amplifiers. Studer 24-bit convertors are used, all digital I-Os are 24-bit, and internal processing takes place at 40-bit floating point precision on a 32-bit bus. The M2 digital core is fully configurable and console capabilities can be expanded by installing additional DSP cards. The M2 features Studer's proprietary Virtual Surround Panning (VSP) and the architecture has been extended to provide smoother and more natural early reflections, as well as the addition of late reflections.
Studer, Switzerland. Tel: +41 1 870 7511.

Behringer mic

Behringer has introduced the B-2 dual-diaphragm condenser mic. The one-inch dual-diaphragm capsule with gold-sputtered membranes is accompanied by gold-plated internal head pins and FET circuitry with switchable omni or cardioid patterns. A switchable high-pass filter and 10dB pad are included and the mic comes with a protective carrying case, a shock mount, and windshield.
Behringer, Germany. Tel: +49 2154 920 6237.

Dolby Surround SFX

Produced by Renaissance Sound Technologies, Renaissance SFX is the first sound-effects library completely encoded and produced in Dolby Surround. It starts with a first package of 7 CDs with a further four to follow as part of a process of constant updating. The library offers a huge quantity of immersive Dolby surround sound-effects and complete sets of musical tools to produce music soundtracks in surround. All have been recorded on location and have been created by using proprietary miking techniques and software tools internally developed by the Renaissance sound engineers to create a natural surround-channel.
A Dolby Digital version of the library will be available on demand.
Dolby. Net: www.renaissancesfx.com

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Ridge Farm Boiler

The latest in what seems like a range of kitchen appliances, **Zenon Schoepe** scrubs up and clears the table for extreme compression

TARGETED AT THOSE WHO LIKE their compression to be strong and apparent the Boiler from Ridge Farm Industries, the manufacturing interest of the Ridge Farm UK residential studio in Surrey, is a dual channel stereo linkable compressor-limiter. Control options are fairly limited and centre around the box's ability to be pushed to excess to yield what the literature refers to as 'boiled', 'boiling' and variable degrees of 'dirt'. Sounds promising, then, if you are seduced by the concept of dynamic control as a creative processing exercise that is distinct from the more mundane business of clamping down and controlling wayward signals.

Connection is via balanced XLRs on a box that is refreshingly different in look to the majority of 1U-high alternatives. It's a heavy-duty and heavy box containing all solid-state electronics which comes in a colour not dissimilar to BSS' Opal range, but with the proud inclusion of a centrally mounted chrome Ridge Farm Industries badge portraying a stylised picture of turn- of-the-century craft.

Remember that it's not the first device from this operation, the Gas Cooker preamp and DI remains a very capable unit that has found homes in numerous control rooms. It also established the culinary connection through the use of old-style chicken-head knobs for its pots.

Controls on the Boiler are similarly few, but as already intimated operation and results relate to how you use what there is. Each channel gets small INPUT and THRESHOLD pots working in conjunction with two 2-position toggles for release and attack constants. Given the stance of the box you'd be right to not expect any associated detailed legending or figures for the controls and this extends only to simple fast and slow settings for the ATTACK and RELEASE switches. Flicking a LINK switch hands side chain, threshold, attack and release control to the left channel with input levels remaining independent. Both channels are bypassed on a single switch and metering on each channel comprises ten LEDs which always follow the output level regardless of BYPASS position.

True to its intentions, this box has character and it is quirky in its operation. The first thing you notice is that a surprisingly varied palette of control is afforded by the few physical controls. This attitude aligns the Boiler more with ancient compressors than it does with the sort of features that are expected, or at least given anyway, in dynamics control devices today.

To add to the manufacturer's descriptive names for its compression effects I would add 'parboiling', 'simmer' and 'autoclave' to attempt to highlight, in the first two, the unit's skills at subtlety and, in the third, underline just how extreme the boiling can be.

Pile in the INPUT level and sweep the THRESHOLD pot around and there is almost a stepped alteration in character at around the threshold point with what appears like programme-dependent alterations of the attack and release settings, although this is probably a trick of my ears. Changing attack and release settings emphasise parts of the spectral content of the original—thickening in some instances and lightening at the leading edge in others. Despite the fact that I felt initially sceptical about their effectiveness, the two envelope switches really do have a great bearing on the performance of this unit.

The scope is enormous and also alters with programme type. Wind the input back and you have a completely different compressor to play with and one that is arguably more akin to more ordinary devices. In this application the results are as would be expected, although the over-threshold excursions do sound more classically squashed than many modern alternatives. The effect is not dissimilar to tape saturation and consequently highly usable on solo instruments and vocals.

At extreme settings you are certainly getting more out of the box than went in as dialled in distortion is part of the deal—look at it as super tape saturation followed by heavy compression and you'll be in the right sort of area. You can pile the level in and the meters will lock solid and what comes out is a fantastic pumping caricature of the input with a fizz of presence, wellington boots full of bass and a roundness that has traditionally been the preserve of older and more exotic devices. It's very retro and not especially transparent, but then this box is about boiling not stir-frying.

I don't get the impression that the circuitry is particularly radical, but it seems that its designers have dared to allow themselves to turn up the wick at the more extreme settings. It's a bold move, but its built on a fundamental character that is pleasing and musical. If you are taken by the concept then you would get a lot of use out of your investment. (£550+ VAT,UK).

Nicely presented, easy to use, easy to be impressed with. Use it when you want to be heard and when you want the effect to be heard. There's nothing else quite like it. Recommended. □

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NEW TECHNOLOGIES

Dialog4 additions

Sountainer is the first portable MP3 recorder-player to be equipped with a microphone input, line I-O and USB interface. Designed for reporter field use, MP3 files can be transmitted over the Internet to the radio-TV station via the USB Interface. Price EURO 430.00 including 64Mb memory and microphone and line cables. MusicTaxi NET is Dialog4's latest codec development allowing the use of ISDN , X.21 , TCP-IP and UDP-IP transmission over 10base tx and ATM network—TCP-IP for point to point, UDP-IP for broadcast and multicast transmission. MusicTAXI NET is based on the MusicTAXI SL-PRO and VP-PRO operating platform and provides all their functions and features.
Dialogue4. Net:www.dialog4.com

Logger and convertors

New from Sonifex is the Net-Log audio logging recorder which is able to record 4 mono or 2 stereo audio streams for playback using TCP-IP. Audio is encoded in MPEG layer 2 format and written to a large internal EIDE hard disk drive. Playback is carried out by streaming the audio across a network onto PCs. All the Net-Log setups and operational features are controlled by PCs, with additional operator over-ride buttons on the front panel of the unit. The RB-ADDA A-D and D–A convertor is 24-96 capable and produces an AES-EBU or SPDIF output from balanced XLR or unbalanced phono stereo input. The RB-SD1 Silence Detection Unit monitors an audio signal and, in the event of the input dropping below a preset level for a predetermined length of time, will automatically switch through to an alternative stereo audio signal.
Sonifex, UK. Tel: +44 1933 650 700.

UCR310 UHF synthesised receiver

The UCR310 is claimed to define a new class of wireless microphone receiver that the manufacturer refers to as 'universal'. The assembly is compact enough for use on cameras or sound carts and in portable audio bag systems and claims RF and audio performance suited for high-end film and studio

applications. The receiver can be powered by four AA batteries via a detachable battery cartridge, or from external DC power. 256 frequencies are user-selectable via two rotary switches. As the frequency is changed, the receiver automatically retunes the front-end filters to keep them centred over the operating frequency. This design provides the high selectivity and IM rejection characteristics of a high performance, fixed frequency receiver while still offering the frequency agility needed for mobile applications. The selectivity of the front-end filters allows maximum sensitivity, to increase operating range and minimise noise even in difficult RF environments. A very high RF overload capability prevents interference problems commonly caused by strong external RF signals.
Lectrosonics, US. Tel: +1 505 892 4501.

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Audio Technica AT895

Audio-Technica's latest baby offers to revolutionise shotgun microphone design through a cunning combination of technologies. **Neil Hillman** is impressed

PAUSE, IF YOU WILL, while you peruse this periodical. Look above the magazine and take stock of the people around you; momentarily listen to the world; remember where you were when you first read this. Because all of these details will become important when the sons and daughters who walk-on in our phonic-footsteps demand to know the minutiae of the moment that you became aware of a significant and remarkable new arrival in our midst: the Audio-Technica AT895 adaptive array microphone.

The AT895 is a DSP-controlled 5-element microphone array that provides an adaptable directional pickup pattern. By using Audio-Technica's DeltaBeam technology, the AT895 system manipulates and filters the output of the array by acoustical, analogue and digital means. In simple terms, the processing

enhances the pickup of a sound source from a desired background relative to the unwanted background noise or interference by means of cancellation up to a maximum of 80dB. It also boasts minimised proximity and nearfield effects on the low-frequency directionality of the array and offers a marked reduction in mechanical handling noise, such as that generated by racking a boom, and a lower susceptibility to wind noise. If it could cook, you would want to take it home to meet mother.

The business end of the AT895 itself consists of one A-T MicroLine element—a 6-inch 'line-and-gradient' microphone—that is centrally mounted, surrounded by four A-T cardioid elements mounted in a co-planar diamond configuration. All five elements are fixed-charge back-plate, permanently polarised condensers. The output of this array is fed to a control pack that allows selection of either of three modes: two 'adaptive' modes and one 'non-adaptive' mode. In the adaptive modes, signals from the MicroLine element and either one pair, or both pairs, of the correcting cardioid elements are used. These signals are processed within the Control Pack by both analogue and digital means to provide a continuously-adapting rejection of off-axis sounds. As the off-axis intensity, direction or even wind-noise changes, the system compensates for the changes.

The Full-Field Adaptive mode provides the maximum directionality and off-axis rejection, with all five elements working. Typically, compared to a larger shotgun microphone, the AT895 can provide an extra 50dB of separation. In the Planar-Adaptive mode, just the vertical pair of cardioid elements are in use, resulting in an elliptical pattern that provides optimum rejection in one plane giving a tighter vertical, wider horizontal pick-up. Obviously, recognising the correct orientation of the unit in its mounting is crucial for the resulting effects to be applied in the envisaged direction. The Line and Gradient mode is non-adaptive, with just the centre MicroLine element being used in conventional shotgun fashion.

The Control Pack provides all the power, DSP functions and control for the unit and is housed in a strong, crackle-finish metal case. It is powered either from three 9V MN1604 batteries, mounted in a clip-on holder that locates securely on the back-plate of the Control Pack, or through a 4-pin Canon external-power socket, accepting input voltages in the range 9V-15V. Battery life for the alkaline MN1604 cells is typically 4–6 hours, with other sources such as NP-1s providing more duration. Either side of the external power socket on the bottom panel are the Mic In and Mic Out sockets. The former taking a 'reverse convention' male 7-pin Cannon; the latter outputting at mic level from a more usual male 3-pin XLR socket. The top face of the Power Pack is home to a ¼-inch headphone jack, with a control pot next to it for direct-monitoring level control. Alongside, a 3-position FILTER slider-switch offers a choice: Flat; a High-Pass of 80Hz with an 18dB/octave gradient or a Band-Pass of 300Hz–5.5kHz with a 6dB/octave gradient. Below this switch is the 3-position MODE slider-switch for the selection of Full-Field, Planar or Line+Gradient settings. A small LCD screen to the centre-right of this top panel displays countdown markers to gauge battery life during operation with the 'internal' MN1604 cells. An ON-OFF slider with a red LED above it, is to the right.

The output impedance is given as 450Ω, with a stated frequency response of 60Hz–12kHz; the dynamic range is said to be 93dB. The 200Hz rejection figures for 90°/270° also make interesting reading compared to other typical shotgun microphones: a whopping 55dB greater than its rivals, by achieving 70dB.

The AT895 is designed to be used on location, in the studio, even hand-held. Raised aloft Olympic-torch style though, and the substantial Rycote 'Zeppelin' cover initially suggests that the system has indeed been fashioned from lead; until fitted into the perfectly balanced pistol grip, with its centre-of-gravity minimising the effect of a 2lb total weight. However, a fish-pole operator on location, with the rod at full extension, will either develop the upper-body musculature of Schwarzenegger or promptly declare 'I'll not be back'. But the natural home for this amazing device will be in revolutionising outside broadcast's sports stadiums acquisition, or in countless television studio's worldwide as the *de facto* Fisher boom microphone. Remember where you heard it first. □

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Audio-Technica, US
Tel: +1 330 686 2600 **Fax:** +1 330 688 3752
Net: www.audio-technica.com

NEW TECHNOLOGIES

Digitranslator

Digidesign's DigiTranslator 1.1 is a Windows NT-compatible version of Digidesign's OMFI session interchange application. DigiTranslator accurately handles the conversion of Pro Tools session files to OMF files, and of OMF files to Pro Tools session files. It can be installed on any Pro Tools system, any Avid system or on a dedicated conversion workstation and can translate files over a network and can batch convert media files in background while other applications are running. With Pro Tools v5.0.1, Windows 98 support will be available for Digidesign's home and project studio products, the Digi 001 and Digi Toolbox XP. Pro Tools 5.0.1 adds support for AVoptionlXL, bringing audio and video postproduction power to Mac OS-based Pro Tools TDM systems. Using Avid's Meridien hardware, AVoptionlXL supports capture, import and playback of uncompressed, Avid-compatible video media directly within Pro Tools. Digidesign, US. Tel: +1 650 842 7900.

New R.Eds

Soundscape will unveil two new versions of R.Ed at the IBC. R.Ed/16 and R.Ed/24 use the same SSAIRA award winning architecture as used in R.Ed/32 but are tailored for different applications and budgets. Both models have fully compatible editing software, interchangeable files and hard drives, and support for additional Soundscape software packages, like the CDWriter Mastering Package and EDL Processor which handles OMF Import Export and support for all the popular EDL formats for auto conform. The full range of available real-time DSP effects and TDM plug-ins can be used with any R.Ed model as a typical networked studio installation might include a number of different R.Ed models, dependant upon the particular tasks performed in each room. R.Ed/24, which supports Emagic Logic Audio 4.5 as it's front-end software providing integrated audio and MIDI sequencing, is aimed at music recording and editing.

The hardware itself is capable of 24 tracks, 24-bit at 48kHz (or 12 tracks at 24-bit, 96kHz), with 24 digital I-OS via three TDIF ports and has one removable hard drive bay plus one position for an internal hard drive providing 274Gb of storage. For synchronisation, the R.Ed Sync LTC option board adds SMPTE time code I-O to the standard MTC. R.Ed/16 is for the user who needs a professional level of recording and editing with balanced I-Os but doesn't need the full track count of a R.Ed/32. Typical applications include radio editing, mastering and restoration work, or integrated with a non-linear video editing system. R.Ed/16 has 16 tracks, 24-bit at 48kHz, or 8 tracks at 24-bit, 96kHz with up to 12 inputs and 16 outputs via 2 in/4 out AES-EBU digital, plus 8 digital I-Os via one TDIF port. The

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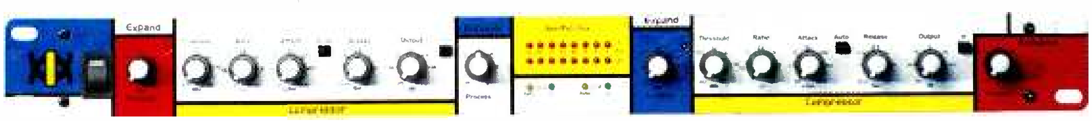
Pure Distribution XIX processors

Mating Mondriaan styling with modest pricing, the Pure XIX series makes sound sense. **George Shilling** checks out three bargain boxes

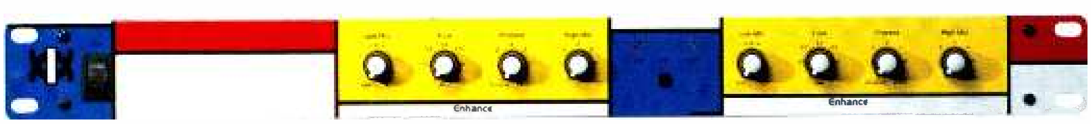
AT THE BOTTOM END of many markets, presentation and marketing are everything. That a new XIX range of effects processors from Pure Distribution comes dressed as if Mondriaan had done the paint job is notice of its very low price tags, although build quality is quite good. The legending is quite good too, but I struggled to read any of the black on blue lettering in any (fairly well-lit) studio. All units are housed in 1U-high rack boxes and are solidly put together, with all controls operating positively. The knobs are stiffly damped and most have a gently stepped feel, which is pleasant on continuously varying controls, and makes recalling or matching settings a little easier. Each unit comes with a comprehensive manual, featuring technical specifications and plenty of helpful tips and suggested applications, although the ME-1903's is less informative. All feature a front panel LED-equipped mains rocker switch, and an IEC mains socket and cable. Indeed, it is good to see that even at the budget end of the market, there is life without wall-warts. Rear panel connectors vary considerably depending on the unit.

The CP-1902 is a dual-channel compressor, featuring an expander and enhancer on each channel. On the rear are input and output connectors on balanced-unbalanced jacks. There are separate 'detector loop' jack sockets for each channel, which provide a useful side-chain input with a signal output on the ring of the TRS connection. Unfortunately there is no stereo linking for any of the functions, so for stereo processing there may be some image shifting, unless you rig up something using the detector loops. In practice, this is not always a problem, but it does seem a bit of an oversight. The expander is a simple single- knob affair, with an LED, which lights to show when gain is reduced. This is very effective, with a pleasant 'dropout' curve when it acts, no doubt due to the INTERACTIVE RATIO CONTROLLER. The range of the knob covers just about all eventualities. It is good for gating, say, guitar hum between licks or other background or processor noise. Range is not specified, but I would guess it to be at least 60dB. The compressor section is richly featured, with a wide-ranging threshold (-40 to +20), a comprehensive ration range (1:1 to 8:1 limiting, with 2.5:1 at the halfway point), and separate attack and release controls, which allow a wide variation of settings, although audible pumping occurs in many situations unless set carefully. The Auto mode isn't bad, disguising much of the pumping, so I tended to use this mostly. Finally there is an enhancer, which interacts with the compressor to compensate for lost high frequencies. This is another simple one-knob control, which is surprisingly effective. No harshness or extraneous harmonics are generated, simply a high frequency boost related to the amount of compression set. Oddly, though, it is placed in the signal chain such that if the expander is gating, high frequency signals are still audible if you turn up the Enhancer. This might be a useful trick, if, for example you wanted to use the unit as an exciter from an auxiliary send. Channel 2's controls are laid out in the same order on the right-hand side, but finding the right one isn't helped by the colour scheme, as pairs of controls feature different background panel colours. The overall audio performance of this unit is pretty good, with excellent noise and crosstalk figures. I couldn't make it sound in any way nasty or distorted, even at full-tilt limiting with the LED Gain Reduction meters glowing brightly. The compressor uses a simple VCA controlled sensor, and is unexceptional, but reasonably good results can be obtained if you are on a tight budget.

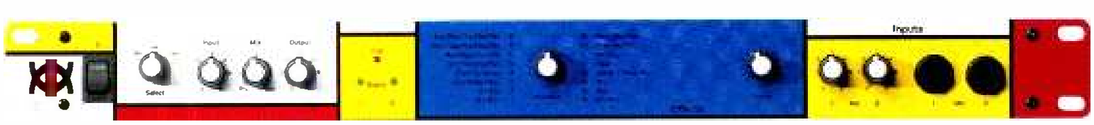
The EH-1901 is a stereo enhancer. The rear panel simply features balanced-unbalanced jacks for input and output connections. On the front are separate controls for each channel, although I suspect these will mostly be set to the same settings for stereo operation. In the centre is a solitary IN button, that lights



CP-1902



EH-1901



ME-1903

NEW TECHNOLOGIES

optional R.Ed analogue interface board provides 2 in/4 out balanced XLRs using high specification 24-bit, 96kHz A-D/D-A convertors. As with R.Ed/D24, the system has one removable hard drive bay plus one position for an internal hard drive, for up to 274Gb of storage but the sync option is more comprehensive offering optional VITC and LTC I-O plus sync to RS422 and video reference signal. MTC sync is also provided. Soundscape, UK. Tel: +44 2920 540333.

Nady studio mic range

Nady Systems recently introduced several new studio mics, the SCM Series, which it claims represents a price/performance breakthrough for large diaphragm

studio and broadcast microphones. The SCM Series includes 5 models: the 900, 910, 920, 980, and 1000, with the SCM-1000 being the top of the line. All models feature true condenser design with large pressure-gradient, gold-sputtered, ultra thin diaphragms and FET preamplifiers. They also feature internal shock-mount construction and the five models have different combinations of controls, with the SCM-1000 offering all options with selectable low-cut filter, 10dB attenuation pad, and omni-cardioid-figure 8 patterns. Nady, US. Tel: +1 510 652 2411.

Surround mastering

The z-Q6/z-VL6 surround sound mastering suite comprises a six-channel digital mastering EQ and six-channel digital dynamics processor all working at 24/96. EQ offers four bell and two shelving filters per channel together with individual channel volume offsets and master volume control. Dynamics include compressor/limiter with variable ratio, attack, release, threshold, and gain make up. Interchannel linking is possible and the package includes wordlength reduction and snapshot automation.
Z Systems. Net: www.z-sys.com

Toa UHF

The dynamic cardioid WM4200 hand-held and electret cardioid WM4300 lavalier mic head elements have adjustable input sensitivity while rubber coating eliminates handling noise. The WT4800 tuner employs Space Diversity technology and a proprietary algorithm

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X-10 Studio Monitor

This Really IS Rocket Science.

Actually, it's cutting edge aerospace technology.

It began with a sophisticated technique developed to maintain control of inherently unstable supersonic aircraft such as Stealth fighters.

Now Meyer Sound has licensed an implementation of this technology from the University of California at Berkeley, adapted it for professional monitoring applications, and built it into the new X-10 High Resolution Linear Control Room Monitor. X-10 is, quite simply, the most accurate and neutral studio monitoring system ever conceived. Frequency balance is consistently linear regardless of monitoring level. And the impulse response looks more like that of an electrostatic speaker than a monitor capable of 136dB peak SPL.

The X-10 is yet another quantum advance from a company accustomed to confronting theoretical obstacles and pursuing a solution for as long as it takes. First it was the patented phase correction technology of the HD-1. Now the X-10 is another experience you can't afford to miss. Put a pair in your studio and run them with the most demanding material you've got. Then judge for yourself. Call Meyer Sound to arrange an audition. But, be forewarned.

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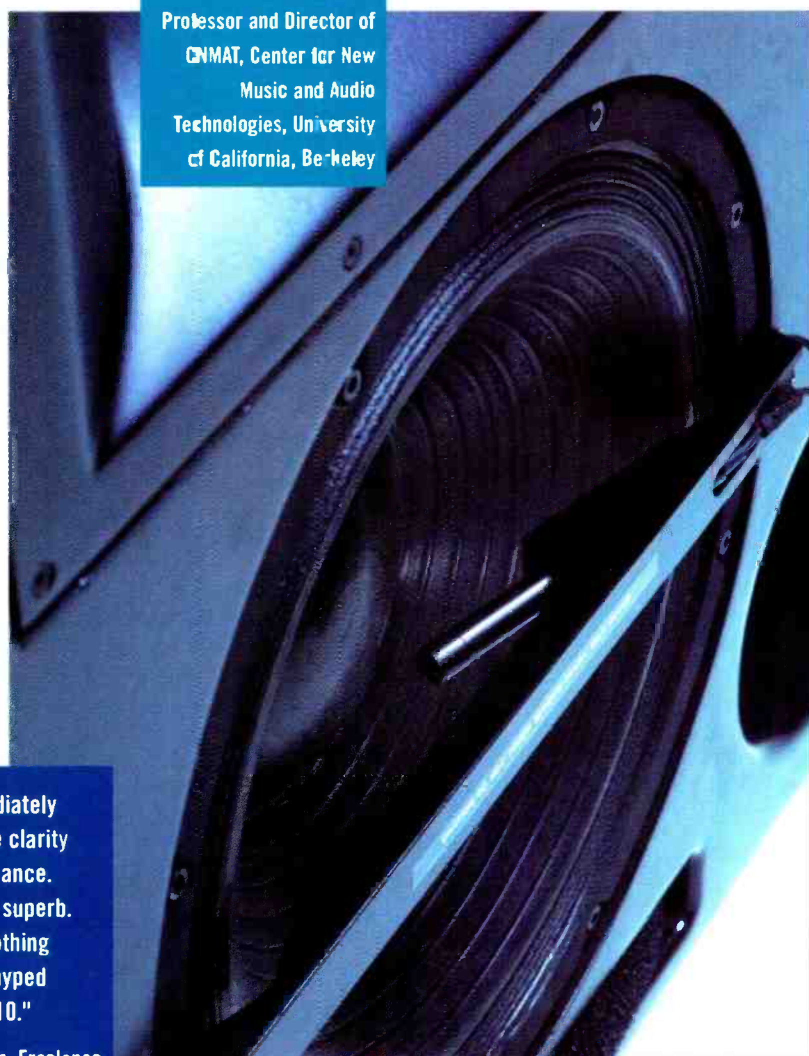
– David Wessel, Professor and Director of CNMAT, Center for New Music and Audio Technologies, University of California, Berkeley

Pressure Sensing Active Control (PSAC)™ (patent pending) tracks the driver's acoustic output and then employs high order feedback technology to precisely align output with the input signal.



"I was immediately struck by the clarity and tonal balance. Imaging was superb. There was nothing artificial or hyped about the X-10."

Jim Anderson, Freelance Engineer (Patricia Barber, Gonzalo Rubalcaba)

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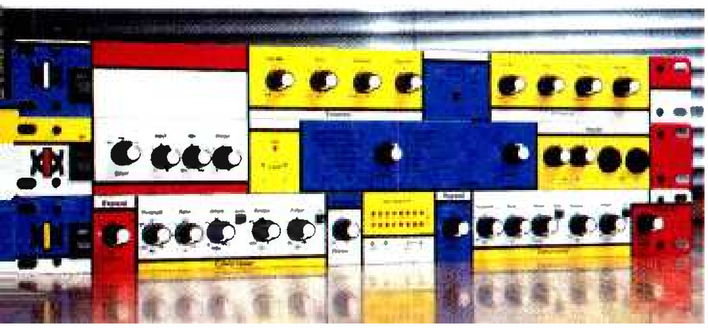
the green Channel 1 LED when in, and the red Channel 2 LED when out. 'Shurely shome mishtake?' Yes, both channels are active with the button in. Anyway, the controls for each channel are LOW MIX, TUNE, PROCESS and HIGH MIX. LOW MIX does nothing in its central zero position, but clockwise is labelled Tight and anticlockwise is labelled Soft, similar to SPL's Tube Vitalizer controls if I recall correctly. Turning the knob either way results in a massive boost of very low frequencies. The manual states, 'Most nearfield monitors are not capable of handling the bass produced by the EH-1901', which is possibly true and just plain silly, and it is easy to go overboard here. The other controls all relate to high frequencies, operating similarly to an Aphex Type B or C Aural Exciter, albeit with less of the nasty artefacts. The PROCESS knob works better than the Aphex' Drive control, as it seems to vary between a subtler Enhance mode and a less subtle Excite mode. It never gets harsh, just considerably brighter, depending on the HIGH MIX setting. By turning up the PROCESS knob the treble is boosted, above the frequency determined by the Tune control, variable from 1kHz to 8kHz, although in practice most of the frequencies boosted seem far higher. HIGH MIX determines the blend between unprocessed and treble-boosted signal. It is possible to use this subtly, but as with LOW MIX, it is easy to go too far. If your signal sounds like AM radio, or you don't know how to clean your tape heads, then you might have a use for this box. It is probably the cheapest unit of its type, and if you are considering a Vitalizer



or similar then this is worth a listen. However, some very bad-sounding results are all too easily possible here. If you have a mixer with a fixed high frequency band this will allow you to tweak frequencies you might not otherwise be able to reach, but I would think it far better to save your pennies towards the...

The ME-1903 multi-effects unit is the wackiest of the three. The manual is quite different from the other two units, seemingly written by someone aiming for a career in comedy writing, with not one knob or switch's description spared from some witticism. The unit itself deserves a prize for the largest assortment of audio connections, with the rear panel featuring stereo inputs and outputs on unbalanced RCA phono sockets, balanced-unbalanced jack sockets and XLR balanced connectors. In addition, the front panel features a pair of microphone inputs on jack sockets, accompanied by dedicated separate gain knobs for each one. So interfacing with this unit shouldn't, in theory, present any problems wherever you are. However, the results of changing the front panel Input Selector are not always what you might expect. This selects between line and mic levels, but in the Mic position there is not much gain from the mic inputs. However, switch to the Instrument position and both sets of inputs become active, with the rear lines oddly reduced considerably in level, but the front panel mic inputs' gain increased to a usable level. Very strange. There are also single controls for INPUT and OUTPUT LEVELS, and DRY-WET MIX, which are all self-explanatory, and these smoothly rotate unlike the clicky knobs on the other units. Finding the presets you want is another matter. To call up presets there are two (rather small and fiddly) 16-position stepped knobs. The first one selects a bank, and the second a program within that bank. The front panel lists the banks (unhelpfully in near-invisible black on blue lettering), but these bear little relation to the order of the lists in the manual, (that humour again?) or even sometimes to the actual programs. So it is all a bit hit-and-miss at first. The unit contains 256 presets, comprised of conventional reverbs, gated and reverse reverbs, along with delays, flangers, phasers, chorus and panners of varying speeds and widths. Many of the programs are combinations of two mono effects, accessed separately by the two inputs. It is well worth wiring up the inputs to two separate sends for this reason. However, some of the single effects are disappointingly mono only, or with many of the modulation effects, dual-mono, so that the only stereo going on is the pan positions of the input signals. Some programs simply blend the two inputs to mono. By the same token, many effects are 'true stereo', so you pays your money... It is all no doubt down to the maximum processing power available. So you need to check carefully whether you are getting a stereo output or just wasting one of those valuable input channels on your mixer. There are some notable goodies—the Fast Panner is a great Leslie-type effect, although, disappointingly, it is mono. There are a number of useful delays, although mostly quite short, but with varying amounts of regeneration, and a large selection of reverbs with different characteristics, with some smooth reverse and gated settings. The longer Chamber and Hall reverbs tend to clang a bit, and sound like fairly primitive digital algorithms. But hey, this could be the new 'retro' sound. And with a sampling rate of 31.25kHz this ain't a Lex 960L, but it is quiet and clean. I liked many of the flanger and chorus-type effects, many reminiscent of those from a Yamaha SPX90, and this box is probably quieter. There are no parameter adjustments whatsoever, but with plenty of programs to choose from it is not hard to find something interesting to use. For a novice, it is probably good training in making decisions. For sheer value for money this is the best of the three, and certainly the most fun, despite its idiosyncrasies.

The compressor and multi-fx unit are both ideal tools for anyone trying to build a first home studio system on a tight budget. The exciter-enhancer is useful for tarting-up substandard recordings, but its potential to do damage to recordings, not to mention speakers, is high. The build of these boxes is terrific for the money, although I'd have to say that the front panel colours are a triumph of marketing over ergonomics. They are not serious high-end tools, but I am sure there will be many satisfied customers, particularly in the younger DJ-studio segment of the market. Anyway, I'm off to spend a few more days fathoming out the ME-1903 effects banks. □

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DK-Audio adds spectrum analyser
A third-octave spectrum analyser has become a standard feature of the MSD600M/SA Master Stereo Display from DK-Audio alongside the FFT-analyser.

Whereas the FFT-analyser is widely used as an accurate measuring tool, the new 1/3-octave analyser shows the energy distribution of the signal. The analyser has 30 bars and a range from 20Hz to 16000Hz. Presentation is in colour on the TFT LCD, and may even be shown on an external VGA monitor. Maximum level of the signal is continuously indicated in a contrast colour 'behind' the actual presentation. Also new is the MSD200F/SA which is a version of the popular MSD200. The new unit is a larger cabinet with built-in universal 90-260V power supply and XLR I-Os and is a direct replacement for the well known stereo oscilloscope from the German company Filbig KG, which is no longer available.
DK-Audio, Denmark. Tel: +45 4485 0255.

Rane pres
Rane's NM84 network mic preamp offers 8 mic or line level XLR inputs and 4 line level outputs employing CobraNet technology. Eight direct outputs are standard. The NM48 network preamp is a 4-input 8-output variant on the theme.
Rane. US. Tel: +1 425 551 1812.

Trident-MTA three
New boxes from Trident-MTA include the Signature One which boasts a single channel mic preamp followed by high and low pass fully variable filters and a 4-band fully parametric EQ. Each band has a bypass. Other features include phantom power, phase reverse and a mic/line selector. The Signature Two is a dual channel mic preamp with 924 series EQ, phantom power and mic line switching. The ix-One provides 16 channels of mic/line preamp with individually switched phantom power, phase reverse and 20dB pad.
Joemeek. UK. Tel: +44 1803 321921.

DCA EQ
The DEQ series from Altair includes a dual/stereo 28-band digitally controlled analogue graphic also available in a slave version and controllable from REC remote software. SCAN automatic feedback suppression is available and 17 units can be chained in a system. REC software permits control of all functions of the equalisers and can integrate with other systems via MIDI.
Altair, Spain. Tel: +34 918 043 265.

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GARY RYDSTROM

With more award-winning films on his CV than any other film mixer, Gary Rydstrom probably also has the best home movies to prove it. **Rob James** talks pictures

FROM A CHILDHOOD in Chicago to the gentler climate of California, Gary Rydstrom has come a long way. From Super-8 movies at home through film school and the University of Southern California he started at Skywalker Sound in 1983. With a terrific body of work behind him he is now director of creative operations and a prolific sound designer and mixer.

What are the favourite film projects you have worked on?

Apocalypse Now. Seeing it recently, I was struck by its audacity. It's about Vietnam, but told like a fifties drug-trip movie. It's so full of hooks, red-meat opportunities to get inside the minds of characters or to be psychological about the sound. There's these internal sounds like Alan Splet did so well in *Blue Velvet* and *Black Stallion*.

Black Stallion is equally audacious, there's a whole section in the middle with no dialogue for reels at a time. The story of how the boy and the horse come to trust each other is told with just visuals and sound. There's a purity to it which is very nice.

I think I am happiest with *Saving Private Ryan* because the sound was allowed to play an emotional role. It wasn't a matter of designing sounds which never existed before, but orchestrating them so they would have some power, a sense of what it was like in battle or the emotions when tanks were coming into town, when you're going to have to fight. Working on *Private Ryan* felt like I'd lived through some battle myself.

Creatively, I struggled most with the *Jurassic Park* movies. Trying to create icons the way Ben Burtt does. It was daunting and pretty scary to be faced with so much that needed to be created out of 'whole cloth'. The hardest thing in sound design, is to try to create a living creature. Whether it's what Ben did with R2D2 or King Kong, things like that have a personality.

Do movies affect you?

On *Titanic* James Cameron said he wanted to make a movie where people could check to make sure their plumbing still worked. Cameron's right, you use movies as an emotional outlet. You feel a lot of emotions you can't or haven't had yet or maybe never will experience in real life. I've always been moved by movies, the scary thing is if you are moved more by movies than by real life.

You've got to use yourself as the canary. It's gut level to me, not intellectual. As much as people might think otherwise, most great movies work on a real gut level and emotional level. I've always considered that to be part of my job, just to react. But sometimes repetition can immunise you against emotion.

What projects were landmarks for you?

The first film I really had any design work in was *Cocoon*. It was really fun to play and see what I could make work. I used *Hot Spot*, an obscure film noir directed by Dennis Hopper, as an experiment to see how far I could go. I tried to be Walter Murch with it or Alan Splet, to see how much off-screen sounds could parallel the on-screen action and emotions. It was fun to use, as the intellectuals like to put it, for on-screen and off-screen sounds, what do they call it?

Diagetic.

That's it, diagetic and non-diagetic sounds. It might be a little bit too high-falutin a word for what we are talking about, which is what makes sound fun. You inject a lot of life if you can hear things you don't see. You can use that disconnect to treat ambient sounds like music and to create tension, a sense of nostalgia or beauty, whatever you are looking for. I used everything from the pace of the cricket chirping to the off-screen trains to try to parallel the action on screen.

How do you approach a mix?

It's nice to have it planned out so I have an idea where the stress, the important sounds and moments are. When I premix I'm also listening to the previous ones so everything is in context. I love the last premix with all the premixes playing straight across the faders so it sounds like a movie as much as possible.

We just did a movie called *The Legend of Bagger Vance*, directed by Robert Redford. It's an almost mystical golf movie, very subjective, all sorts of weird things happen. The real sounds may drop out, you lose the sense of reality and go into his brain and back out into reality. Those moments were all built into the premixes the way I thought they should work.

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I always elect to make all those decisions as you go, as opposed to the approach of leaving as many options as possible.

Do you usually start with dialogues?

We usually work in parallel rooms, dialogue and effects. Dialogue isn't something I ever thought I was great at. I've done it on movies where I've been the only mixer. My fellow mixers may want to kill me, but I can understand the single mixer approach, you are intuitively making everything work together, you can more accurately intertwine the three major elements. You don't have to go through the extra step of diplomatically arguing with other mixers to help you out. I did a documentary called *Into the Arms of Strangers*, dialogue, music and sound effects. It was very satisfying to be where the buck stops for all of those elements. A lot simpler than a *Titanic*, but creatively satisfying. It's counter-intuitive, but I found documentary more freeing, less tied to reality, than a lot of feature films. Anytime the sound can be subjective rather than objective, I'm happy.

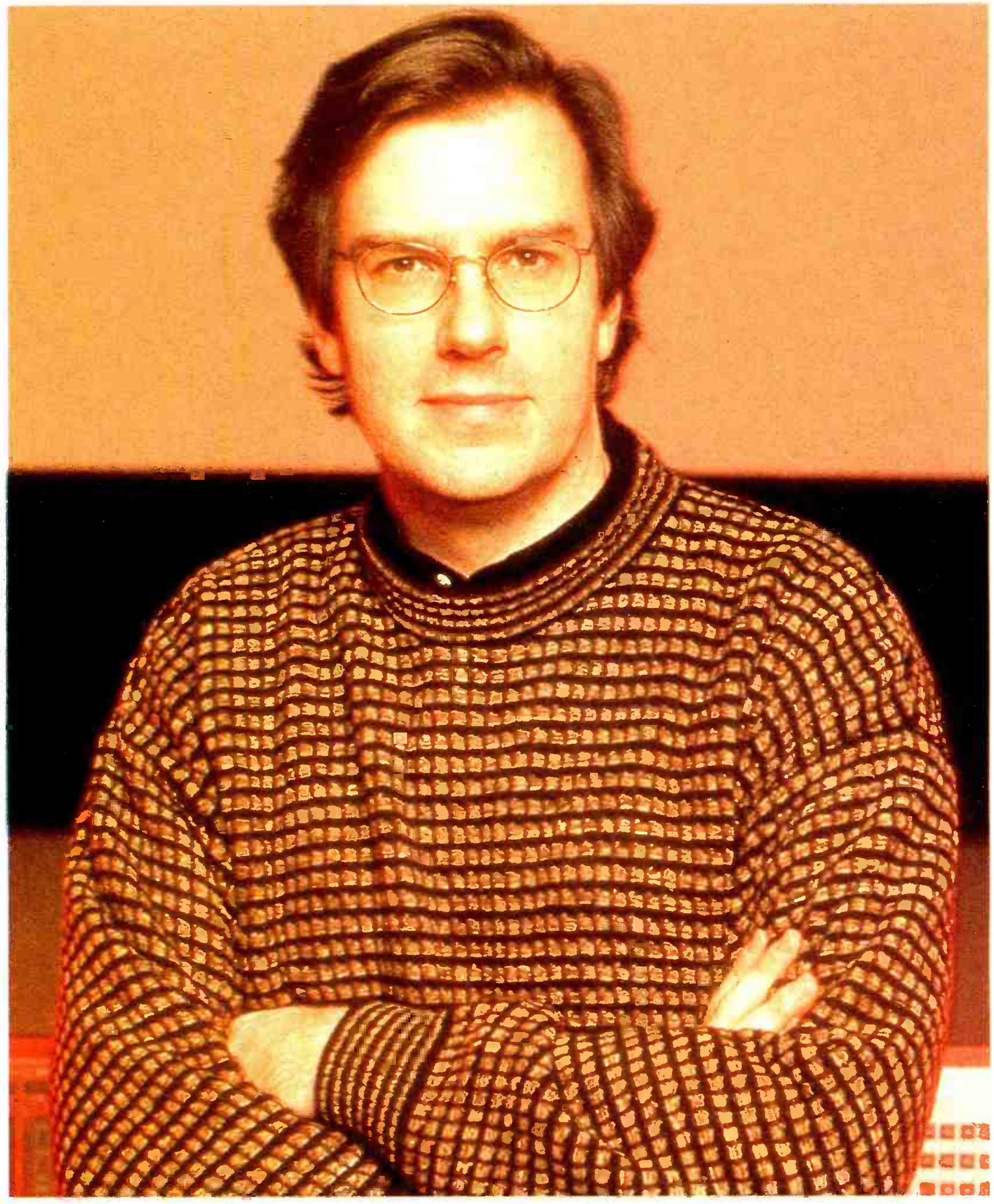
Do you keep settings in your head for certain things?

Sometimes I use a starting point and sometimes I just play. There are starting points especially with dialogue, you know you're not going to need any sound below 80Hz or above 8kHz. A close mic on a dialogue or on a Foley footstep is maybe a little bit unnatural so you do things to make it sound like you're not four inches away from the source. Experience saves a lot of time because you don't go through six ways of doing something you've tried before that don't work.

The danger, the tension, between experience and inexperience is you can use experience as a crutch. You have to try something that pushes the envelope. Every film I try to think of something, an approach that's just different. It's nice to store new things up, but with the fabulous crutch of experience behind you.

There was a lot of discussion about the dialogue levels in Jurassic Park?

Too low do you mean? Yeah, it's a tricky thing, it's probably on the edge. If you make a movie where you are trying to push the envelope and make it dynamic, theatres sometimes turn it down. Then what you



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thought was a comfortable dialogue level is too low.

We went for a fair amount of dynamics. I'm always struggling with that issue, obviously dialogue should be intelligible. But if you think about dialogue as a sound effect it has to fit within the scale. A human talking shouldn't be the same level as a T-rex roaring. I probably am guilty in that movie more than anything else of stepping on dialogue with one of my damn sound effects.

Are movies getting too loud?

Yes. Especially when digital first came in, I think we were all equally guilty. Sometimes it's just a bad call on the director's part, insisting, for a show to be dramatic they want to keep it loud. Sometimes it's just bad mixing. But I don't want to give up this fun new toy, this great new dynamic tool and I don't want to overuse it.

You have to make sure you're not asking the audience to sit through a movie that's the equivalent of sitting next to a 747 engine for two hours. You've got to give them more than that. Really pay attention to the quiet moments, peaks and valleys. I think that's a big part of our job and unfortunately that means convincing directors.

How do you hold attention?

The most under-used technique is silence. If the movie is loud, full and bombastic, often the audience will just sit back, eat their popcorn and take it in like a TV show. But if you go down to silence, simplify the track and really focus on what is happening, the audience will pay more attention.

The same goes for music. There are a couple of really emotional scenes in *Toy Story 2* and the more emotional the moment, the more reason to take the music out and everything of mine too. You just play it with the simple silence of a character you like, having to make a tough decision.

Music over emotional peaks can flatten them. I like the way the music was spotted in *Private Ryan*. The intense moments had no music. Music was a transition out of those scenes, soldiers just walking, or the after effects of someone's death. It's kind of like getting two for one. The moment itself and time to reflect.

Without blaming anybody, the sound is often not all that well planned. Out of fear a film-maker ends up having the composer put music from start to finish.

People do their aspect of the movie without really hearing what everyone else is doing. Everyone goes for the same moment, it's like a car wreck.

In the, getting somewhat tiresome, action genre, people try to one up each other. I've heard of one film where any time there was even a micro-moment of silence the director would have somebody fill it. It becomes like sound sparkle, filling every crack with sound. That comes out of fear of losing the audience.

Terminator II was a classic action film, but James Cameron did a really good job of planning it so it has peaks and valleys, moments of quiet, slowness and silence so the big moments and the movie seem bigger. Spielberg is the same way. When he does those kind of films like *Jurassic Park*, he knows how to build con-

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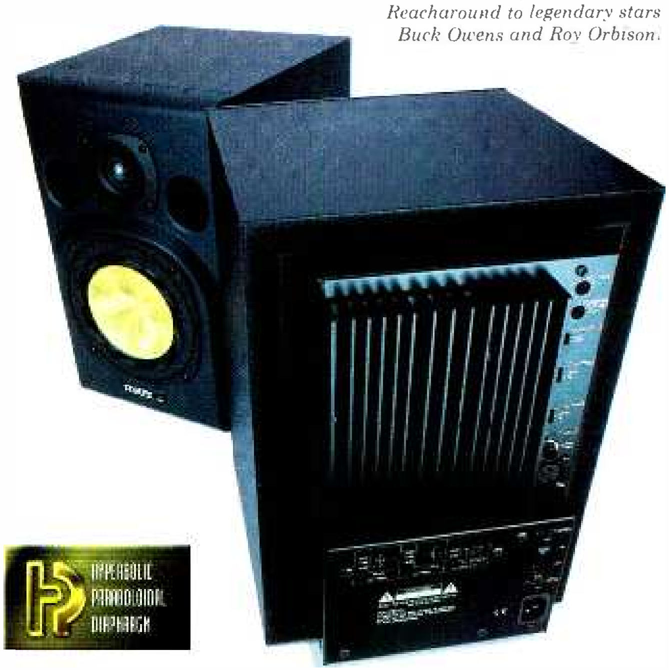

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trast into every element of the film.

How do you use booms?

More now than before. I just did a DVD remix of *Terminator II*. One of the main things I did was to add more boom. Originally, we did the full final mix and didn't deal with adding any boom until the very last pass, kind of like that last little bit of dessert! That way, you're not entirely dependent on the boom. A lot of movies now, if you take the boom away, there's nothing left.

Now I go the opposite way and start adding boom from the premixes on, a cleaner way of doing it because you're just subharmonically generating one isolated element. You're not throwing general crap down there, you're actually being more precise about it.

How do you know when you're getting it right?

It's almost as if the sound and the image snap together. It feels 'of a piece', the moment when it sits in there and you realise, oh, that's it. Like wearing 3D glasses, when they come together, you know it.

How is the process changing?

One bottleneck is interchanges. Films change radically up to the last minute. A frustration for directors is, with a fairly simple picture change, how complicated it is to ripple through everything. I'd love to see something that would take the picture change, talk to the workstations and consoles, and make a quick non-destructive attempt at making the change. This would involve the systems knowing the difference between conforming ambience or a line of dialogue.

At some point you have to be in a big room for acoustics, but there's no reason decisions about panning, EQ, balance and everything else shouldn't be made in the editorial room. By 'sound designing' that's essentially what we mean here. So, our approach is we're building sound design rooms, bigger than our traditional editorial rooms.

There are still attitude issues, mixers who don't want sounds brought to them with judgement of level, let alone EQ or panning.

The tricky thing during these transition years is how to use everyone well. But the editor of the future will be skilled at both. It's a bigger job, mixing preparation as well as editing.

And the big consoles?

I still haven't found the perfect console. You wish it could be like in those cheesy 'B' horror movies, you'd put these little steel helmets on the different consoles and blend them.

The Capricorn's the first digital console we got here. It's been five years or so now. It allowed us to do things we'd never been able to do before. The first feature I mixed all the way through on the Capricorn was *Titanic*. In the old days when we tried to do a fix we'd either put fix units up on dubbers or try to spin something in from a tape deck. After that, this was amazing, just to have the freedom to say, 'well that creak doesn't really work let's find another one' and every time you go back it'll be there, fully automated. It gave such flexibility it was astounding. Before, you'd spend most of the time worrying about getting in and out.

It's not just a revolution for technology's sake. It's allowed us to do better work, if we do a bad job, it's really our fault, not the technology.

Do you have favourite toys?

My favourite design toy by far over the years has been a Synclavier. At the final mix I could call up a sound and play it on the keyboard. Lock it to time code and do a lot of sweetening. It's just a great creative tool. I've yet to find the perfect replacement for it. Mostly because of the antiquated but useful interface Synclavier built, which is partly why it cost a ton of money. You get a keyboard with a thousand buttons on it and once you learn them, they give you very quick access to things. Nowadays interface usually means another computer screen, and computer screens aren't as fast as real buttons no matter what you do.

How will we be mixing movies in 20 years time?

Mixing is going back to an idea that was around a while ago, but never really took off. Integrated source and console. Film consoles have yet to talk to the workstations, but there is no reason this won't be happening soon. What it requires is to get companies that make the best of the workstations and the best of the consoles to talk to each other.

Plug-ins become the other element of a consolidated mixing-workstation system. You want to have plug-ins that are applicable to both.

How do you spend your time as director of creative operations?

Well all that means is essentially 'in addition to'. While I'm doing films, I at least keep an eye on what we're doing as a company and try to have some creative vision for both the process we use here and the people that do the work. Trying to make a reputation for doing good work. Ideally that brings you more work.

Do you still have ambitions?

It would be nice to see how far sound can go, to see what sound can add to a movie. The more emotional and psychological use of sound is what appeals to me. The kind of stuff which was used so well in *Apocalypse*. It's an interesting realm, I'd like to get into that more, where sound is not so literal but more purely dramatic.

Theme park ride sound design

I DID A LOT OF THEME PARK STUFF FOR DISNEY. They have completely wild and elaborate sound systems in the theatres and control over it, six channels and up. What pushed me was doing these Disney shows where we had 12 channels or 15 channels.

We did a *Muppet Show* for Disney where the Swedish Chef was the projectionist. He was in the back, always shouting at the audience and it was fun so I always remembered that and that's why, on *Episode One of Star Wars*, we tried to at least add one more channel and have a rear channel from Dolby EX.

With most of the Disney things we try to emulate the layout of the ride in our studio. A long time ago we did Startours, a flight simulator ride at Disneyland. We took the actual sound system that was in the ride and set it up in our studio. But then you do a final pass on the ride itself. It's a luxury and a pretty big one to mix in the room and the only room it's going to play in.

It's great to be able to hear it with an audience and fine tune it. In Disneyworld's *Honey I Shrunk the Audience* you are supposed to believe that some mice have been let go from the stage and are making their way back through the theatre. They have a great little device which makes you feel the whip of a rats tail on the back of your foot. As part of the design there are a series of speakers on the floor, so we put the sound of rats waving from the front to the back of the theatre which worked great when we were just in there alone. But every time we'd bring an audience in, they screamed so loud, you couldn't hear the rats squeaking. If I brought the level of the effect up to the point where you would hear it, it was abominably loud, so we had to give up.

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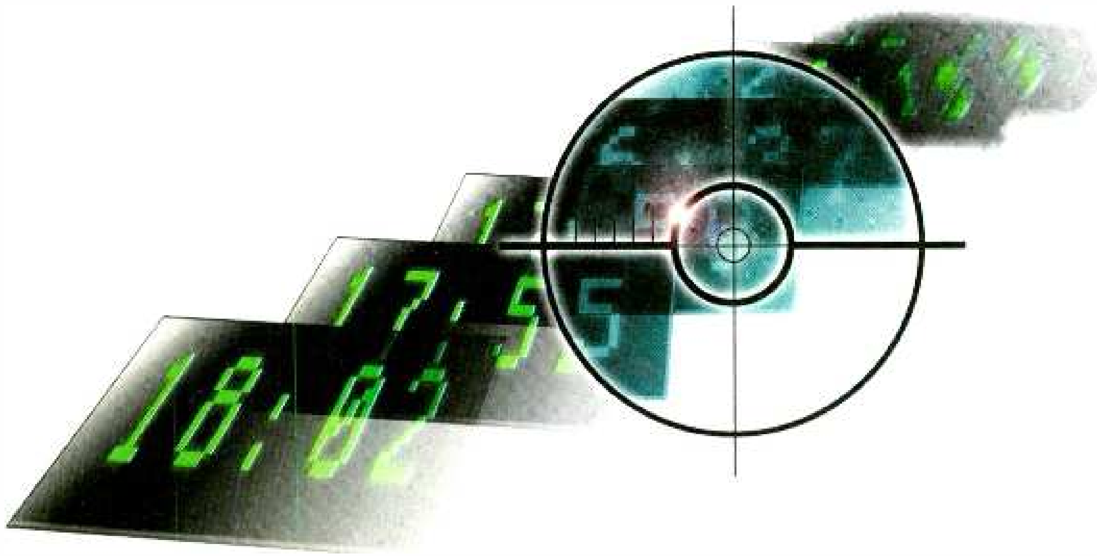
Never say Never again...



Your working hard, the new movie trailer's final mix is due tomorrow by 10 am, everything is going great and then you hear it, somebody made a mistake, you think it might be about twenty edits back, but you don't have a backup!

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In this ever changing fast paced world of studio project deadlines one mistake like this can quickly add up to hours, or even days of extra work costing you both time and money.

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But Nuendo goes even further. You can process any audio segment with a wide variety of powerful edit operations and plug in effects with the integrated Sample Editor, or even from directly within the Project or Browser windows. And as each segment has its own individual off-line history which is stored with the Project, the edits you make today can be selectively removed, or even re-applied with new parameters within seconds tomorrow, giving you total freedom of choice.

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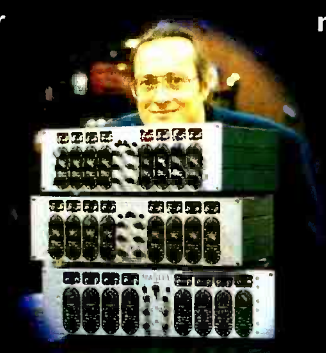

Craig 'HUTCH' Hutchison designed these monsters... The MASSIVE PASSIVE is a two channel, four band equalizer, with additional high pass and low pass filters. "Passive" refers to the tone shaping part of this clever new EQ design not using any active circuitry. Only metal film resistors, film capacitors and hand-wound inductors sculpt the sound, kinda like a Pultec EQ on hyper-steroids. Super-beefy, hugely-high-headroom Manley all-tube make-up gain amplifiers deliver your tunes into the next realm. You'll need to experience this.

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DVD-AUDIO

The limitations of CD performance have conspired with manufacturers' preoccupation with new formats to bring us DVD-Audio. **George Cole** explores its history and politics

WHEN THE DVD FORMAT was first conceived, it was defined by its name: Digital Video Disc. Very quickly, however, it was transformed into the Digital Versatile Disc. The new name was designed to show that DVD could be used to carry a wide range of entertainment media, including audio, video, data and games. The first DVD format to reach the consumer market, DVD-Video, carries movies and music videos on a CD-sized disc. DVD-Video was launched in Japan in November 1996, in the US the following year, and in Europe, in spring 1998.

DVD-Video has since become the fastest-selling consumer electronics format of all time, reaching world-wide player sales of more than 10m units just three years after its launch. Formats such as the colour television, VHS video recorder and audio CD took much longer to reach this milestone. Some DVD-Video players are now selling for the same price as a VHS recorder, but those consumers who like to keep up with the latest developments in AV technology are now going to have to make room for another type of DVD offering—the DVD-Audio (DVD-A) player.

DVD-Video players can play a variety of audio formats including, Linear PCM (LPCM) which conforms to the audio CD Red Book standard (44.1kHz sampling, 16-bit encoding), as well as multichannel (5.1) surround-sound formats like Dolby Digital, DTS and MPEG2. DVD-Video players can also offer LPCM which exceeds the existing audio CD standard, with 48kHz or 96kHz sampling rates and 20 or 24-bit quantisation. In fact, a number of specialist record labels have launched audio titles designed to take advantage of DVD-Video's superior LPCM performance. But the latter feature should not be confused with DVD-Audio, which offers much more, both in terms of audio performance and features.

DVD-Audio background

The DVD format was originally developed by a consortium composed mainly of consumer electronics companies, and now known as the DVD-Forum. The companies included Toshiba, Pioneer, Sony, Philips, JVC, Matsushita (Panasonic-Technics) and Thomson. In December 1995, the consortium set up the Working Group 4 (WG4) to develop a standard for a new audio format that would offer much better performance than the audio CD, as well as new features like multimedia. At the same time, the music industry formed the International Steering Committee (ISC) to put develop a wish-list of features content providers would like to see on a new audio format. Regular meetings between WG4 and the ISC resulted in all items on the wish list being incorporated into the DVD-Audio specification.

In November 1997, WG4 membership was expanded to include companies from many other fields, and over 40 companies joined it including, Intel, IBM, C-Cube Microsystems, Adaptec Japan, EMI, Dolby Laboratories, Digital Theatre Systems (DTS), Nimbus and Sonic Solutions. Version 1.0 of the DVD-Audio standard was set in February 1999, with the first players set for launch pre-Christmas that year. But as we shall see later, the DVD-Audio's launch was delayed until summer 2000.

DVD-Audio follows the same basic disc specifications of DVD-Video, that is a 12cm disc that is 1.2mm thick. The discs are created by bonding two 0.6mm disc substrates. Note that 8cm discs are an optional extra and could be used for portable audio players and other devices. DVD-Audio discs can be single-layer (4.7Gb capacity) or dual-layer (8.5Gb). The second layer could be used to extend playing time or adding multimedia content. It could also be used for hybrid discs that have a single layer of DVD-Audio content plus a second layer of Red Book audio. This means that a hybrid disc could play on both DVD-Audio and audio CD players.

Although the DVD-Video specification includes provision for double-sided discs, DVD-Audio is a single-sided format. Like the audio CD, DVD-Audio's standard playing time is 74 minutes, but this can be extended to 25 hours of mono sound using a technology developed by the UK audio company Meridian (see below).

DVD-Audio offers six audio sampling rates: 48kHz, 96kHz, 192kHz, 44,1kHz, 88,2kHz and 176.4kHz. There are also three levels of quantisation, 16-bit, 20-bit and 24-bit, giving 18 possible permutations. The format also offers two to six hi-fi audio channels (Table 1), the latter being used for multichannel sound. DVD-Audio's maximum bit rate is 9.6Mbps, higher than both DVD-Video and audio CD. A comparison between DVD-Audio, DVD-Video and CD audio specifications is given in Table 2. DVD-Audio offers a 96kHz frequency response and dynamic range of 144dB. The audio information encoded on a DVD-Audio disc can be up to 1000 times greater than on an audio CD.

DVD-Audio discs can also offer additional content such as text information (such as song lyrics) for displaying on a TV screen or an LCD screen on a DVD-Audio player, audio commentary, video clips and URL addresses for adding web links to the content (Fig.1). DVD-Audio has been designed to be as flexible as possible, allowing engineers and producers

Panasonic DVD-A

Amazingly, DVD-Audio can store a rich plenitude of valuable information above and beyond its high-density audio signals. This includes album and track titles, artist name, lyrics, liner notes and more, displayable as text or menus offering interactive functionality. Full-motion video, still images, and computer data can also be included, so that video clips can be viewed on a connected monitor or Internet website home page URL addresses embedded in the disc can be accessed to download artist concert information.
A leading edge copy protection system incorporating encryption and digital watermark technology is employed to guard against illegal copying and pirate editions, giving the content provider copyright management capability.

DVD-Audio Contents

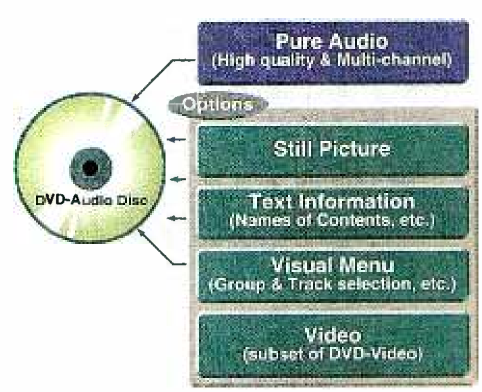

Fig.1

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Playback times for different discs (Linear PCM)

Combination of audio contents	Configuration	Playback time per disc size			
		12cm disc		8cm disc	
		Single layer	Double layers	Single layer	Double layers
2-channel only	48kHz, 24 bits, 2 channels	258 min.	469 min.	80 min.	146 min.
2-channel only	192kHz, 24 bits, 2 channels	64 min.	117 min.	20 min.	36 min.
Multiple-channel only	96kHz, 16 bits, 6 channels	64 min.	117 min.	20 min.	36 min.
Multiple-channel only	96kHz, 20 bits, 5 channels	61 min.	112 min.	19 min.	34 min.
2-channel & multiple-channel (same contents)	96kHz, 24 bits, 2 channels + 96kHz, 24 bits, 3 channels & 48kHz, 24 bits, 2 channels	43 min. each	78 min. each	13 min. each	24 min.

Table 1

Panasonic DVD-A

to create the mix of media they want.

Not everyone listening to a DVD-Audio recording will have a multichannel system in their home, so the signal will need to be down-mixed to two channels. Another system, Smart Content (System Managed Audio Resource Technique), also allows studio staff to control the audio playback by saving mixdown coefficients as control information to a data channel on the DVD-Audio disc. This means that when a multichannel DVD recording is played on a 2-channel system, the listener gets to hear the sound exactly as the artist intended in a stereo environment.

DVD-Audio has been designed to offer much creative flexibility. A producer could decide, say, to have equal or split sampling rates and bit depths for the front and rear channels, to use a separate 2-channel mix or a programmed multichannel fold down, or to only use LPCM or add another audio format such as Dolby Digital or DTS.

Protection and Regional Coding

Not surprisingly, DVD-Audio uses powerful copy protection technology including data encryption and watermarking. DVD-Audio's launch was delayed when in October 1999, a 17-year old Norwegian boy broke the Content Scrambling System (CSS), developed by Matsushita and Toshiba to protect DVD-Video titles from piracy. The teenager then decided to publish his exploits on the Internet, with the result that DVD-Audio's launch was delayed while the DVD industry developed a more powerful encryption system. The new system, initially called CSS 2, was originally designed for DVD-Audio, but will now be used by all DVD formats.

The new copy protection system was completed by the end of December 1999, and was renamed because the CSS brand had been tarnished. CSS2 is now known as CPRM (Copy Protection for Recordable Media) and CPPM (Copy Protection for Prerecorded Media). The complete system is called 4C, after the four companies that developed it, Toshiba, Matsushita, Intel and IBM.

Watermarking involves burying a fragile signal in the audio waveform. The presence of the watermark is detected by the DVD-Audio player before it will play the disc. If the audio signal is copied or compressed, the watermark vanishes, and when the copied disc is placed in a DVD-Audio player, it registers that the watermark is missing and refuses to play the disc. The watermarking system developed for DVD-Audio has proved controversial, with some hi-fi listeners believing that it affects the quality of the playback signal. But DVD-Audio companies say that the system has been cleared by 'golden-eared' listeners employed

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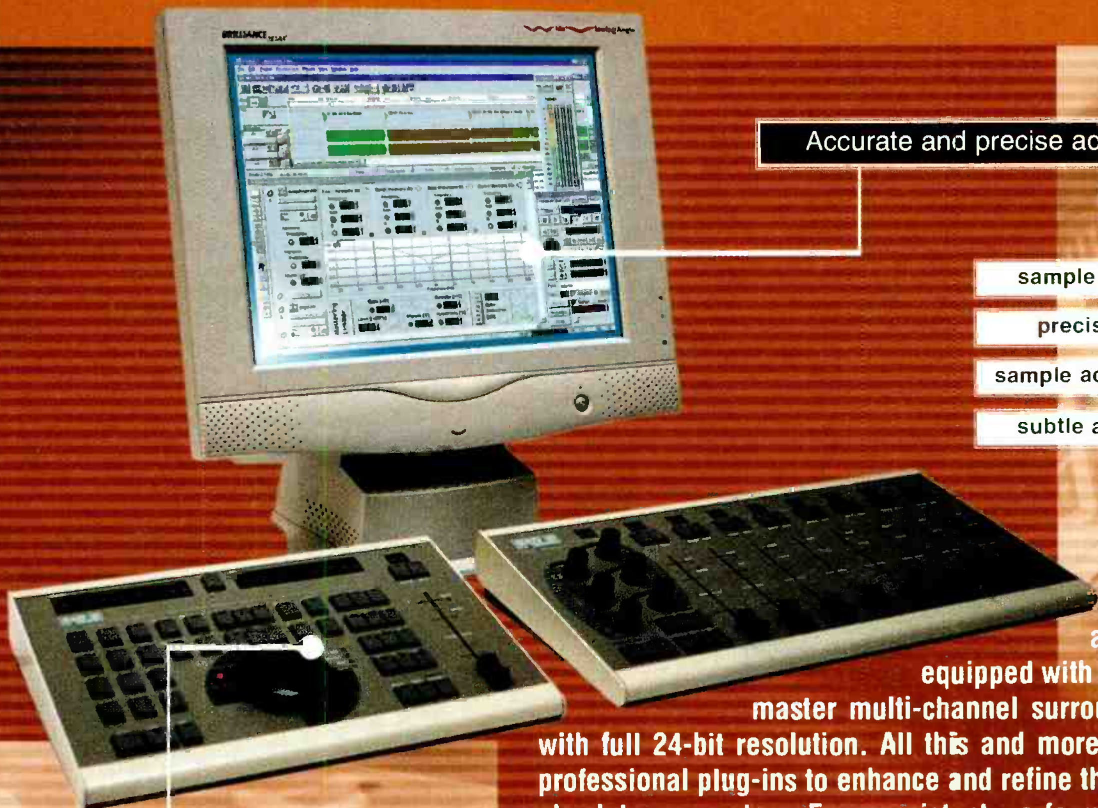

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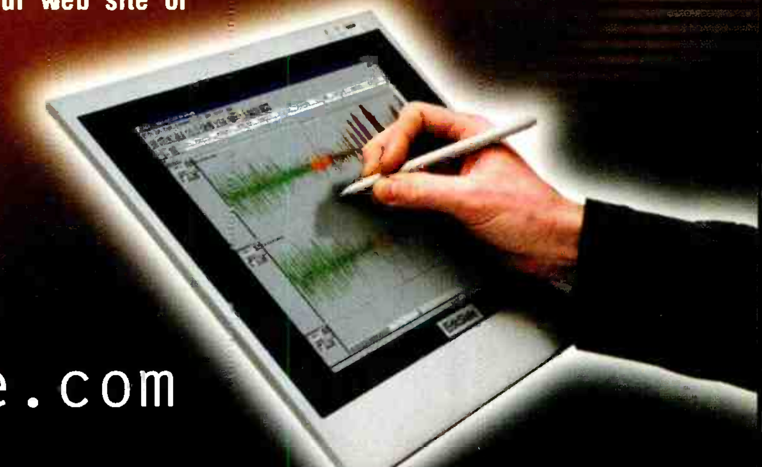

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DVD-Audio Major Specifications vs DVD-Video & CD

Item	DVD-Audio(single layer)	DVD-Video(single layer)	CD
Capacity	4.7GB	4.7GB	650MB
Size	12cm, 8cm	12cm, 8cm	12cm, 8cm
Channels	6 max	8 max	2
Frequency response	DC~96kHz max	DC~48kHz max	5~20kHz
Dynamic range	144dB	144dB	96dB
Recording time	74 minutes or more in all modes (including 96kHz/24 bits/6 ch and 192kHz/24 bits/2 ch)	133 minutes average	74 minutes
Max transfer rate	9.6Mbps	6.1Mbps	1.4Mbps
Audio signal			
Audio signal format	PCM	Dolby Digital, MPEG, PCM	PCM
Audio options	Dolby Digital, DTS, MPEG, etc.	DTS, SDDS, etc.	—
Sampling rate	(2 ch) 44.1, 88.2, 176.4kHz or 48, 96, 192kHz	48, 96kHz	44.1kHz
	(multi-channel) 44.1, 88.2, or 48, 96kHz	48kHz	—
Quantization	16, 20, 24 bits	16, 20, 24 bits	16 bits
Smart Contents	6 ch→2 ch		
Functions, etc.			
Still images	Yes	Yes	No
Real-time text	Yes	Yes (subtitles)	Yes
Regional code	No	Yes	No
Simplified play data	Yes (SAPP)	No	Yes (TOC)
Interactive features			
Liner note display	Yes	Yes	No
Message play	Yes	Yes	No
Website access	Yes	No	No
Lyric link	Yes	Yes	No

Table 2

by the record companies to evaluate the system.

The DVD-Video format uses a Regional Coding system to control the global distribution of DVD titles. The system adds ID flags to DVD titles which identify the territory they were produced for. This means that if someone tries to use a disc created for the US market (Region 1) on a European player (Region 2), the disc will not play.

Hollywood insisted on Regional Coding in order to protect its carefully controlled release windows for films and packaged video media. In many cases, a blockbuster film appears on DVD in the US before it has even reached European theatres. Although it would have been technically possible to have added a Regional Coding system to DVD-Audio, the music industry rejected this option, so all DVD-Audio discs will play on all DVD-Audio players regardless of where they were bought. Many believe that in a world of the Internet and global shopping, film companies will have to adopt the same model.

Lossless Packing

A disc data capacity of 4.7Gb sounds a lot, especially when you consider that it is seven times greater than for an audio CD. However, multichannel hi-fi sound, and features such as video clips and graphics can soon eat-up all of the available data capacity. In fact, a 5-channel PCM audio track using 20-bit encoding and 96kHz sampling would have a maximum playing time of just 65 minutes on a 4.7Gb disc. And a 6-channel audio track with 24-bit encoding and 96kHz sampling would require a bit rate of 13.8Mbs, well above DVD-Audio's maximum data rate of 9.6Mbs.

The answer is to use a data compression system, and a small UK company developed the system used by DVD-Audio. Meridian Audio, based in Cambridgeshire, created the Meridian Lossless Packing (MLP) system, which is a mandatory part of the DVD-Audio player specification. There are many data compression (or more accurately, data reduction) formats around including, Dolby's AC-3 (used by the Dolby Digital format), DTS, MPEG and Sony's ATRAC used on the MiniDisc format. These use algorithms designed to mimic the way the human ear and brain perceive sound. Some audio frequencies are hidden or masked by louder sounds of the same frequency, and so (in theory anyway) they can be discarded, reducing the amount of data that needs to be encoded and stored.

These so-called perceptual coding systems work remarkably well, although hi-fi purists insist that even masked sounds contribute to the final fidelity of the audio signal. It was certainly clear that a high-end hi-fi system like DVD-Audio could not use a lossy or data reduction system to increase playing time or disc capacity.

The story of MLP begins in 1992, when Meridian's chairman, Bob Stuart and friends Michael Gerzon and Peter Craven worked on audio coding techniques. In 1994, Stuart was asked to become the new chairman of the Audio Renaissance for Audio ARA, an organisation set up to develop a new standard for audio disc technology. It was also around this time, that proposals for a new high-density disc (which would eventually become DVD) were being made. Stuart felt that such a disc would make it possible to encode all the sounds a human ear could hear. ARA formed a technical committee and in April 1995, circulated a document proposing a new audio standard which included LPCM coding up to 96kHz, 24-bit and 6-channel sound. At first there was hostility to the proposal, especially from Japanese companies who remembered the fiasco over quadraphonic sound.

But Stuart was convinced there was great potential for offering high-quality multichannel sound on a disc, and Meridian developed a lossless coding system targeted at DVD-Audio. The system was demonstrated to the ISC and the RIAA, and the breakthrough came when Dolby and Warner Music expressed support for MLP. Once the principle of lossless coding was accepted, Meridian found itself

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

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

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

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Compatibility between Disc and Players

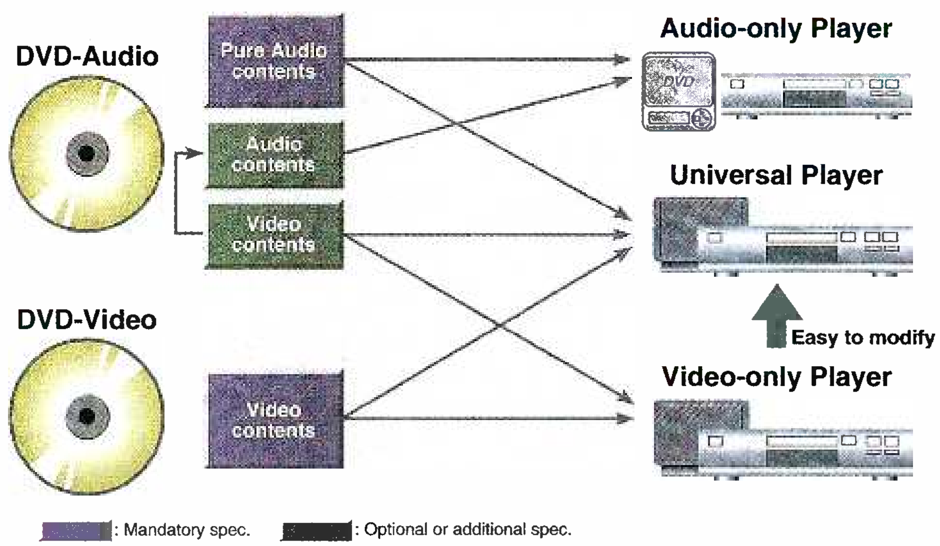

Fig.2: Disc and player compatibility

Panasonic DVD-A

facing stiff competition from rival systems, and a competition to select the best lossless coding system for DVD-Audio took place in 1998. MLP emerged as the winning system.

MLP makes use of the fact that high-rate formats like DVD-Audio carry more information than is necessary for the human ear, as well being be beyond the capabilities of modern microphone and converter techniques. This means there is redundancy in the DVD-Audio audio stream. MLP uses an audio coding system that detects redundancy and packs the audio into a smaller space. However, the coding allows a decoder to recover the original signal, bit-for-bit. MLP uses three methods to reduce the data capacity: lossless processing and lossless matrixing, which reduces the correlation between channels; lossless prediction to reduce inter-sample correlation (waveform prediction), using a large palette of filters, and third, Huffman coding.

Meridian now sells professional MLP encoders, which run on PCs using Windows 95, 98, NT4 or 2000, with a 100MHz or faster Pentium processor, a minimum of 16 Mbyte RAM and 5Mbyte of hard disk space, as well as extra space for the created MLP files. (The MLP encoder costs £5,875 inc VAT. For more details on the MLP encoder go to www.meridian-audio.com/m_mlp_in.htm)

Players and Prospects

There is provision for several different types of DVD-Audio players. Some are pure audio players, and even within this group, several types are available. Some DVD-Audio players will have built-in multichannel decoders; others will simply play down-mixed 2-channel audio. However, most of the players designed to be compatible with DVD-Audio are so-called Universal Players, which can play both DVD-Audio and DVD-Video discs. These players will also play audio CD discs and some will even read Video CD discs.

However, DVD-Audio is only partial compatible with DVD-Video players, even those offering 24-bit, 96kHz LPCM audio. DVD-Video players cannot play DVD-Audio files but if a DVD-Audio disc also carries a video clip, this will play on a DVD-Video machines (video clips are likely to use Dolby Digital or DTS audio). Fig.2 shows the compatibility between various discs and players. So far, Panasonic and Technics have launched DVD-Audio/Video players and mini-systems in the US, and DVD-Audio should reach Europe this autumn. JVC UK has a Universal DVD player in its latest product catalogue. Not surprisingly, these early DVD-Audio players command quite a price premium over standard DVD-Video players, and sell for around £650–£850. However, prices are expected to fall and in time, it will make more sense to purchase a Universal DVD player than a bog-standard video player.

Supporters of DVD-Audio say the time is right for a new digital audio standard. They argue that the audio CD is now over 20 years old and that technology has moved on considerably since its launch. But others argue that there is little suggest that most consumers are dissatisfied with the performance of the audio CD, and that DVD-Audio will only appeal to hard-core hi-fi enthusiasts. Critics add that most homes will not have an expensive, multichannel audio system to take advantage of DVD-Audio's superior sonic performance.

Another fly in the ointment is a VHS vs Betamax-type battle that has broken out between companies supporting DVD-Audio, and Sony and Philips, which have developed a rival and incompatible system, Super Audio CD. SACD offers a similar performance to DVD-Audio, although it uses a different audio encoding system known as Direct Stream Digital. As DVD-Audio emerges, Sony has launched second-generation SACD players designed to appeal to a broader base of consumers.

The appearance of two rival 'super CD' systems on the market together has not helped matters, and nor has the fact that most major record companies have been slow to release titles for either format although (not surprisingly, Sony Music is actively supporting SACD). Many believe that a new generation of 'super universal' players will emerge, with the ability to play DVD-Audio, DVD-Video and SACD discs. But it remains to be seen whether even this development will convince many consumers to switch from today's CDs to the new generation of audio discs. □

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COOL, BUT IS IT COOL ENOUGH?

The long-awaited DVD-Audio comes late to the entertainment landscape to find even newer technologies waiting in the wings. **Dan Daley** asks: Is it too late?

THE LONG-HELD PROMISE of high-density audio formats has excited audio industry professionals like nothing since the introduction of console automation systems 20 years ago. Just one component of these new formats—multichannel surround—has become a cottage industry in itself, with its own trade magazine and numerous seminars. The option of 8-channel output buses on new consoles has become the equivalent of power windows on new cars, and speaker manufacturers slather over the prospect of tripling unit sales. Furthermore, the conception of the basic DVD format, which increased the real estate that a standard-sized optical disc could hold to upwards of 17Gb, against the 750Mb of the conventional audio CD, was contemporaneous with the movement of pro-audio equipment into the 96kHz, 24-bit range. Thus, it seemed that the stars were in alignment for high-density audio's roll-out and that the new high-density formats and the pro-audio business were a match made in heaven.

But such marriages are never as glossy on the inside as they are on the exterior. And since DVD-Audio is the offspring of the broader high-density optical media format, a little family history is useful.

DVD itself had a difficult birthing, with a number of major movie studios, including Fox, holding off on committing to it until the last moment. Their concerns were twofold, and both have an impact on high-density audio's fortunes: piracy and competition with existing formats.

The CD was always a loaded gun that the music business had pointed at its own foot. Each digital disc was a potential pirate master, and that's precisely what they were used for globally, a situation that has only been exacerbated by the arrival of affordable CD-R hardware and media and which has turned music piracy into a $15bn-a-year problem world wide. The film business, which was the first customers of the DVD format and which were deeply involved with its development, were quite aware of this and that obstructed unanimity in their ranks. It's also worth noting that the DVD format has over 16 participating major patent holders, some of which include major content developers, such as Warner Studios. In contrast, the CD was launched in 1982 by two main protagonists, Sony and Philips, in a situation where fewer cooks made for at least a faster stew. The corporate pair obviously hoped that lightning would strike twice when they formed their own proprietary high-density audio venture, SACD, three years ago, based on Sony's Direct Stream Digital technology. The film industry wanted some guarantees regarding copy protection, which led to the development of the CSS encryption codec and to the decision (now increasingly ignored) to include regional coding on the discs.

Secondly, though less well-documented, is the fact that the VHS business was booming throughout the mid 1990s (and still is, despite DVD-Video's initial success), when DVD was being discussed and prepared for introduction. There was quiet concern that a second format, digital though it may be, would confuse consumers. In addition, some factions said that releasing a non-recordable format to consumers used to being able to record on their VCRs was doomed to failure. Record labels in the US, where all DVD formats would be released first, were particularly wary about this as they watched their industry grow from the $10bn mark in mid-decade to over $14bn this year, a 40% increase in sales, almost all of it based on standard CDs. That more than anything explains the reticence on the part of major record labels to commit to high-density formats, since piracy has always been an issue and a longer word length and higher sampling rate weren't going to dramatically increase piracy. In fact, the only real commitments until very recently came from small, audiophile or dedicated multichannel start-up labels.

Thus, when DVD-Audio began to emerge from the corporate cocoon last year, it faced a somewhat hostile, or at least indifferent landscape. It was made even more so when a group of Norwegian hackers last November cracked the CSS2 encryption code, which was designed specifically for DVD-Audio, and posted it on the Internet for free. A new encryption codec, Content Protection for Recordable Media and Pre-Recorded Media (CPRM/CPPM), has since been developed by the 4C Entity group of companies—Matsushita, Intel, IBM and Toshiba, for DVD-Audio.

Those who had been eagerly awaiting the DVD-A or SACD high-density formats won't take comfort from this analysis of the long-range market, by the UK-based international research consultancy Understanding & Solutions, which tracks the optical media market. According to an extract from a report issued earlier this year, 'In the short term the overall impact of either DVD-Audio or SACD is likely to be minimal. The global CD market accounts for well over two billion units, with nearly a billion of these sales from the USA alone. Combined DVD-Audio and SACD [disc] shipments are anticipated to account for only 6% of the total music market in five years' time. In the short term, these new formats are more likely to have an incremental rather than substitutional effect on the music market.'

And, the report goes on to state, 'The majority of consumers are not renowned for adopting 'incremental' formats offering subtle improvements—S-VHS and DCC are two prime examples of this.'

If high-density audio had a difficult runway to approach, it also found its destination crowded with new faces, technological celebrities, as it were, in the form of MP3 and search engines like Napster, which rendered DVD-Audio a sort of has-been before the

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fact. Computer file-based music ignited the most desirable demographic in the entire music industry: those between 14 and 21, the most computer-savvy, economically affluent and terminally bored cohort in the US, which saw MP3 and Napster and Gnutella not so much as statements against the corporate music machine as just plain cool. And cool, as trends forecaster Faith Popcorn points out, wins every time.

Nonetheless, high-density audio will likely survive in some form or another, if for no other reason than that it offers a clear alternative to the compressed formats of computer file music. Understanding & Solutions' report on high-density audio's marketplace underscores the format's specific attractions to a niche market when it states, '...the key drivers will be the formats' more obvious features, such as surround sound and the inclusion of DVD quality video, rather than 24-96/192/DSD, etc'. In short, the palpable quality difference that high-density formats offer over both solid-state and conventional optical formats will be appreciated by only a small slice of the market; rather, it's the bells and whistles—multichannel audio and the ability to include multimedia elements—that will drive most of any success DVD-Audio or SACD have.

Those on the pro-audio side of high-density audio equation continue to feel optimism, optimism that they contend is not misplaced but which has been considerably frustrated. Jake Nicely, co-owner of Seventeen Grand Recording in Nashville, and one of a handful of surround music mixers, along with Chuck Ainlay, Ed Cherney and Elliott Scheiner, who have become the so-called poster boys of surround and DVD-Audio, voices that frustration when he says, 'I don't think the music industry has done a good job of promoting or marketing DVD-Audio. No one wants to be first, and no one wants to put any marketing money into it. But in doing so, they're letting the opportunity to promote what is a very good format pass them by.'

Nicely agrees that the most spirited enthusiasm for DVD-Audio has been centred on certain pro-audio circles and within special divisions at record labels, usually nestled in their new 'new media' departments. (Which, ironically, also have to address MP3 issues.) But he bristles at the suggestion that as a result those camps have become insulated incubators of cheering sections for high-density formats, unaffected by the kinds of realities outlined in research reports such Understanding & Solutions'. 'You want reality?' he asks testily. 'The reality is there. Panasonic just released their DVD-Audio players last week [in July in the US—the Panasonic deck will retail for $999.95 and the Technics brand model will cost $1,999.95. Matsushita, which owns both brands, also announced an in-car DVD-A player for later release]. But marketing DVD-

Seventeen Grand's Jake Nicely: ' I don't think the music industry has done a good job promoting or marketing DVD-Audio'

Audio needs to be a concentrated and co-ordinated effort between music companies and hardware makers.' An effort yet to become manifest: Panasonic has done little to herald the introduction of the players, which are backwards compatible with CD players, and the total number of titles available for DVD-Audio players is less then 500 thus far. (That number is an estimate which includes several dozen titles created three years ago by DTS as a marketing tactic in that company's unsuccessful bid to make its multichannel format the preferred one for the DVD-Audio specification. DTS is included as a secondary format in the DVD-A spec, so those titles would conceivably be playable on this generation of DVD-Audio players.)

Nicely also acknowledges the role of newer media in delaying DVD-Audio's acceptance. 'All the enthusiasm for DVD-Audio has been at the special divisions level at labels,' he says. 'Very little of it has filtered down to the group level, where everyone is totally concerned and worried about MP3 at the moment. It's been a terrible distraction.'

But he is unwavering in his conviction that, despite projections that it will never exceed single-digit market penetration, DVD-Audio will ultimately replace CDs. He is equally vociferous in his desire that SACD just go away: 'We'll have to replace every bit of gear we have to work on SACD because it's not PCM audio,' he says. 'All it's doing is confusing consumers and record labels.'

Chuck Ainlay does concede the possibility of ivory tower thinking in how DVD-Audio's proponents have proceeded. 'It's natural that studios and engineers want this to happen, and that they'll talk positively about it,' he says. 'But the reality is, we just don't know until consumers get it in their hands.' And, Ainlay adds, the bottleneck there is twofold: the lack of fast, simple, reliable and affordable authoring systems, and the reluctance on the part of content holders to spend what he believes it costs to do DVD-Audio mixes correctly.

'There's only going to be a few titles out at first, so they had better be the best they can to show consumers all that the format can offer,' he says. 'Word on the street is that labels are looking to spend around $20,000 for remixes or surround. That's not enough when you figure the studio and the engineer will cost you each around $2,000 per day, for eight to ten days for a whole album, plus the systems and media costs. It's not enough. But we don't need someone doing these remixes on the cheap with a few extra speakers set up in their living room.'

David Kawakami, the director of Sony Music Entertainment's corporate strategy group in New York, which is spearheading the SACD initiative, agrees that there's more to creating DVD-Audio than many labels have realised. 'There are a lot of surround mixes in the can, but there's more to it than that to make a DVD-Audio disc,' he says. Sony Music has put out about 60 of the estimated 100 titles available for SACD thus far; Kawakami adds that Sony as a record label is 'format-agnostic' and will release DVD-Audio titles, 'when there's a market for them.' The only other major label that has committed significant numbers of titles to DVD-Audio is Warner, which was a hands-on developer of the DVD format and pioneered its premastering and replication processes. But few of anyone's titles have made it to retail shelves yet. This Christmas is expected to tell the tale for that.

What support there is for DVD-Audio from major record labels appears muted. Warner Music Group, the leading proponent of the format, will say only that they are 'working hard to get titles out this year,' according to Jordan Rost, the group's vice president of new media. Otherwise, most titles are coming from small or startup companies, such as Los Angeles-based 5.1 Entertainment Group, which has announced 17 titles (16 classical and one jazz) on its Silverline label.

Mastering engineer Denny Purcell, who has done 15 DVD-Audio projects thus far, put it most succinctly: 'This is a science experiment. We're taking the best audio ever—the closest a listener has ever gotten to the studio—and asking if anyone cares. And this is probably the last physical media introduction the music business will ever have. So let's see what happens.' □

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RORY KAPLAN

Already a veteran of DTS surround music mixing, Rory Kaplan has worked with the greats on more multichannel remixes than anyone else. **Richard Buskin** rounds up his experience

RORY KAPLAN'S INITIAL EXPERIENCES in the music business came as a keyboard player, programmer, songwriter and engineer. He toured with luminaries such as Chick Corea, Herbie Hancock, Stevie Wonder, Christopher Cross and Michael Jackson, and played in the studio for Yes, Joe Cocker and Belinda Carlisle.

After producing an album for Edgar Winter, Kaplan was introduced to DTS by Winter's A&R man and learned that the company intended to make its entry into the consumer market for multichannel music in order to 'pioneer and spearhead where quad hadn't been able to go before because of the medium delivery'. By the time that Kaplan joined DTS in 1996, the relatively small setup had already enjoyed considerable success on the theatrical side with *Jurassic Park* and other motion pictures, but it would be largely thanks to Kaplan's own record industry contacts that it would be able to diversify, commencing with Elliot Scheiner's multichannel remix of The Eagles' *Hell Freezes Over* album.

'That was a real success story,' Kaplan recalls, 'because it went from a DTS CD—which did very well and gained us a lot of publicity—to being a laser disc that generated even more sales, and then to a DVD. In just a little over a year that thing has sold more than 400,000 units, and it's only got a DTS track on it. Elliot's perspective was that he wanted the audience to almost be in the orchestra pit, with The Eagles playing in front while the additional musicians were on the sides and at the back. We did that without picture, but when you locked it to picture it somehow made sense. It was really phenomenal.'

DTS excursions on the part of Al Schmitt, Ed Cherney, David Tickle, Chuck Ainley, Rob Jacobs and George Massenburg have since resulted in surround remixes of recordings by Bonnie Raitt, Sting, Roy Orbison, Steely Dan, Diana Krall, Dave Grusin, Vince Gill, Trisha Yearwood, Olivia Newton-John, Don Henley, The Mavericks, Peter Frampton and Lyle Lovett, among many, many others.

'I've worked on about 80 of these projects, and you really pick up on the details of how to use reverbs correctly, what delays to use and what new equipment is supporting this format,' says Kaplan. 'George Massenburg was waiting for the proper verbs to come out before he would try it, and I know that Lexicon and TC Electronics have both come out with some great machines for working in this format, where the reverbs are coherent, time-aligned and not out of phase with each channel—in other words, all of the headaches that we had to go through in the beginning.

'So, right now I'm getting excited watching all of these engineers discover this new format, because it's like a whole new canvas for them to paint on. It's a chance for them to relive music in a format which they would have liked to have heard it in from the beginning, and the result is that they're producing some stunning results.'

What ground work needs to be done in terms of preparation for a multichannel remix?

Quite a few things. One is figuring out what studio you're going to work in, and that often relates to the tapes. You see, if you're dealing with a project that's 15, 20 years or older, the condition of the tapes is a big issue. These are your master multitracks and you don't want to lose anything, so you have to take the utmost precautions. Capitol Recording Studios [in Hollywood] has a great convection system, and so we sent the analogue tapes of Don Henley's *End Of The Innocence* album there to be transferred to 3348, which means that the label now has a backup.

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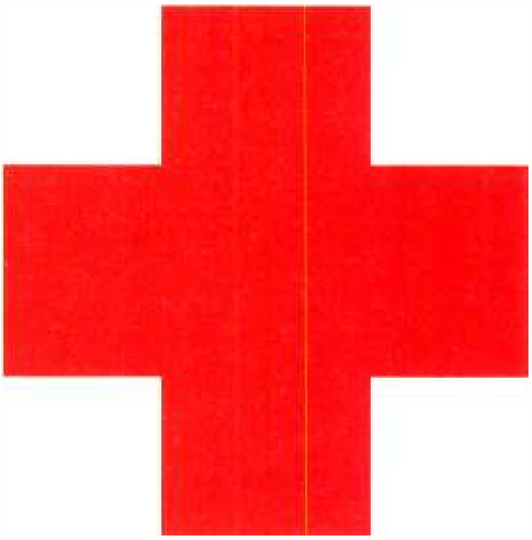

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Anyway, once the tapes have been secured in good condition you then have to make sure that the room in which you'll be working is set up for 5.1. Otherwise, if it's not, you have to make sure that you have enough buses and whatever box they want to use to monitor 5.1, along with enough automation faders and the whole nine yards. After that, it comes down to what medium you store it on—originally it was D-88s with 20-bit convertors, whereas during the past two and a half years everything with DTS has been done 24-bit —and then there's the outboard gear.

I mean, the good news is that we're not using half as much compression now—we're only using it on a kick drum or a bass or a vocal if necessary, and we're doing so as an effect of that instrument, not for popping things into perspective through stereo imaging. So, it depends on the project in terms of how big it is and how many tracks are available, but it's good to have at least a 48-input board, because you're going to want effects returns and have extra faders to do sweeping if your board is not set up for it. On the other hand, there are going to be a lot of small places popping up that advertise surround, but you've got to make sure that they really have the right outboard gear and are prep'd correctly.

How do engineers generally arrive at their choices of monitoring system and configuration? what are the common mistakes and misconceptions?

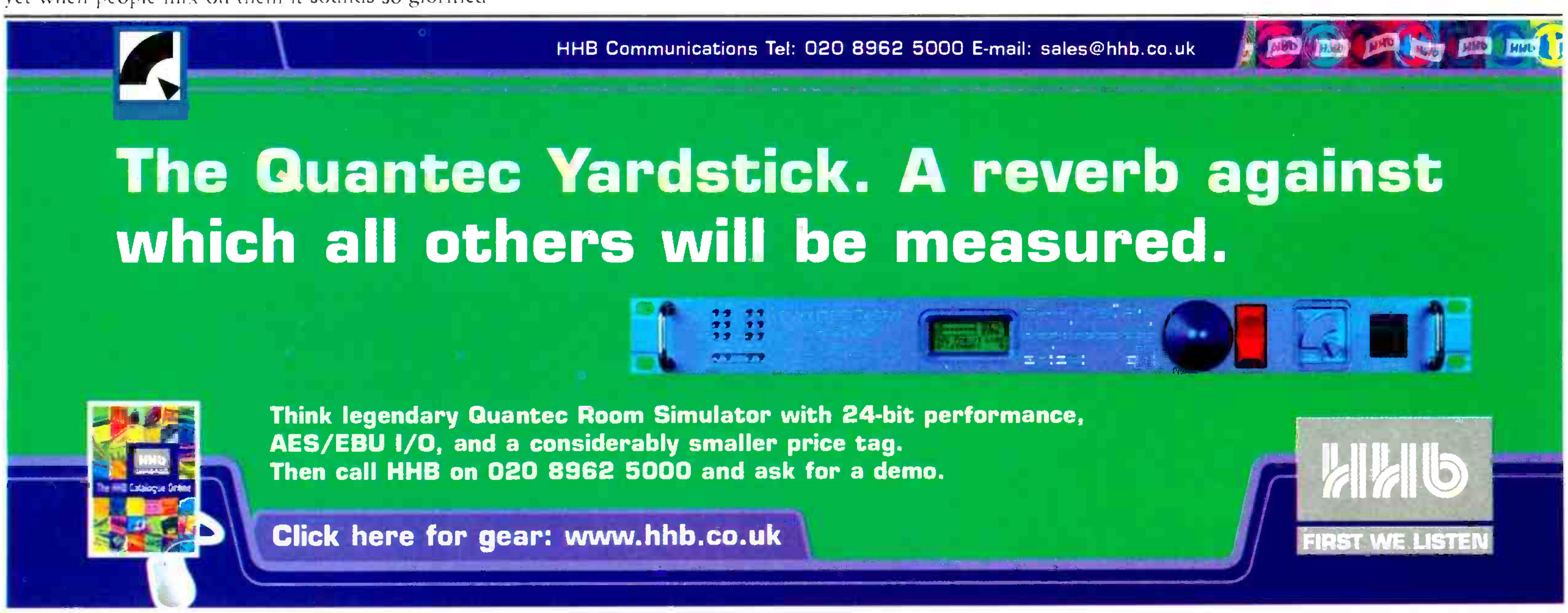
Well, it's really interesting. For instance, Don Smith engineers for Tom Petty and The Rolling Stones, and I went with him and Mike Campbell to do a 5.1 mix of Mike's own material at Cherokee Studios. They have these sort of mid-range monitors—not soffited big speakers but free-standing Klipsch floor speakers —and the room is kind of narrow, and I couldn't tell what the heck was going on in there. I encoded the stuff, we played it back, and the level was down on one side and a number of things were wrong. The encoding was an input-throughput process and we hadn't level-changed anything, so for Mike that room clearly represented a false perception in terms of the speakers. They therefore went to another room in North Hollywood which had near-field monitors and all of a sudden it made sense to him.

So, every engineer has his own particular monitor. For instance, I've noticed that the Genelec 1031s or 1032s are beautiful-sounding speakers for playback, yet when people mix on them it sounds so glorified that they miss some of the details because they didn't work that area. Some guys are still using NS10s because they like the subwoofer crossing over 85 or 100 cycles down, and then there are other guys who are using these KRK Exposé 8s, Al Schmitt is using his Master Lab Tannoys. So, what I've found is that the speakers that these engineers are used to using in stereo suit them best in 5.1 with a decent subwoofer. That's really the trick. I still think there's no substitute for near-field monitoring in this format. You can't use the big soffited speakers. If you want to A-B to them to get a feel for them, that's one thing, but to me it's really a danger to try to mix everything on big speakers. You lose all of the perception of depth.

Is it best to have a preconceived idea of sound placement and soundfield, or does experimentation yield the best results?

The straightest answer is that every song dictates what's going to happen. Like when I listened to Sheryl Crow's album [*The Globe Sessions*], at the beginning of the song 'Something More Than Nothing' there are these great ambient tracks going on, and then a little drum rhythm comes in, the piano sort of trickles a bit and her voice is real light. Well, in my head I can hear all of the ambient stuff floating around the back sidefields in the mix. Most of these songs I can hear where they're going.

Producer-engineer Chuck Ainley (left) relates his adventures in surround mixing, as Rory Kaplan listens intently at the Pro Sound News US conference



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PSN US E-Studio Summit: Jake Nicely (standing) introduces his e-studio panel (left to right) Kerry Moyer,Chuck Ainley, Rory Kaplan, Hank Williams and Denney Purcell

When we did The Police's *Greatest Hits* album with David Tickle, he did a great job figuring things out because some of the tracks from the early days were really sparse. You know, six channels of drums, bass, guitar and vocal. It wasn't a complicated production. So, he had to figure out how to take the voices and delays and verbs and rhythm guitars, and where to space them so that there was some ambient cohesiveness to it. It wasn't that easy, but then you get an album like Sheryl's where, for instance, I can already hear those great Rolling Stones-type horns on 'There Goes The Neighbourhood' probably in the back sidefields.

So, each song dictates what you do, but the flip side is that you still want to maintain the integrity of the intention of the song. Like you don't want to make some artistic move and say, 'Oh, it'll be hip to have her vocal come out of the rear-left speaker, detached from everything else', unless it's theatrical like a Pink Floyd project.

What are the common mistakes that engineers make with respect to their first remixes?

Yes, well, the common mistake is either overloading the centre channel with too much or too little information, or overloading the subwoofer with everything just because it's there; in other words, kick drum, low end of the piano and occasionally bass. I mean, people now have home systems with bass management that is going to drive all five speakers of everything below 100Hz down to the sub, so you

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want to make sure you're not going too far and for that there are bass management monitoring systems.

Trying to use stereo verbs at the front and back is also really kind of a no-no. It's much better to use mono verbs for delays for the rears. You should also try to isolate as much as you can and create your own sound stage, while using the rear speakers as full-range speakers, not as the old Dolby Pro Logic approach where you're out of phase and get a little effect back there. We're using full on, and somehow DTS has created that standard for full perceptual audio all of the way around.

Is there a best plan of approach when it comes to building a multichannel mix from scratch that will reveal placement opportunities and ideas?

I'll tell you what most guys are saying... and it took me a while to agree with them, but I have to now. What they're doing is they are taking the original stereo mix and they're recalling or mixing their stereos first. They're going to recall everything as close as they can to the stereo so that they'll get a perceptual idea of the track—how the vocals sit with the track—and then they'll go ahead and pan out from there.

So, the building block is to get your stereo imaging first, or get as close as you can to the original, and that accommodates quite a few things. It gets you back to the integrity of the song, it gives you a feel for what was intended for that song, and then, when you start placing things around, you have something to refer to so that you don't get too far off base.

That has been the easiest way to build. Of course, get your rhythm tracks up first—your percussion, your drums—because that gives you a great soundfield to go from. Your bass is usually diversed a little off to centre-left and right, with a little bit of a point of it in the centre speaker and in the sub. As for lead vocals, on the remix of Gaucho Donald Fagen for instance loved having his voice in the centre speaker, and Elliot [Scheiner] got some criticism for that—from outside people, not those in the industry—and he thought, 'O-oh, I can't do that anymore.' However, that's one of the greatest mixes around, so we had a long talk about it and I said, 'You can't worry about public opinion. You are creating the standard.' It's like saying, 'That's a horrible song because I don't like it.' It's a personal opinion.

Why I think the issue of the centre speaker was such a problem in the beginning is that four years ago most home theatres were of the old thrown-together Radio Shack variety, where the centre speaker was less than an Auratone quality. So, if you loaded up too much information it would crap out and you wouldn't hear what the engineer had done. The other trick was to have the engineer not mix to the lowest common denominator, but mix to the highest and make the guy buy a new speaker. As a result, the speaker companies have now jumped on the bandwagon and they're selling complete 5.1 systems for under a thousand bucks.

In terms of where you place the speakers, does this vary between film and music work?

Well, the ITU standard specifies a 60° width at the front and 110° for the rears, but while that's fine for the film industry it doesn't work quite the same for the music. I went over this, in fact, with Elliot Scheiner, David Tickle and Al Schmitt—they tried going out wide on the rear, but you're not just putting effects back there; you're actually using them as full-range speakers. So, we drew more of a rectangle as opposed to a wider circle, and as a result you get better depth of perception from verbs and delays, especially from front to back, but also from back to front. When you go too far wide I lose that perception, and it turned out that a lot of other engineers felt the same way. So, it's kind of interesting, because I'd hoped that music would catch onto the sort of merge used between film and home theatre, but when you're mixing these things it really doesn't work.

I did Don Henley's *End Of The Innocence* with his engineer Rob Jacobs at the Record Plant using an SL9000, and the software of the board wasn't set up for a 5.1 mix, so we had to figure out how we were going to use the pan and small faders to derive what we needed out of it. Then, once we did, we tried the wide separation at the back to see if ITU would even work in that room and then we put the speakers back

TIAL DO

multichannel remixing

1) 'Absolutely find a speaker that you are comfortable with.'

2) 'Use the centre speaker, but always be careful to diverge what is in there.'

3) 'Try to use the original audio chain that was employed on the stereo mix.'

4) 'Do try to use bass management for monitoring back and forth with and without.'

5) 'Don't be afraid to use your subwoofer when applicable.'

6) 'Keep the music intact and don't worry about the gimmicks.'

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where I normally set them up. I set them up at an equal distance from the engineering position —technically what I do is put a mic stand where the engineer would sit, take a tape measure, go out to the centre speaker and take an equal distance to the left and right speakers. Then what I'll do is move the mike stand about a foot closer to the board and do the measurement again, so there's like a convergence zone. As a result, if you sit forward eight or ten inches, or sit back eight or ten inches, at least you have a converging area as opposed to having to keep your head in the exact same place. Likewise, I set up the rear speakers in precisely the same way, and it really works out well. I mean, so far I've done that for 99% of the projects that I've been on and it's been failsafe.

We even did a 6.1 mix on one of Henley's tracks. Dolby came out with a matrix rear-centre—with DTS we go up to 16 discreet channels, and so they decided to do a discreet centre-rear and see how it works. On the Henley track it worked fine, and so now I might even mix Sheryl Crow's last album [*The Globe Sessions*] in 6.1. I'll do a 5.1 as well for the DVD-Audio, but I think we'll go ahead and have it available for the 6.1. That'll be kind of interesting.

What role do—or should—the surround speakers have on the total mix? Is it realistic to expect a whole lot from them in the average domestic listening environment?

Well, you know, there's this great thing called the WAF—the Wife Acceptance Factor...

That applies to a lot of things.

Yeah, and apparently some people have their rear speakers up behind a bookshelf, ten feet back to the side. Their ears are not even at the speaker level. So, there are all kinds of issues, but you have to take into account that most people will hopefully have their speakers all in phase in the right place, and you've got to mix with that in mind.

DTS has a test disc that it sends out, and it checks out your speakers, your level settings and everything, so when you listen to this stuff it's set up properly. Nevertheless, that's an education. That's a whole other article, just dealing with the education of the public.

We were talking before about how there's no preset formula, because every record has different requirements. however, what considerations must be given to the multichannel remixing of an established and much-loved classic album?

For example, Steely Dan's *Gaucho* album is accepted as one of the standard CDs, and so when we went to remix it we had to bear in mind that the original board it was mixed on was a Neve 8078. We therefore remixed it on an 8078 at Donald [Fagen]'s studio, River Sound, in New York, and Donald and Walter [Becker] were in the room, making the calls with Elliot. I mean, aside from the equipment, the other thing which DTS has established—and this is largely because of my own music background—is that, wherever possible, you should try really hard to keep the original people involved and thus keep that integrity.

In line with the need for consistency, how important is this with regard to the formats being used between the stereo and the surround mixes?

That's a good point. For instance, with Don Henley, originally they used a Rev1 on like three of the tracks for his lead vocal, and I happen to have one. So, I brought my rack with me to the studio and we did use some of the original verbs of that time. Basically, you try to match the original audio chain—I mean, if the original audio chain off the tape was that it went to an LA1 compressor, a Neve 1073 and then to a Lexicon 480 XL, in most cases the guys will recall that chain as far as possible. There again, it's not always possible—or cost-effective—to rent certain vintage gear, and in some cases the equipment now is better, so it depends. After all, you've got to watch your expense, and therefore if I have a choice between a Fairchild and an Avalon, and the Avalon sounds just as good and the engineer says, 'Hey, look, that's all I need', then I'm going to go with the Avalon. That's where the production value of experience comes in, because you don't want to compromise the music but you've also got to know what gear does what best. It's a tough call.

CARDINAL DON'Ts of multichannel remixing

1) 'Overloading the centre speaker with too much information.'

2) 'Overloading and not checking your bass information to the subwoofer.'

3) 'Not aligning your speakers properly.'

4) 'Not taking budgets into account for original equipment... That creeps up on you.'

5) 'Not taking into consideration the original intention of the song.'

6) 'Going in with a preconceived idea.'

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2000 OLYMPIC GAMES

This time around it is the city of Sydney that has the honour and the privilege of staging the world's most prestigious sporting event. **Richard Clews** sets the scene on its international broadcast

THE STAGING of the Sydney 2000 Olympic Games is possibly the most eagerly anticipated broadcast event of the year. In Australia, the Games have been a front-page story for months on end, with various controversies and rumours serving to provide even more pre-event speculation than usual.

What is known for certain is that for the 16 days of the Games themselves, the International Broadcast Centre will be the world's largest TV production house. The A$80m warehouse conversion, located at the Sydney Olympic Park in Homebush, holds 70,000 square metres of broadcasting facilities. This includes 50 television studios which are being fitted out by some of the 190 organisations who have purchased Olympic television rights. More than 3,400 hours of competition from 300 Olympic events will be televised to an estimated peak audience of 4bn people.

Orchestrating this massive enterprise is the Sydney Olympic Broadcast Organisation (SOBO), a full-service broadcast company created by the Sydney Organising Committee for the Olympic Games (SOCOG). In the lead up to the Games, one of SOBO's concerns was to ensure the highest quality sound for the coverage of events, improving on the high standards set out during the last Games held in Atlanta in 1996.

Equipment playing a major role in the broadcast effort includes two 60-channel Calrec Q2 desks and a 60-channel Calrec S2 for NBC Olympics' main Broadcast Centres, where they will be used in mixing programme feeds for NBC's US broadcast and cable transmissions. The BBC's Sound Control Room will also house a 36-channel C2 and an M3 mixing console, while a 60-channel Q2 will be used by Australian OB company Global TV.

From across the water in New Zealand, Moving Pictures, a division of Television New Zealand, has commissioned a new OB mobile vehicle especially for the event following its securing of a contract with the host broadcaster, Seven Network, to provide a variety of production facilities during the course of the Games.

The truck is built around Moving Pictures' second Euphonix CS-series console, a CS3100B. 'Moving Pictures is committed to providing its customers with quality outside-broadcast equipment to support a team of professionals that bring a high level of experience and expertise to every assignment,' said General Manager Maureen Ross. She further maintains that the resettability as well as the sound quality of the CS3100B sits well with their brief.

Meanwhile, Audio-Technica, who over the last two years has developed a series of microphones to improve the sound of sports coverage significantly. The company was selected by SOBO and NBC Olympics, responsible for the US broadcasting of the Olympics for the next 10 years, to provide more than one thousand mics in total.

Bob Dixon, project manager-sound design for NBC Olympics said: 'Picture this: we're at an Equestrian event with horses coming over a hill, and they have to jump over these big logs and then into a pond of water. Since our cameras can't be too close, we are going to have an AT815ST stereo shotgun mounted directly on the camera for stereo ambience. We will then use A-T's U100 Series Wireless and place the mic in proximity to where the horse jumps, allowing us to capture the up-close sound corresponding to the picture, with the proper ambience. It will be climactic.'

Making sure the climaxes are reached according to plan has required A-T's attention over recent months. Dennis Baxter, Sound Designer at the Atlanta Games and consultant to Audio-Technica during the Sydney Olympics, gave an idea of the hurdles to be faced.

'One of the biggest problems in Sydney is the weather. The Games take place at what is officially the start of the spring in Australia, when the weather varies. Audio-Technica has had to pay close attention to protecting the microphones from wind and rain, using windshields. You have to expect the worst, but hope that in the end the weather will smile on you.'

Whatever the weather, Audio-Technica plans to have a huge range of mics available for broadcasters. Among the mics that will feature are the new AT815ST and AT835ST stereo shotgun mics, AT804 omnidirectional hand-held mic, AT825 and AT849 stereo mics and AT895 adaptive-array mics.

Another potential source of problems is cabling.

Below: SOBO audio director Al Craig will be producing the stadium and arena audio for the Games from his vantage point behind two of the 10 Panasonic SX-1s.

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The control rooms are one kilometre away from the venues themselves. Two Otari Side Winder systems have been installed to cope with the demand, working alongside Telstra's Millennium Network (TMN).

More than nine years in development, the TMN links the IBC with the Games venues via 4,800 km of optical fibre. Some 3,200 audio and 280 video links are on hand, in a network using Synchronous Digital Hierarchy geographically diverse self-healing ring topology. Of utmost importance to broadcasters, this means there are always several ways to get signals from point to point. The TMN has been proven a success at 36 Olympic test events, and looks set to handle the massive coverage over the 16 days of the Games.

Elsewhere, ARX has responded to requests from Panasonic Broadcast and SOBO for an Audio for Video Switcher based on its Sixgate unit. The new VCS-6 DC controlled Audio Switcher-Gate is the result of several months' development and testing to meet Panasonic's needs. In addition, ARX is providing major audio contractor Greater Union Entertainment Technologies (GUET) with 106 MaxiSplit Line Splitters. At each of the 34 Games venues (15 of which are located in the Sydney Olympic Park) there will be a PA and broadcast audio splitter system based on three MaxiSplits. GUET will also press 33 Quadcomp 4-channel compressor-limiters into service for level control.

Once the Games get under way, American TV viewers with Dolby Pro Logic decoders will be able to enjoy the Olympic broadcast in Dolby Surround, thanks to NBC. Multiple mics will be placed in the audience at many of the major events, including the opening and closing ceremonies, swimming and gymnastics competitions and track and field sports. The NBC Broadcast Centre is being fitted out with Dolby Surround sound equipment centred around 13 SEU-4 Dolby Surround encoders and 10 SDU-4 decoders, and 5 Dolby 430 noise-suppression systems.

Away from the stadium complex, Sydney's Studios 301 has recorded the music for the opening and closing ceremonies, effectively 'bookending' the broadcast.

The recently refurbished and reopened studio has been the base for the recording services provided to the Olympics Committee for the opening and closing ceremonies of this first Olympiad of the new millennium. Studios 301 will provide all location recording services for these events, which will then be mixed at the studio's impressive new facilities in the Sydney suburb of Alexandria. Studios 301 will employ hundreds of graduates and students from the School of Audio Engineering as Assistants and General Assistants to support the recording projects.

'Being the facility to document the ceremonies to these Games is an honour and one that Studios 301 is worthy of, both in terms of its own history and legacy, and in terms of what we can bring to the event,' said Tom Misner, owner of both Studios 301 and the SAE. 'Studios 301 is the flagship not only for SAE, but also for the Australian recording industry. We believe that the world is looking upon these Games as an important moment of peace and worldwide co-operation at the start of the next millennium. We will play our part in ensuring that it is seen and heard throughout the world.'

Add in the Australian accent, and the picture of a country eager to supplant its reputation for soap operas and novelty singles with a broadcast equal to the status of the Olympic Games is complete. In all, the Sydney Games should prove to be a comprehensive test for everyone involved—both on and off the track. And given the long-term nature of Olympic planning, audio requirements at the Athens 2004 Games are sure to be on everyone's agenda in the very near future. □

AS THE BIGGEST TECHNOLOGICAL achievement in Olympic history, the 2000 Games boast a lot of firsts. Among them is the complete broadcasting of all events in stereo as well as significant live sound reinforcement. Obviously, this has ramifications for all equipment down the line. There is only one location, the archery venue, where the audio is being sent as an embedded signal back to the IBC. This is because of power generators being used nearby. Other than that, every event will be sending multiple stereo feeds separately via fibre. Those feeds will come back from Stadium Australia, site of opening ceremonies, track events and the soccer final, and the swimming and basketball venues through 10, 52-channel, Panasonic SX-1 consoles. These boards output stereo to DVCPro50 video decks being used as audio loggers in each of the control rooms and are then fed back to the main control room of the IBC and onto the individual broadcasters (the US rights holder is NBC) to do with as they please. Overall there will be more than 50 audio mixers including those at remote venues.

In addition to the stereo feeds for broadcast, significant effort has been made to upgrade the on-site audio for those in attendance. Included in these setups are the large in-house systems with an overlay for the Olympics comprised of 34 Ramsa sound systems. For example, Stadium Australia has a large Bose system installed permanently, but will receive the Olympic overlay to augment sound.

The raising of the audio profile for these games has had a profound effect throughout the city. Not only is the top local recording facility, Studios 301, busy tracking opening ceremony music, but the famed Sydney Opera House is the site for another Olympic recording.

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HALLYDAY IN PARIS

As Johnny Hallyday pulls an unprecedented crowd to watch him perform in front of the Eiffel Tower, **George Shilling** steps backstage to report on its live broadcast and surround recording

FRANCE HAS FEW STARS bigger than Johnny Hallyday. His recent Paris concert took in the lighting system on the Eiffel Tower and attracted over 500,000 avid fans. 'Estimates range from 500,000 to 800,000,' offers Mega Studios' Thierry Rogen, who with Le Voyager's Yves Jaget was responsible for engineering a live broadcast and recording of the mammoth event. 'I think it was about 600,000 people, but we will never know exactly.'

'It was a very big show', agrees Jaget, with a twinkle in his eye. He rolls a VT to show me a truly unbelievable crowd. 'From front to back measured about 700m, about 60m wide,' he observes.

Make no mistake: Johnny Hallyday is big and his shows are big events and big news to his fans.

'In France Johnny is a little bit like The Boss, Bruce Springsteen in the US, especially on stage' Jaget confirms. 'Each year he does a new, crazy show. He was the first guy to do the Stade de France. He has done so many crazy shows in France that everyone is waiting for the new concept. To put on this show, we're talking about FFr40m (£4m). The pyrotechnics were amazing, and the guy who did the lights had control of all the lights on the Eiffel Tower from his board.'

We are sitting in Mega Studios in Suresnes, Paris. Outside, the road is significantly narrowed by the presence of Le Voyageur's V1 mobile. Five days previously, the pair had handled the recording and broadcast sound for a huge concert by the French star, coproducing with musical director Ivan Cassar. Things were complicated by a seemingly impossible mix deadline. After the Saturday show, mixing was scheduled for Sunday to Thursday, with mastering on Friday. Two factories had been reserved for pressing over the following weekend, with the CD in the shops on the following Monday.

Using Le Voyageur's SSL Axiom MT-equipped V1 truck, Rogen recorded the concert, while Jaget handled the mix for broadcast on TV and radio from the Neve VR-equipped Voyageur V2.

'We rehearsed for three days before recording this show,' says Rogen, 'recording three warm-up shows in Toulouse at the Zenith, which holds 10,000 people, to be safe technically and artistically.'

The recordings of the warm-up concerts were taken onto the V2 mobile to set the initial balance. 'The TV Channel TF1 broadcast the concert live, and the sound was simulcast on RTL in stereo,' adds Jaget, 'so we had a lot of work to control the delay, because between the TV and radio you have a different delay: you work with satellite for the TV, but the radio is a fibre channel, so I employed a guy who specialises in this kind of work, so just before the antenna he checked everything. In the V2 truck we had a little laboratory. He got the line two days before and repeatedly checked to make sure the TV and radio were in sync. With the radio there is a 600ms delay, but with the TV it is about one second more. In France the radio technicians and the TV technicians work in different worlds and don't care if the delay is different, so we do the work for them. We also multitracked to a Sony 3348 in the second truck in case of failure'.

Fortunately this proved unnecessary, but 'if Yves had had a problem, he could have used my balance for the broadcast,' Rogen explains. 'At first, the company asked me to do everything with the V1—the recording and the broadcast—and I said "no way", because if we had a problem I would have taken a plane to Africa and you would never see me again.'

The band for this concert consisted of Hallyday and guest lead vocalists, five backing vocalists, two guitarists, two keyboard players, bass guitar, drums, eight string players and a 4-piece brass section. However, Jaget explains: 'The biggest work for this show was to design and install the audience mics, because we prepared the stuff to have a real 5.1 recording, and also you can't rehearse the audience. We focused on an area of 300m², and we split the design into two parts, because I needed to have lots of mics for the broadcast stuff, but I couldn't use the very distant mics because there was such a big delay, so we designed a special part on the front of the stage to have the closer audience, and used the further mics just for the recording, and move it in the ProTools. At the far end of our square the delay was 600ms. We used six shotguns on the front of the stage, split into three positions for LCR, and also two stereo pairs. The 42 mics were recorded on 16 tracks.'

As regards the onstage miking, Rogen continues, 'The PA engineer Jean-Pierre Ganneaud is a good friend of ours, so we agreed with him what microphones to use. He used a lot of radio mics—for the horns, backing vocals, lead vocals, bass—only the drummer was without a radio system. There were about 44 radio frequencies, plus the in-ear monitors for each musician, so it was crazy, and there was some interference to start with.'

Jaget: 'We were parked 200m from the back of the stage, just under the feet of the Eiffel Tower inside a large security square that was mainly for the very, very big pyrotechnics. We had a lot of pressure because everything was buzzing and noisy one hour before the show. We installed the equipment two days before the main show, but the problem was that we were not on the same generator as the PA system. We had changed everything and were isolating with transformers, but just one hour before the show we changed to the same generator as the PA, and—no noise.

'We recorded audio to two 48-track Sony HR machines in the V1, but in 16-bit because our v1 Axiom only allows you to hear 20 bits. The new version is 24-bit. In the truck we use two Pro-Bel interfaces for AES to MADI. We have 96 mic pres, which were all made use of for this show. We recorded the audience to four Tascam DA-88s. The single 48-track machine in the other truck is not a fully digital recording so the sound is not the same. But this was just for backup, and wasn't needed, as we had no problems.

'The mic signals were split from the PA to the SSL remote mic inputs (which connect to the SSL Rio fibre-optic system) on stage with a splitter box using Jensen transformers. There were two fibre-optic cables for the 96 mic lines—you can do it with just one, (it is an 8-way multi-fibre), but always for safety reasons we use two. In the rehearsal somebody unplugged a mic pre, and in this version of the software that means you need to reboot—you can't reroute the mic pre. It works fine on v2.' Jaget prefers the v2 software, but reverted to v1.3 for compatibility with Mega for the mixing. Rogen has yet to update the software in the otherwise identical Mega console, as he has several ongoing projects where recalls are needed.

The Axiom-MT v2 was designed to be compatible with v1, but Rogen, perhaps understandably, did not want to risk compromising any tracks recorded with v1 by upgrading to v2 in order to be compatible with Le Voyageur's software. The latest Axiom-MT software release, v2.2/39, addresses upwards compatibility issues, including some which were outstanding at the time the concert was planned.

Jaget had a cunning plan to achieve a seemingly impossible deadline for the mixing by parking the V1

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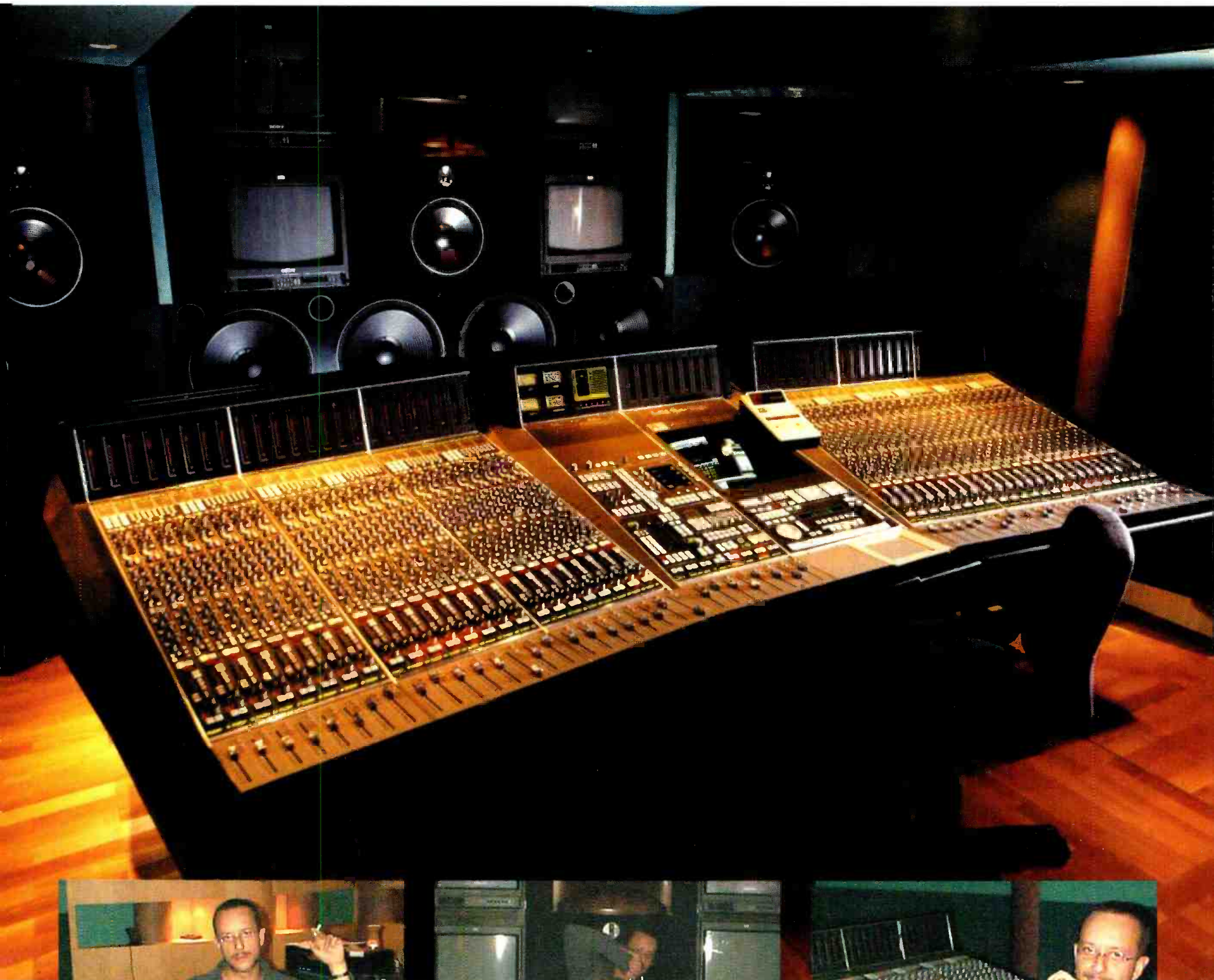

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Le Voyager's Yves Jaget

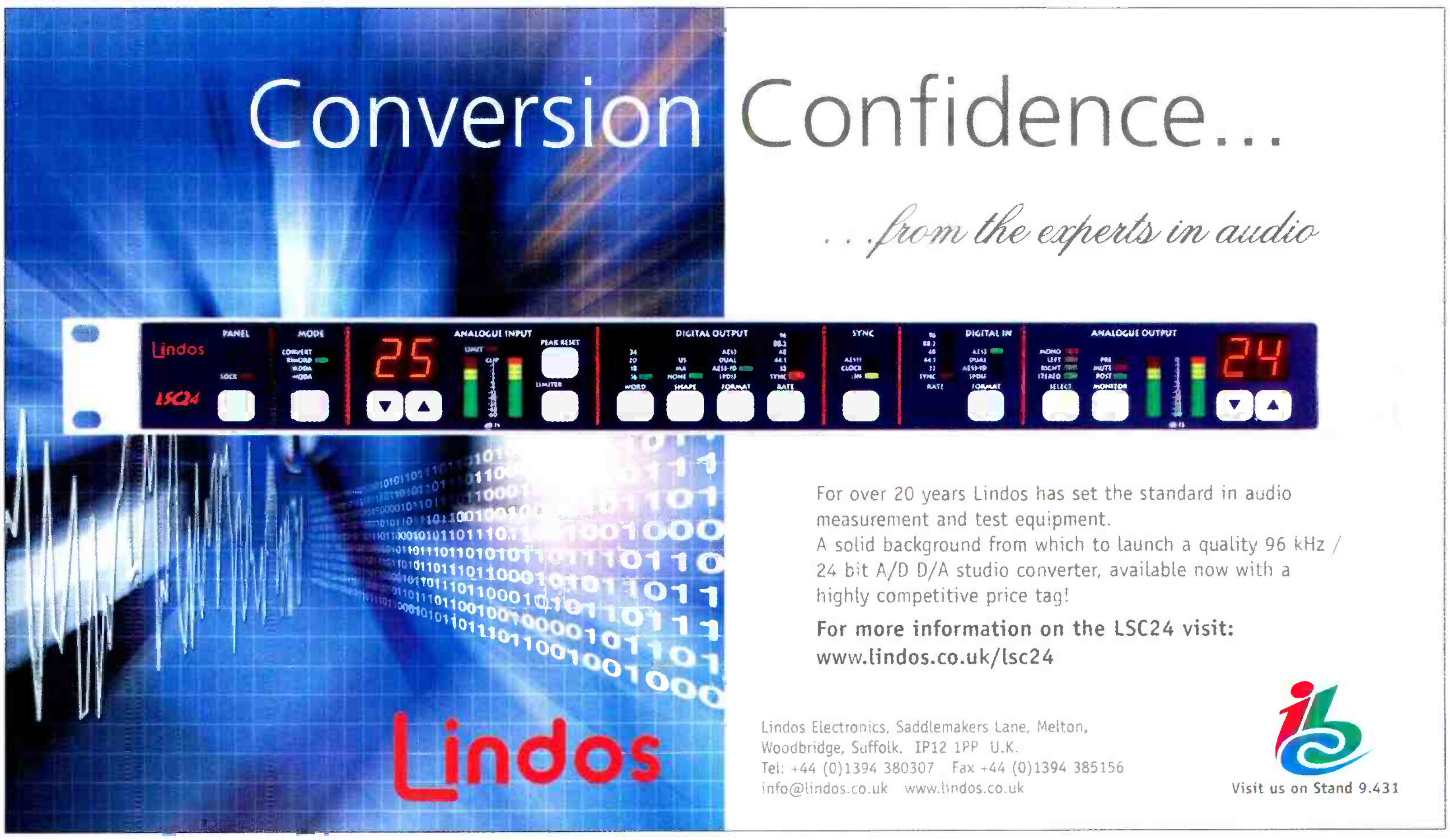
outside Mega for simultaneous mixing. 'We are friends with Thierry, and the project is so big, and we have to mix so quickly, that the idea was to begin the project on this Axiom, and copy all the setup to the other one and mix different sections simultaneously. So we split the concert into four sections, and are mixing the whole show during the five days.'

Rogen: 'An hour after the show we came back to Mega and used two Pro Tools—two control rooms all night to clean up the gaps in Pro Tools for two of the four sequences that split the show into. Even for just a 2-bar gap in the horn parts we would clean it, and we use a crossfade in and out every time. Two years ago, mixing live stuff I spent hours and hours muting channels because of the problem of spill. Now you never see a cut on my automation, everything is open all the time. It's fantastic because you win a lot of time, you don't have to spend time muting stuff because it is noisy—this is not music, it is a boring technical thing. So when you forget this you just have one thing to do, which is make music and get a good balance.

'On, the day after the show, Yves and I came into Studio B to make the balance together, and when we were happy, I made an M-O disc for Yves to put onto the desk in the Voyageur. We are friends—I think it would be impossible to do this project if you didn't have that kind of relationship, because the time factor is crazy—only five days to mix 25 songs.

'We have two lines between the truck and the studio—I send them my mix, and he sends me his mix so we can compare sometimes to make sure we are on the same wavelength'. These were straightforward AES-EBU connections with using a thick coaxial, 60m long.

Although the stereo mixes that were hurriedly being completed were primarily for the imminent CD release, they will also be used for the VHS soundtrack. So the pair were working to picture. I was privileged to see one particularly entertaining song, which featured the dancing girls from the Crazy Horse, who were notable for their rather revealing costumes. The unedited broadcast pictures had been transferred to a Doremi Labs V1 hard-disk system, which compresses the data almost imperceptibly, and chases time code from the SSL with wordclock connected to the Pro Tools. Jaget: 'It's a fantastic system, we used two 9Gb disks for the show which give about three hours of video. Using the SSL pen I can change songs instantly.' Rogen: 'Doremi uses great algorithms and looks great. We have a Doremi in house all the time, and we rented another one for the truck. So we did a copy onto both machines. And the same with the Pro Tools, there is one here, and for Johnny we bought another Pro Tools, and we installed one on the V1 to have the same data. So if we make any modifications here, we take the



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Mega Studios' Thierry Rogen

hard disk onto the V1 and just load it, and it's 100 percent compatible. We have three Pro Tools v5 systems—one for the editing and cleaning and two for the mixing, each using 64 tracks and about 38 plug-ins. We are also tuning vocals, and sometimes the musicians have made mistakes. We don't have time for overdubs, but it is almost always possible to find another place in the song where you have the same chords, so we take that to fix it. We used the Studer ADAT-to-MADI interface for the transfer, so we take the MADI from the 48 tracks to the interface, then optical to the ADAT bridge for Pro Tools, so we lose nothing. The Pro Tools in Studio B is connected to the console with eight 888s using AES going to the SSL Rio at up to 24 bit. We multitracked at 16 bits because when you record live music at high level on the tape there is not much difference. But using the Pro Tools on 24 bits for processing and plug-ins makes a bigger difference.'

Jaget: 'When we mix the DVD we will take the M-O discs from the stereo mixes into Studio B. There is a lot of work to do for 5.1 with movement. But we'll have the edited pictures then, and put those into the Doremi.'

Rogen: 'Now Yves has finished his first set of mixes, we transfer from the V1 to this console with the AES-EBU link, then from this console we go to the Avalon EQ and compressor, then directly to the half inch. [15ips with Dolby SR] It seems funny for people who use digital all the time, but we love to use digital for multitracks, but I don't know any machine that's as good as a half-inch machine, especially when you put an analogue limiter before it.'

The mastering was to take place the day after my visit, and while I was there they committed the mixes of the first songs completed in the truck to half-inch and DAT in Studio B, and immediately couriered these to the mastering house, so that Jaget and Rogen could approve it.

'Generally I master at Metropolis because I like to work with Tony Cousins, but because we have a very short time we will master at DM Music in Paris. But they start mastering tomorrow, and I want to listen tomorrow evening to be sure it's right. In 30 minutes we will have the first two sequences finished, so they will go to the mastering studio tonight, so that we can listen to the sound of the mastering before we finish mixing sequences three and four,' says Rogen. 'Later, in two or three weeks, we mix the DVD. 'This session is only for the stereo stuff—for the CD and for the VHS, and the next work is for the DVD in 5.1, but I think we'll have much more time for that.' This will take place in Mega Studio B with Jaget and Rogen, possibly even working together in the same room. □

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THE HOLLOW MAN

With its main character absent from the screen, Hollow Man makes dramatic use of SDDS to fill in the picture. **Richard Buskin** talks to the invisible sound men behind Sony Pictures' psychological drama

THE SETTING is a top-secret military laboratory. Its inhabitants are some precocious young scientists who have just discovered the formula of invisibility, and the protagonist is one Sebastian Caine, the team's arrogant know-it-all leader, who opts for adventure over common sense in order to test the risky procedure on himself. There are no prizes for guessing what happens next: in time-honoured Hollywood tradition, Caine performs the great disappearing act only to realise that he can't rematerialise. Worse still, the power-hungry scientist actually enjoys his new-found anonymity, and while his colleagues struggle around the clock to come up with an antidote, the out of sight ingrate starts to perceive them as a threat to his very existence.

Starring Kevin Bacon in the title role, alongside Elisabeth Shue, Josh Brolin, Kim Dickens and Joey Slotnick, *Hollow Man* is veteran director Paul Verhoeven's second film—after *Starship Troopers*—to boast 8-channel Sony Dynamic Digital Sound. In short, SDDS features an additional pair of left-centre and right-centre speakers on either side of the centre one, thus offering the director and sound crew additional creative and technical options. Indeed, spreading the soundtrack across five front speakers provides a wider soundscape for effects, music and dialogue, and this was especially useful on *Hollow Man*, where the primary intention was to use sound effects sparingly in order to reinforce the main character's point of view, while adding to his presence by using subtle effects in conjunction with the musical score.

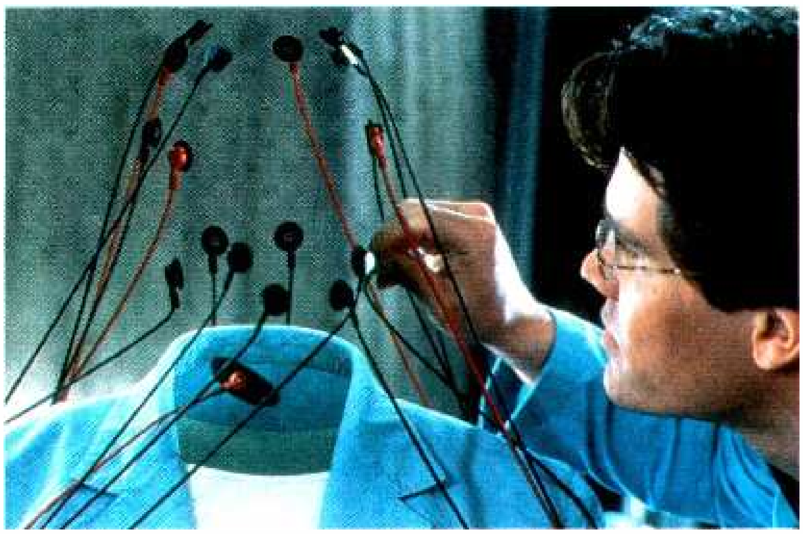
'Having access to five—rather than three—front screen speakers meant that we could isolate sounds more easily and make use of an extended dynamic range,' says the film's supervising sound editor, Scott Hecker. 'With SDDS 8-channel, effects and music have their own spaces to live in.'

The 1997 movie *Last Action Hero* was the first to be released in the SDDS format, and it has since been followed by titles such as *Erin Brockovich*, *Girl, Interrupted*, and *The Messenger: The Story of Joan of Arc*. Upcoming 2000 releases include *All The Pretty Horses*, *Charlie's Angels*, *Sixth Day* and *Vertical Limit*.

Audio postproduction on *Hollow Man* was completed recently on the Todd-AO West dubbing stage at The Lantana Centre in West Los Angeles. Under the supervision of Scott Hecker, Michael Minkler handled dialogue and music, and Gary Gegan oversaw the effects. Minkler worked on the aforementioned *Last Action Hero*, and Gegan gained SDDS experience on a movie titled *First Knight*, yet for Hecker this was his inaugural 8-channel project, and one that resulted in a soundtrack that is 'open, spatial and dynamic, without being cluttered. The music, in particular, is wide and rich,' he says.

The score for *Hollow Man*, composed by Jerry Goldsmith and delivered to Mike Minkler on 24-bit Pro Tools, was recorded in London by Bruce Botnick and mixed with the 8-channel configuration in mind. 'When the music is playing very much by itself and it is very large, you can really hear the articulation in the orchestra,' Minkler says.

The dialogue, on the other hand, recorded digitally and also cut on Pro Tools, was 'in excellent shape' according to Minkler with regard to when he first heard it. 'The production recordist did a fabulous job,' he says. 'He had a lot of practical locations, but he also had motion control cameras that he had to deal with, which make a lot of whirring noises. He used radio mics and boom mics, and it all turned out great. Where we did looping, it was mostly because of performance or some other artistic choice. Very rarely was it due to noise problems.'



The sound team, left to right, Mike Minkler, dialogue-music mixer; Ken Hall, music editor; Scott Hecker, supervising sound editor and Gary Gegan, effect mixer

Scott Hecker's involvement with the *Hollow Man* project commenced at the start of February, 2000, when he supervised the overall sound editorial process from his office at Todd-AO in Hollywood while dialogue, ADR and Foley work was taking place on the Sony Studios lot in Culver City.

'I initially sat down and spotted the movie with the director, film editor and producer,' Hecker recalls. 'I was there for the dialogue spotting and the sound effects spotting, and really the director didn't hear any sounds until we got to the predub stage because there were no temp dubs. Both the editors and myself would decide which would be the best effects to use. We would spot the reel, talk about how each effect would be used in conjunction with what other effects, and then, when the editor had completed what we'd talked about, the work was always reviewed and then it was refined according to my personal sensibilities.

'Paul Verhoeven is a very intelligent, discerning and demanding director. I'd worked with him in a different capacity on *Total Recall* and *Basic Instinct*, and so I'd already had some experience as to his tastes and inclinations. Every detail must be attended to, and at every turn you go through all of the available options in order to come up with the best choice. The beautiful thing about working with Paul and his producer Alan Marshall is that not only are they demanding, but they also give you the time and the financial resources to do the job that they're expecting. This is as opposed to a lot of other films that we sound people work on, where the demands may be just as high, but the resources and co-operation are not nearly as adequate.

'For instance, on *Hollow Man* Paul and Alan allotted me a 20-week post sound schedule, and they're very smart that way. They give you plenty of time, and so I basically felt like they gave me enough room to hang myself. I loved it. I knew what the demands were and I felt like, "Well, I don't have any excuse for not being able to deliver to them exactly what they want".'

He didn't need to. Benefiting from a comfortable schedule and the technological know-how, the sound work on *Hollow Man* was fairly straightforward. Nevertheless, certain scenes did present some interesting and time-consuming challenges.

'The film doesn't have car chases and it doesn't have gun battles, yet it is very insidiously busy,' says Hecker. 'For the effects we made a number of high-quality location recordings using 4-channel microphone arrays to capture front- and rear-oriented stereo

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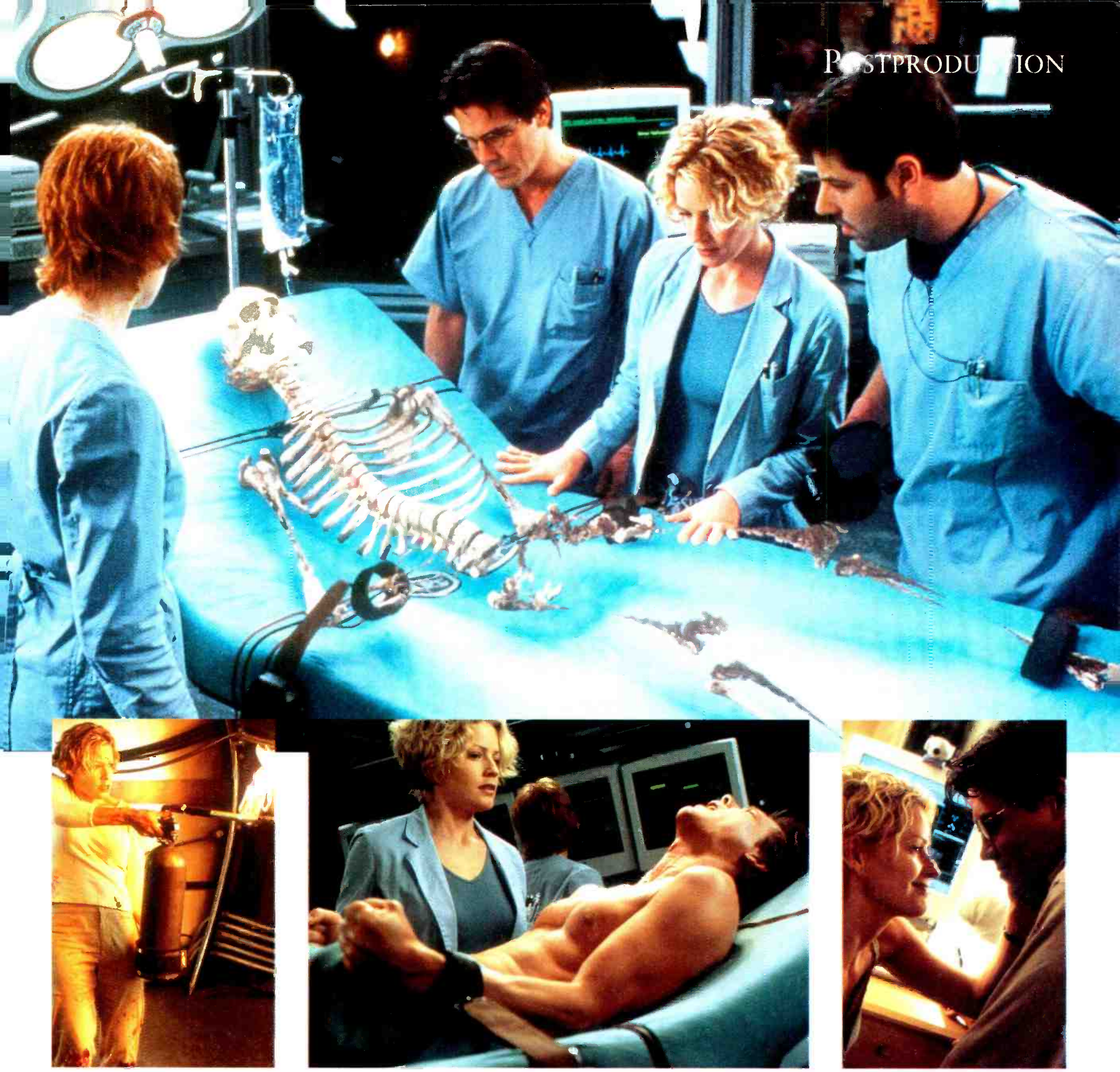

soundfields. Well, we didn't have to figure out how to do that, but sometimes just implementing it was a logistical challenge.'

Sample the big scene towards the end of the film in which the two lead characters are being threatened by a runaway elevator car. In an effort to record as many elevator-related sound effects—crashes, impacts, rattles, creaks, groans and falling debris—as well as the ambience of the elevator shaft, this called upon Scott Hecker, effects recordists John Fascal and Eric Norris, and assistant Carmen Flores to climb inside a shaft at the Edison Power Station in Eagle Rock, California, just north-east of Los Angeles.

'Just trying to figure out how to get into the shaft and mount microphones in certain positions there was very complicated, not to mention physically a little scary,' says Hecker. 'You know, hanging off of ledges and taping microphones to the sides. Obviously, we couldn't be in the elevator car to do that because we wouldn't get to the elevator wall that way. I mean, we knew what microphones we wanted to use—that wasn't an issue—but it was just a case of how do we get these mics mounted to the side of the elevator shaft, midway down? That was a bit of a hurdle to get over, but it turned out really spectacular, especially with the use of the SDDS format. We were really able to accentuate a lot of details.'

The same applies to spotting the exact location of the aforementioned invisible man himself, most notably in the scene where he chooses to play cat-and-mouse with his fellow scientists and fool them as to where he is standing. The liberal movement of his voice around the room makes for some ear-catching use of the surround channels, with very smooth pans between the speakers.

'Interestingly enough, we did a little preview here with our final dub and we had to dub it down to 5.1, and it was amazing how even then it retained its smoothness,' says Hecker. 'We didn't feel pans jump-

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ing around that much. I'd expected it to not sound as smooth, because we had been so spoiled listening to the SDDS all of the time, but it actually translated quite well.'

And then there is the fly, whose scene-stealing performance also necessitated extravagant use of all eight channels in one particularly notable instance. 'We'd fly it around the room, have it go off into one corner and sort of fade out a little bit, and then come back and swing around the room again,' explains Hecker. 'At various junctures you also see it appear on camera. That was the most obvious use of SDDS.'

Nevertheless, while scenes such as these—and another featuring a poor gorilla that is prey to the scientists' experiments—are among the audio highlights, perhaps the greatest impact of the 8-channel configuration is on a more general and subliminal level.

'There are a lot of small things that you can do to improve the soundtrack as a whole,' says Gary Gegan. 'Rather than just make a specific sequence jump out at the audience, you can generally improve the imaging. I mean, I'm not sure that having those two extra speakers is going to radically alter the way you would play a sequence, but the main effect of it is an overall improvement in terms of the clarity and the imaging and less distortion.'

At Todd-AO all of the sound effects were edited on Fairlight MFX3s, while the sounds were manipulated via Synclaviers and a variety of outboard processors.

'The Fairlight isn't the strongest tool as far as bending sounds is concerned,' says Hecker. 'It's just a beautiful editing tool. Most of the sound manipulation was done with Harmonizers, vocoders and the Synclavier itself, which has some really interesting processors, so we weren't limited in that way.'

Effects predubbing commenced at Todd-AO West in May, where the entire show was mixed on an Otari Premiere console.

'The exceptional sound editing team really paid attention to making sure that all of the sound effects could be articulated,' says Mike Minkler. 'There wasn't an abundance of overlapping material, it was very intelligently laid out, and so based on that and the high level of quality that the director demanded, we had the gratifying knowledge that we could make it all sound very special.' For his part, as the only person present who had heard all of the requests, wishes and demands of the director and producer, Scott Hecker served as the conduit of information appertaining to how things should sound.

'We were very experimental,' he says. 'Mike

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Minkler and Gary Gegan are very creative people, and we didn't do much of anything that was just down-pat. Using this SDDS layout with eight channels, we really experimented a lot with speaker placement—"Well, let's try it over here... let's try it on the hard left... maybe bend it in a little towards the left extra... let's try that in the surrounds..." There were always various permutations and combinations.'

Which is all very good, but doesn't the availability of extra channels sometimes encourage experimental overkill and a tendency towards Mickey Mouse effects?

'No, we had that filter on the whole time,' asserts Hecker, 'and because there were three of us in the room we all checked one other continually. From the get-go we are all quite sensible people, and we don't like gimmicky soundtracks. Definitely our main goal was to keep the film sounding very natural, while also dynamic and spectacular, without drawing attention to any one speaker. We spent a lot energy trying to make it seamless and not place things in the surrounds that would prompt audience members to whip their heads around and go, "Wow! What was that?" Instead I think the dub is very smooth and fluid, and we used all eight channels very effectively.'

'The basic premises of mixing a good soundtrack don't change with 7.1,' adds Gary Gegan. 'Theoretically it's the same, but you just have better tools to accomplish your goals. For instance, many times you get a stereo sound effect that you don't necessarily want to play extremely wide as that wouldn't be appropriate for what you can see on the screen. However, you would still like to keep it stereo, and it's much easier to keep a discreet stereo on the inners if it's a somewhat smaller image than in, say, 6-track, where your option is to start collapsing the stereo into the centre. That really just muddies things up, you get phase cancellation and all sorts of funny things can start to happen. By keeping it stereo you can really make it live.'

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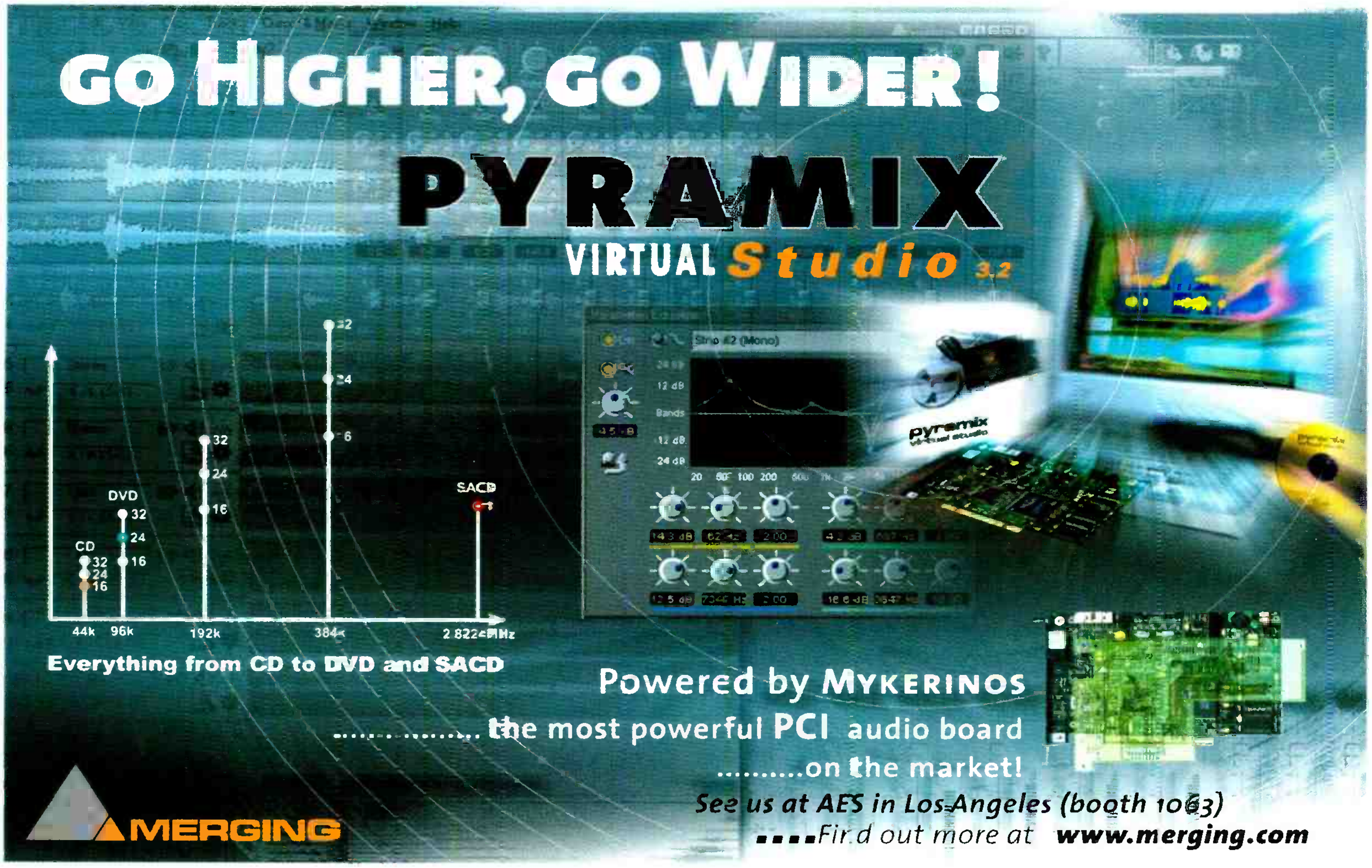

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BEATING THE SYSTEM

Leitch's Diamond Audio range was designed to enable a stereo broadcast infrastructure to meet 5.1 or multilingual broadcast needs. Leitch's **Ian Puszet** explains compression and application

WITH THE PRESENT RAPID RATE of technological progress, the consumer has come to expect more from the modern broadcaster. As video has moved from black and white to colour and now to high-definition, so audio has moved on. Currently broadcasters are transmitting stereo to the home but the consumer can get 5.1 surround sound from DVD and other media and now expects the same from the broadcasters. In response to this, Leitch's Diamond Audio is a range of audio compression products that will enable broadcasters to pass 5.1 audio—and potentially more—through the existing broadcast infrastructure (see Fig.1). Diamond Audio uses the apt-X compression algorithm explained here along with the applications appropriate to the Diamond range.

When choosing a compression system it is most important that the compression system best suits the application. In order to achieve this, these requirements must first be understood. Hence any compression-based solution must address the real-world requirements of an integrated audio-video contribution environment. These include the following: use of AES3 signals for transportation in standard infrastructures; nultiple passes with minimal degradation; low encoding and decoding delay (latency); robustness to errors; switching versatility (synchronous and asynchronous switching); capacity for extra data (metadata); synchronisation of compressed audio to video; auto detection between linear and compressed signals; integration into mixed audio-video and embedded environments; and concatenation with other compression systems.

The objective is that a compressed AES signal can in practice be recorded, switched, routed and embedded in a 'transparent' system in the same way as a normal linear AES signal. In view of theses requirements, the compression algorithm chosen for Diamond was the enhanced apt-X.

The existing apt-X 16-bit audio coding system is currently in use world-wide, in many applications. These include studio to transmitter, radio frequency and other fixed digital cable links, satellite and ISDN based outside broadcast links, and cinema and film theatre surround systems. The enhanced apt-X algorithm is an enhanced version of the existing apt-X 16-bit algorithm which aims to code, transparently and in real time, very high quality digital audio signals. It provides at the output of the coder a 16-bit (20-bit or 24-bit) word that represents four 16-bit (20-bit or 24-bit) linear PCM samples at the input, thus achieving 4:1 compression. In this application it operates at 48kHz and will deliver eight channels of audio with bandwidths of 22.5kHz with a total bit rate of 1.536Mb/s (1.92Mb/s or 2.304Mb/s).

The three key components of the algorithm which collectively achieve this degree of compression are: subband coding, linear prediction and adaptive quantisation.

The complete enhanced apt-X (coding and decoding process is illustrated in Fig.2. The input linear PCM audio sample is filtered into four frequency bands of uniform bandwidth using Quadrature Mirror Filters (QMF) (Fig.3). This process allows the audio signal to be eventually reconstructed again using a QMF in the decoder while at the same time retaining good

Fig.1 Leitch Diamond Audio

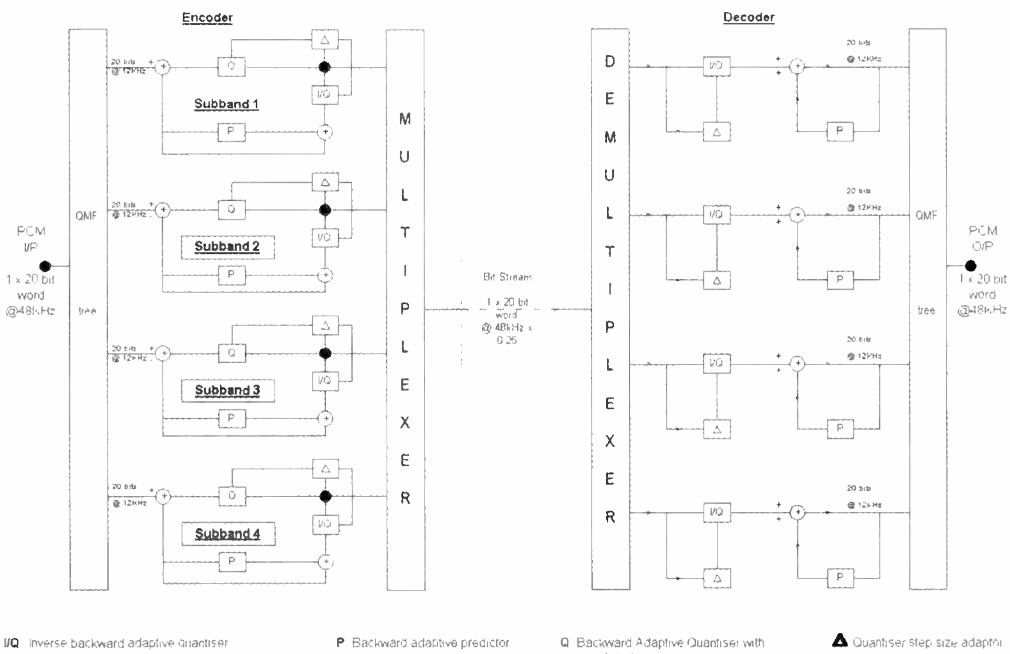

Fig.2: Enhanced apt-X coder and decoder

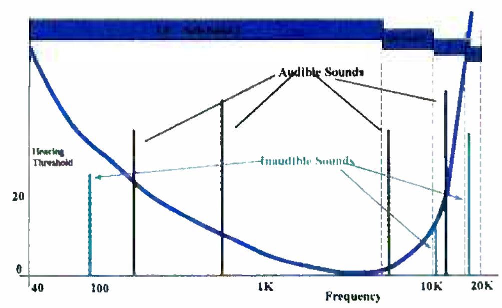

Fig.3: Subbands

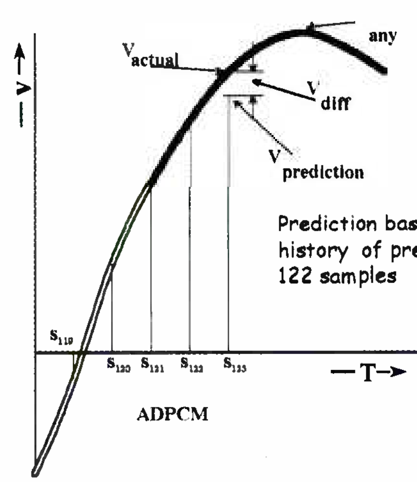

Fig.4: Prediction and difference signal

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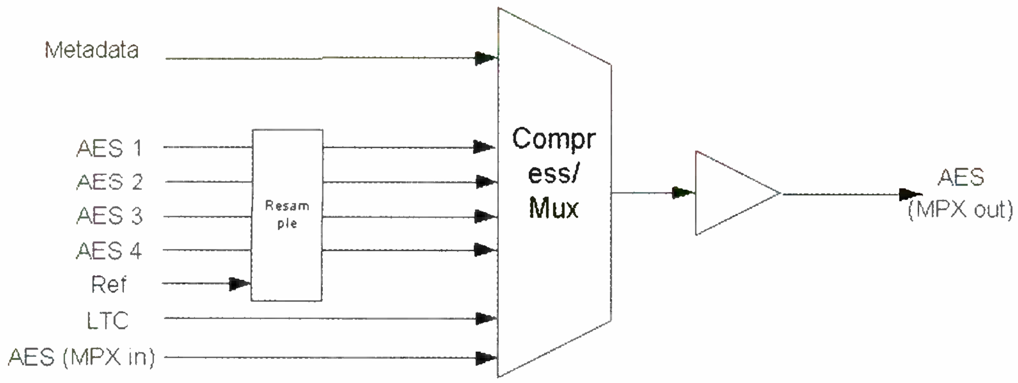

Fig.5: ACE-1600 audio compression encoder

signal quantisation noise ratios and constant coding delay within each band.

Each of these subbands is processed separately by a backward adaptive predictor. Redundant information is removed by subtracting a predicted signal, derived from coder look-up tables, from the incoming subband signal. This creates a difference signal (Fig.4) that may then be re-quantised. If the prediction is accurate enough then the magnitude of the difference or error signal will be quite small, much less than that of the original subband signal. Relatively few bits are needed to code such a signal effectively with a high resolution.

Each of these subbands is processed separately by a backward Laplacian quantiser. The coder uses a Laplacian quantiser with backward adaption that extracts the step size information from the recent history of the quantiser output. This avoids the problem of estimation delay and range transmission overheads. The backward adaptive quantiser operates with sufficient headroom to accommodate signals with the highest slew rates. Thus it is non optimal, and for most signals displays a variation in the signal to quantisation noise ratio in each of the subbands. The code words at the output of each subband section are then multiplexed to produce a single 16-bit (20-bit or 24-bit) word at the output of the coder.

The initial stage of the decoder demultiplexes each word and feeds each subband inverse quantiser with its respective code word. The output from each inverse quantiser is a difference signal. A signal identical to that predicted and subtracted in the coder, and similarly derived from look up tables is added back again to the now decoded error or difference signal. This enables the original linear PCM signal to be reconstructed with minimal loss of information.

Finally the four reconstructed, now 16-bit (20-bit or 24-bit), subband signals are inverse filtered and combined to form 16-bit (20-bit or 24-bit) linear PCM samples in the time domain.

Overall, the advantages of this compression system are that it is non psychoacoustic, it has a lowcoding and decoding delay (1.9ms per process), it is robust to random bit errors, has high accuracy (98%) and offers 16-bit, 20-bit or 24-bit quantisation resolution.

The Leitch Diamond audio product sets out to achieve one fundamental goal, this is to provide a simple solution to multichannel audio contribution on a single stereo infrastructure. There are two core products in the Diamond range, the Audio Compression Encoder (ACE-1600) and the Audio Compression Decoder (ACD-1600).

The ACE-1600 enables the multichannel (multilingual or 5.1 surround sound) broadcaster to encode up to eight audio channels into one AES compatible signal. Physically the encoder is a 1U-high box with up to five AES inputs, one LTC input, one metadata input, one reference input and one AES output (Fig.5). Depending upon the configuration required, the unit is available with BNC AES interconnections for unbalanced operation or 25-way D-type AES interconnections for balanced AES operation.

The unit receives the linear PCM signals for compression on four of the AES inputs. These can be synchronous or asynchronous with each other and the local station reference. If the signals are asynchronous then the unit has built in resampling circuits which will resample and lock the sources to the station reference. The reference for the resamplers is taken from the reference input which is selectable between black and burst or AES reference. This ensures that the audio channels will remain in phase and frequency locked to each other. If the sources are already locked to the station reference the resamplers can be bypassed.

The LTC input enables the user to carry LTC with the compressed audio signal. This is a useful feature as it allows the user to compare the LTC through the audio chain and the LTC through the video chain. By doing this the user can calculate the delay (if any) between the two chains and therefore adjust the delay to match, ensuring no lip sync errors will occur. Likewise it is also very useful when editing the audio.

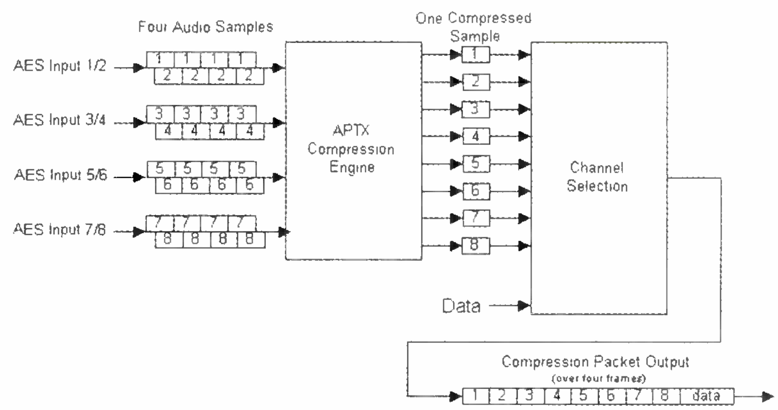

Fig.6: Compress

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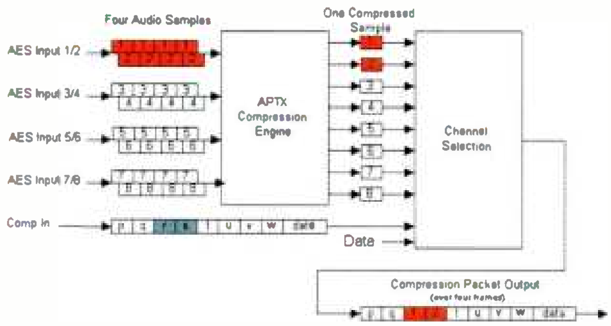

Fig.7: Replace

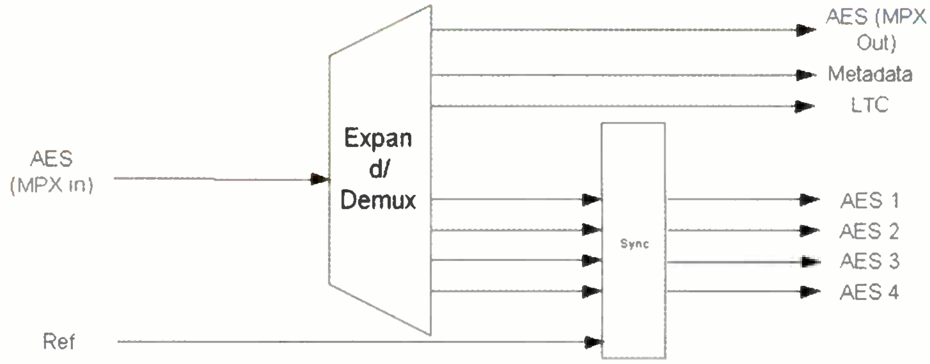

Fig.8: ACD-1600 audio compression decoder

The last, metadata, input allows the user to carry metadata along with the audio. By combining the metadata with the compressed audio, the metadata will always have the same delay as the audio and therefore be relevant to that audio and not perhaps out of phase resulting in possible artefacts.

The AES output from the unit is the compressed audio. This output can be a 16-bit, 20-bit or 24-bit AES signal, and is therefore compatible with all AES distribution equipment provided the equipment does no processing (sample rate conversion, word size reduction, or floating point conversion) and passes the relevant number of bits. So for example, a 16-bit video embedder will not pass a 20-bit AES signal but will pass a 16-bit AES signal.

The fifth AES input is an already compressed AES signal input from another ACE-1600 source. This input allows the unit to carry out an additional function over and above the normal compress (Fig.6) function, the replace function. The replace function allows the channels of audio to be replaced by new channels of audio without the need to decompress and compress all the channels (Fig.7). This is useful for voice-overs, station ID and censorship.

The ACD-1600 audio compression decoder is the complementary piece to the ACE-1600 encoder. Like the ACE-1600 it is physically mounted in a 1U-high frame and is available with balanced or unbalanced AES inputs and outputs. It has five AES outputs, one AES input, one LTC output, one metadata output and a reference input (Fig.8). Four of the AES outputs are the decompressed AES source signals, the fifth output is a loop through of the compressed AES input which is a compressed feed from an ACE-1600 encoder. The LTC and metadata outputs are the extracted LTC and metadata corresponding to the extracted AES audio. The decoder performs one basic function and that is to decompress the audio from compressed to linear PCM audio (Fig.9).

The decoder has to be slightly more intelligent than the encoder so that operational errors can be compensated for. Operational errors may mean that the decoder will have a linear PCM signal fed to its input instead of a compressed input. In this case the unit bypasses the signal to all its AES outputs with a matching delay as if the unit was still decoding. Alternatively the compressed audio may be mapped incorrectly within the compressed feed. The unit has the ability to remap its AES outputs. For example in a multilingual installation English may be swapped with German, the decoder can swap them back again. The last but certainly the most important is that the decoder can automatically detect the status of the compressed stream coming into it and therefore apply the correct decompression method—it will autodetect

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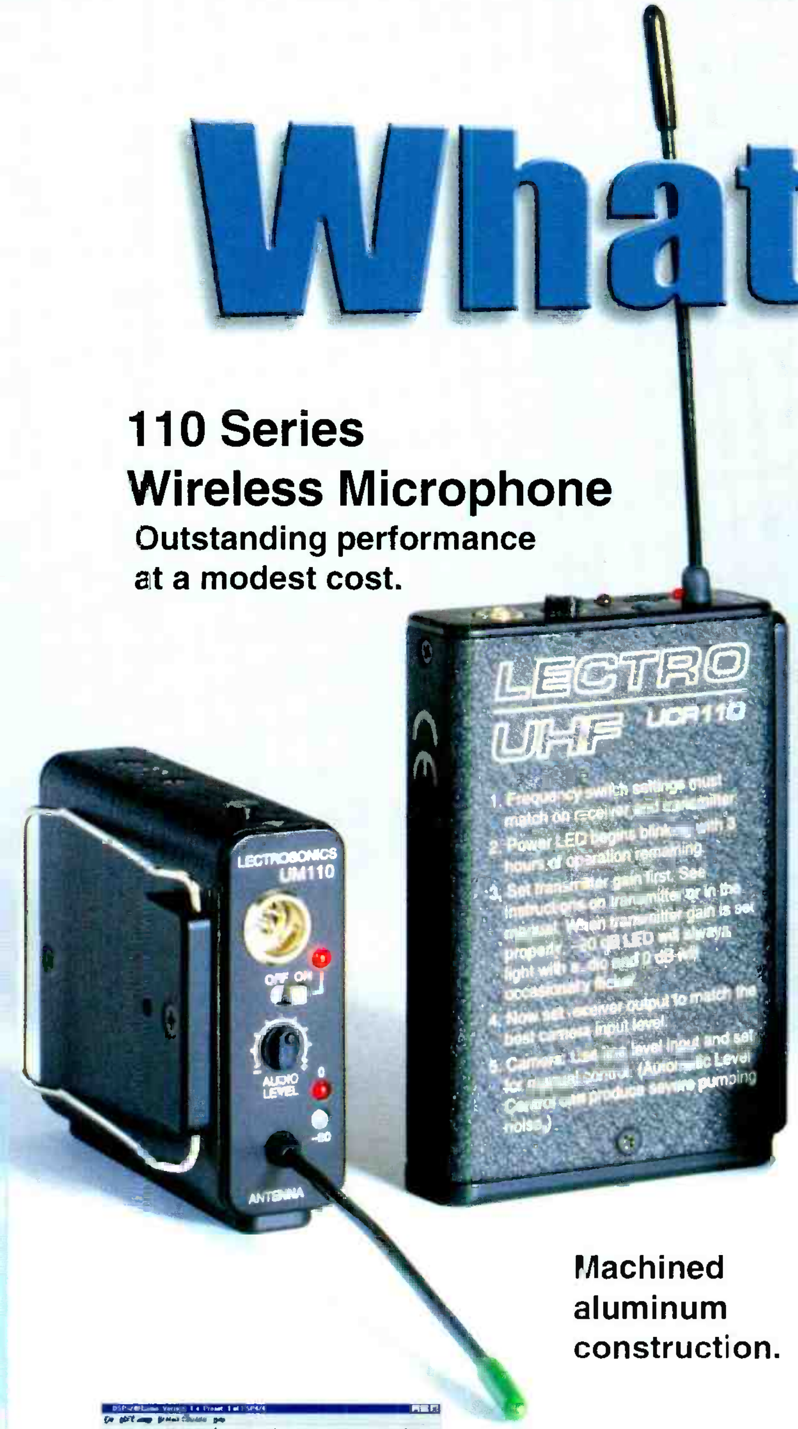

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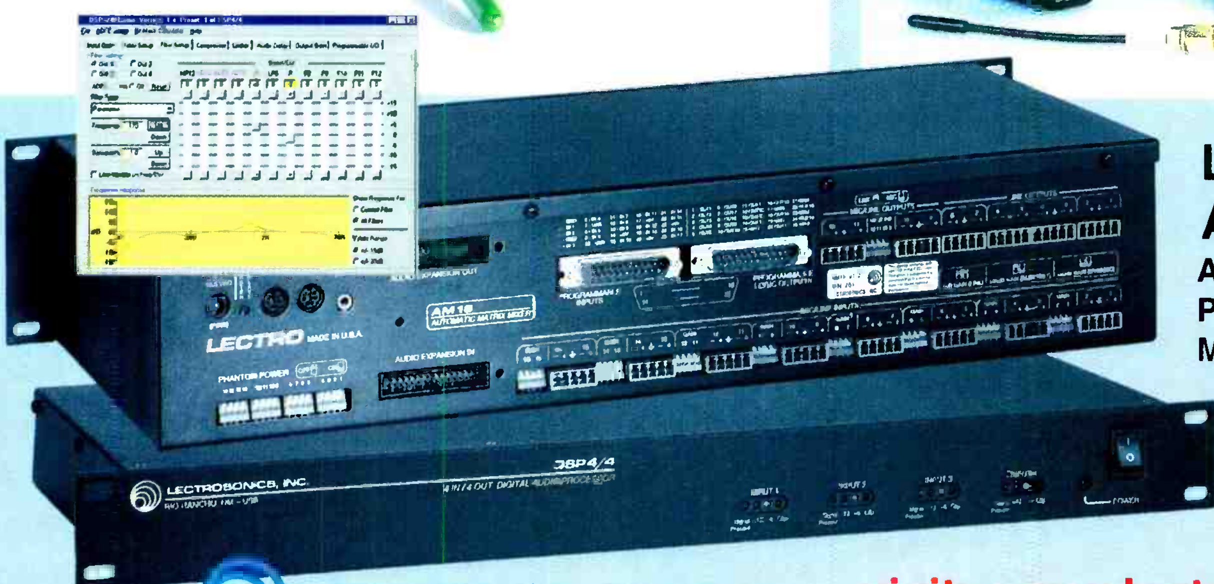

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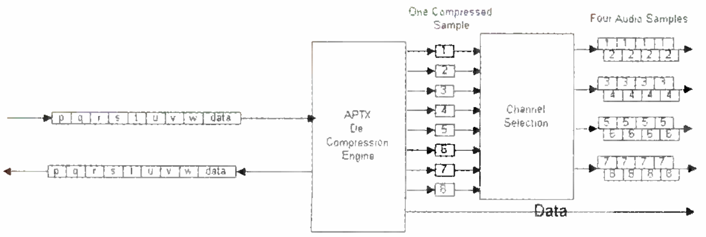

Fig.9: Decompression

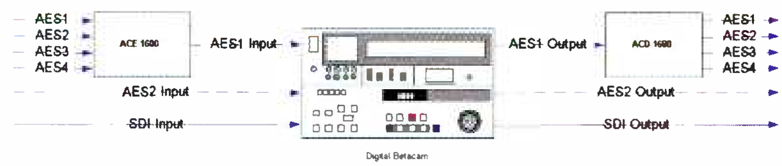

Fig.10: 10-channel audio recording-playing

the AES transport, the AES quantisation and the number of channels.

Finally one unique feature to the Diamond products is the handling of AES source and destination IDs. The linear PCM signal has within it channel status and part of this channel status is a four ASCII character Source ID and a four ASCII character Destination ID. These IDs exist for each discreet channel within the PCM signal. Using these it is possible to label each channel with a unique name. For the multilingual broadcaster it could be the language, for the surround sound broadcasters it could also be the language or it could be the channel—left surround, sub...

The Diamond range was designed to simplify the installation of a 5.1 or multilingual broadcast installation. The basic problem here is that the broadcaster has until now only been broadcasting a single stereo language and as a result the infrastructure has been designed for this purpose—it will only carry two or four channels of audio. To meet this need, system designers and manufacturers have designed their systems and products to cope with this requirement. As a result VTRs record typically four channels of audio.

The broadcaster having identified the problem has a choice of two possible solutions. The first is to expand the infrastructure. So, for example, if the target is eight channels (5.1 + LtRt), and currently it is a 4-channel AES system, then extra D–As, six extra levels of AES routing and extra wiring would need to be installed. This is expensive and finally falls over when it comes to recording the signal because, as already noted, most existing VTRs only record four channels of audio. The alternative is to install a second machine (a DAT), which will record the channels that the VTR cannot record. Although this is a viable solution it is open to operational error. The programme maker will now be supplying the source material on two tapes and human error may mean that the wrong DAT tape is inserted with the VTR tape.

The second solution is to use audio compression. This route is more simple than the first, as the first principle here is to use the existing infrastructure and just compress the AES audio at the ingest point. There are several reasons why this is simpler, the first is that having compressed the AES channels there is no longer the need for a DAT machine to record the extra audio. This results in the programme maker supplying one tape to the broadcaster and therefore the correct audio will always be broadcast with the relevant video.

Typically, the ACE-1600 would be on the front end of the contribution system, or alternatively on the input to the recording media. With these boxes the bottleneck created by the recording media is thus solved. By putting an encoder on the input side of a VTR the tracks are expanded by a factor of four, and putting the decoder on the output side will expand the audio back to its baseband AES form (Fig.10).

To conclude; by using compressed audio as against expanding the infrastructure, the broadcaster is able to use the existing infrastructure with little or negligible change to operational practices. As a result the system design is simplified, the operation is simplified, and the scope for error is greatly reduced. □

The author would like to thank Fred Wylie and the Audio Engineering Society for their help in the production of this article. 'Ref: apt-X100: low bit rate subband ADPCM Digital Audio Coding', by Fred Wylie, published in Collective papers on Digital Audio Bit-Rate Reduction, pp83-94, Sept 1996.

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STAR TURN

Antares' unique software has established a profitable niche for itself in the effects plug-ins market. Vice president of marketing Marco Alpert talks to **Simon Trask** about the Auto-Tune effect

IF THE EFFECTS plug-ins world has a star turn, it's Auto-Tune, the pitch correction plug-in from a small Californian company called Antares Audio Technologies. Auto-Tune first blazed across the plug-ins firmament in 1997, starting out as a TDM plug-in. To reach a wider user base Antares also produced a stand-alone version for the Mac; this ran within an audio I-O shell program, AudioStream. The stand-alone version in turn led to plug-in versions for MAS and Mac VST as users clamoured for a version that was integrated into their software of choice. To this list today can be added versions for DirectX (the Windows platform) and now RTAS (just recently introduced to plug in, so to speak, to the Digi 001 market).

Also available is a scaled-down, entry-level version, Auto-Tune LE for Mac VST, which provides the software's Automatic mode, but not its tweakable Graphical mode. Introduced at the January 1999 NAMM show as a more affordable version of the Auto-Tune software for users of budget VST-based programs, the LE version can be upgraded to the full Auto-Tune VST offering.

'We've had tremendous success with Auto-Tune,' confirms Marco Alpert, Antares' vice president of marketing. 'It was the right product at the right time, and as a brand name it's almost universally recognisable in our particular little niche of the world.'

It was the success of Auto-Tune, and the resulting decision to bring out a hardware rack version, the ATR-1, that led to what had previously been essentially a one-man operation evolving into a fully fledged company, complete with a CEO, Stephen V Tritto, and a VP of marketing, Alpert, both of whom had worked previously at E-mu Systems (as president and VP of marketing respectively). Alpert had also done a stint consulting for Akai's EMI division before the opportunity to join Antares came up. Today Antares is up to around 10 or 11 people strong, spread around three different facilities divided into engineering, marketing, and sales and distribution.

The roots of Antares actually go back to 1990 and a company called Jupiter Systems, that was set up by the man who would later be responsible for creating Auto-Tune, Andy Hildebrand. From 1976 to 1989 Hildebrand had worked as a research scientist in the geophysical industry, first with Exxon Production Research then with Landmark Graphics, a company that he both co-founded and helped guide to a successful IPO. In fact, last year he was selected by the Society of Exploration Geophysicists to receive its Enterprise Award for his breakthrough work in the development of the geophysical industry's first stand-alone seismic data interpretation workstation. What Hildebrand had done, during his time at Landmark Graphics, was to develop a way to visualise the Earth's crust using 3-D acoustical data from seismic surveys—work that would, as it would turn out, allow him to approach musical audio problems from a new perspective.

'What he did in geophysical exploration was largely digital signal processing on repetitive waveforms,' explains Alpert. 'Now, the frequencies were way different than what we deal with in the music world, but the principles were more or less identical. You're looking at waveforms, sonic exploration in the oil industry for instance, and you're shooting waves down into the ground and analysing what comes back and trying to develop a picture of what's down there. That's where he came from and even within that industry he developed entirely new ways of looking at such things.

'In our industry there's music DSP literature. Some of it comes from Stanford, some from IRCAM, some from practice within the industry over the years, and that's what people turn to. When Andy came over into the music industry, he came over with an entirely different DSP literature as his resource.'

It was in 1989 that Hildebrand decided to return to one of his first loves, music, and went to study music composition at the Shepard School of Music at Rice University. When he found himself faced with the problem of creating seamless sample loops of multiple instrumentalists for his music, in true one-man-and-his-garage tradition he set about devising a solution, drawing on his background in DSP in the geophysics industry.

'It was a given in the sampling world back in those days that you couldn't do seamless loops in section sounds, because you didn't have one nice clean repetitive waveform,' recalls Alpert, himself a veteran of that sampling world through his years with E-mu.

Positive reaction to the solution that he came up with convinced Hildebrand that there was a market for his ideas, and he formed Jupiter Systems to further develop and market the technology, which was released in 1992 as a Macintosh program called Infinity. Today the stand-alone program is still part of the Antares software lineup, a testament to its enduring value.

After developing Infinity, Hildebrand turned his attention to the nascent effects plug-ins market, first developing MDT (Multiband Dynamics Tool), one of the early Pro Tools plug-ins, that was released in 1994. He followed this up with JVP (Jupiter Voice Processor) in 1995 and SST (Spectral Shaping Tool) in 1996 before developing Auto-Tune. Today MDT and JVP are still in the Antares product lineup, as TDM-only products.

Also in 1996 Jupiter Systems became Antares Systems. Then, when the company incorporated in 1998, it became Auburn Audio Technologies dba (doing business as) Antares Audio Technologies. That same year Antares also merged distribution company Cameo International into its operations. Back in 1994 Hildebrand had drawn on the services of RiCharde and Associates, a distribution company specialising in plug-ins, to provide sales and distribution services for his software.

When Auto-Tune came along, it became RiChard's largest-selling product line, and founder Neil RiCharde merged his company with software and CD-ROM distribution group Invision Interactive to form Cameo International and focus primarily on Antares products. The subsequent merger with Antares became a logical step, with RiCharde assuming the position of VP of Business Development.

While software plug-ins are often seen as the 'sexy' side of technology these days, hardware still has its own attractiveness.

'There are still a lot of studios out there that have racks and racks of effects devices, and there's a reason for that,' says Alpert. 'As the workstations become capable of supporting more channels, you're going to run out of power on your computer if you want to run lots of DSP processes on lots of channels. Plus you also have the added risk of updating your system and half of your digital recording functionality disappears until the conflicts are resolved. Particularly in time-critical environments, then, there's still something seductive about having a piece of gear that does one thing, that always does it, that doesn't have to share its resources with anything else. We want to serve both those markets. We don't have a particular sense that one is more important than the other, it's just that, given that we have some unique technology, we want it to be available—and obviously want to be able to profit from it being available—to as wide a part of the market as possible.'

And Alpert says that it's a lot easier for a small company to get into hardware these days than it was a decade ago.

'We're sitting in the middle of Silicon Valley, and one of the things that Silicon Valley offers is very cost-effective contract manufacturing,' he explains. 'Back in the early days of E-mu that wasn't readily available; we had to invent our own manufacturing department, hire people, get space and invest in inventory and manufacturing equipment. For a small company that's a huge risk, and in the old days it used to make getting from software into hardware an enormous undertaking. Now a small, modestly capitalised company like we are can move into hardware with a lot less risk than was possible as recently as eight or ten years ago.'

Another factor that makes getting into hardware a lot easier these days is the integration of more and more functionality into fewer and fewer components.

'The integration of computer DSP functions over the last ten years has been amazing,' says Alpert. 'Going back to when I joined E-mu full time, there were three

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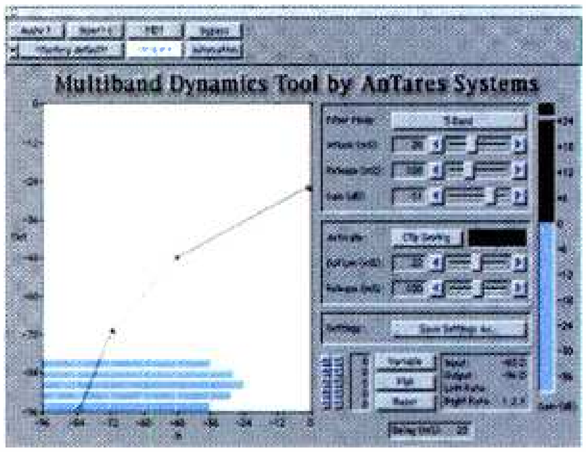
of us there, and even though I had the rather grandiose title of General Manager I would sit there with a soldering iron and build the circuit boards in those first microprocessor polyphonic keyboards. And it would take me three 8-hour days of hand-soldering to build one board for that keyboard. And that entire function could probably be programmed into two chips today.'

In another way though, the situation has come full circle—to the benefit of small companies. As Alpert explains: 'When things were analogue you didn't need anything custom, you just built out of the various analogue components that were available. Then when things started getting digital there was this barrier to entry where you had, for the most part, to get custom VLSIs if you wanted any special functions. The investment to get those chips was a $100,000 or more, and you had to commit to huge quantities, and a huge lead time, and it was an incredible risk. A company had to be of a certain size to even play in that arena. Nowadays it's crossed back over. The general-purpose DSPs you can go out and buy on the market are getting more and more powerful, but also cheaper and cheaper. So there's zero risk, you're buying off-the-shelf parts and just writing the code for them, and once again it means that small companies who never would have had the resources to get a custom chip can get into the fray.'



The latest addition to the Antares product line-up is the AMM-1 Microphone Modeller plug-in, introduced last September. Again initially a TDM plug-in, it has subsequently been released in versions for MAS, VST, DirectX and, most recently, RTAS. Based on Antares' patented Spectral Shaping Tool (SST) technology, as its name suggests Microphone Modeller creates software models of the sonic characteristics of a variety of microphones, which users can then draw on to create the sound of the desired mic. The plug-in also gives users control over mic-specific options such as low-cut filter on-off, as well as features such as mic placement and wind screens; new models are made available for download from Antares' web sites (www.antarestech.com and www.antares-systems.com both work).

At this year's January NAMM show Antares previewed new speaker modelling technology, again based on its SST technology, which it says will allow artists, producers and engineers to use their current monitors to audition mixes through virtual digital models of the sonic characteristics of a wide variety of speakers; in addition it gives users control over the effects of speaker placement. The company plans to release Speaker Modeller as a TDM plug-in in the Summer of this year, with a stand-alone rackmount version to follow in the Autumn.

With its small, yet unique product range, Antares has sought to maximise its user base—and therefore its revenue—by making Auto-Tune and now Mic Modeller available for not only TDM but also a range of native formats, as listed earlier. Alpert reveals that the TDM and DirectX versions of Auto-Tune—which in most cases represent the opposite ends of the user-spectrum—are actually neck and neck in terms of the revenue they're bringing in, while VST comes somewhere in between. Clearly, then, what the DirectX version loses in terms of price and profit per unit it makes up for in terms of sheer market numbers.

'On the DirectX side you've got a lot of people who are just dabbling,' says Alpert. 'But there are so many of them that even a small portion represents real money. Also there are a lot more people using PCs in a professional environment these days.

'There are very few dabblers in the TDM world, though, because there's such an investment in learning time, and, of course, financial investment, that these people, with very few exceptions, are people that are making a substantial portion of their living from using their TDM systems. If they see a tool that will save them just a little bit of time, or will do anything just slightly more than what's in the standard system, they don't think twice about buying it, because at the hours they bill, or the value of their own time if they're working on their own projects, it will pay for itself in a month.

'So, although TDM's priced at the top, the Mac Native formats are down from that, and then DirectX down from that, there's sort of an expectation there on the part of the various users, and rarely do we get any comments about pricing.

'But then, one of the things about Auto-Tune is how fast it processes. You get over to the DirectX side and you're limited by the fact that DirectX itself makes it impossible to guarantee a certain amount of the CPU's power in a timely manner, so you do have to deal with the vicissitudes of Bill Gates' scheme over there. You're dealing with latencies that are inherent in the computer platform rather than the software, and you can't really predict, necessarily, from one moment to the next what that's going to be. So, there are certainly advantages in performance that you get from the TDM systems. Although, assuming that the soundcard is good enough there shouldn't be any difference between DirectX and TDM in core audio performance.'

The most recent platform to get the Antares treatment is Digidesign's Digi 001, with the port of Auto-Tune and Mic Modeller to RTAS format.

'We had a slight wait-and-see attitude at the very beginning with it,' admits Alpert, 'because frankly we didn't have much choice, we were working on Mic Modeller TDM so we didn't have a lot of resources to do the port. But it's clear now that there's going to be a substantial Digi 001 installed base out in the field. The indications we get are that Digidesign is doing very well with the 001. They're clearly one of the strongest brand images in our market for what they do. Being able to put that brand behind the 001 is really powerful for that product, and they're selling a ton of them. Now we're all waiting to see how many of those people lust after plug-ins as much as the more expensive TDM people do. We're certainly hoping for us it's going to be a good undertaking.'

Another way in which a small company with in-demand software technology can grow is to license to third parties. This is a direction that Antares has started pursuing, and the first result is a deal with Mackie. This has seen Auto-Tune made available as a third-party plug-in for the mixer company's D8b digital mixer, via the mixer's new UFX card and OS 3.0 software.

'We have two things to offer: we have the functionality of our products, and we have the brand equity that goes with Auto-Tune and hopefully will also go with Mic Modeller,' says Alpert. 'As a small company we can't conceivably pursue directly all of the markets that these things could address. So we're given the choice of either not pursuing it and watching someone else come in and meet those goals, or working through non-competitive, compatible strategic partners. We've been aggressively pursuing the second choice.'

No high-tech market stands still for long, and the plug-ins market has evolved to a point where, according to Alpert, companies need to offer something beyond the conventional.

'In the early days there wasn't a lot of functionality standard in the platforms themselves, so things were really ripe for plug-ins. You didn't get a decent reverb or a decent compressor or whatever, so you could always see the value of adding a high-quality one. There are a lot of little companies that are still successfully turning out a suite of conventional channel-strip effects that I think should be worried. Because very soon if not now in a lot of cases you'll get virtually everything you need for standard channel-strip processing right in the platform.'

Antares is in a good position through having technology that provides a distinctive addition to the standard effects functionality. The success of Auto-Tune has given the company a strong base from which to build growth. And through licensing deals such as the one with Mackie it's clear that there's still mileage to be had from that core software technology. At the same time, Antares is more than a one-product company that struck it lucky with Auto-Tune.

'The uniqueness factor has allowed us to grow to where we are now,' Alpert acknowledges. 'If we'd come out with just another good compressor and another good limiter and another good de-esser, the quality would be great, and we'd sell some, but it would be hard for us to build growth on that sort of thing. So our corporate goal is to continue to introduce innovative new technologies. We want people to be thinking 'What is Antares going to come out with next?'. Because they're going to know that whatever it is it's going to be something interesting. We need to make a splash with unique stuff because we don't have the resources to crank out endless products quickly. At least not yet. But we recognise that continued growth is also going to depend upon a wider range of products that are attractive to a larger part of the market.

'Our priority now is to grow our company, and that means releasing the products that we have in the stream on time, at the right price, with the right functionality, so that people buy them and like them. It means finding the right strategic pro audio partners, like Mackie, and looking for other places to put our technology through additional strategic partners. Also looking for investment to bring more capital into the company to allow us to grow faster. We know we can execute the plan we have in place right now self-financed, which is quite an undertaking these days, but we also know that with more resources we can execute it faster and reduce the risk by getting the products out sooner, before any of our competitors have a chance to do anything like them.' □

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MEETING METADATA

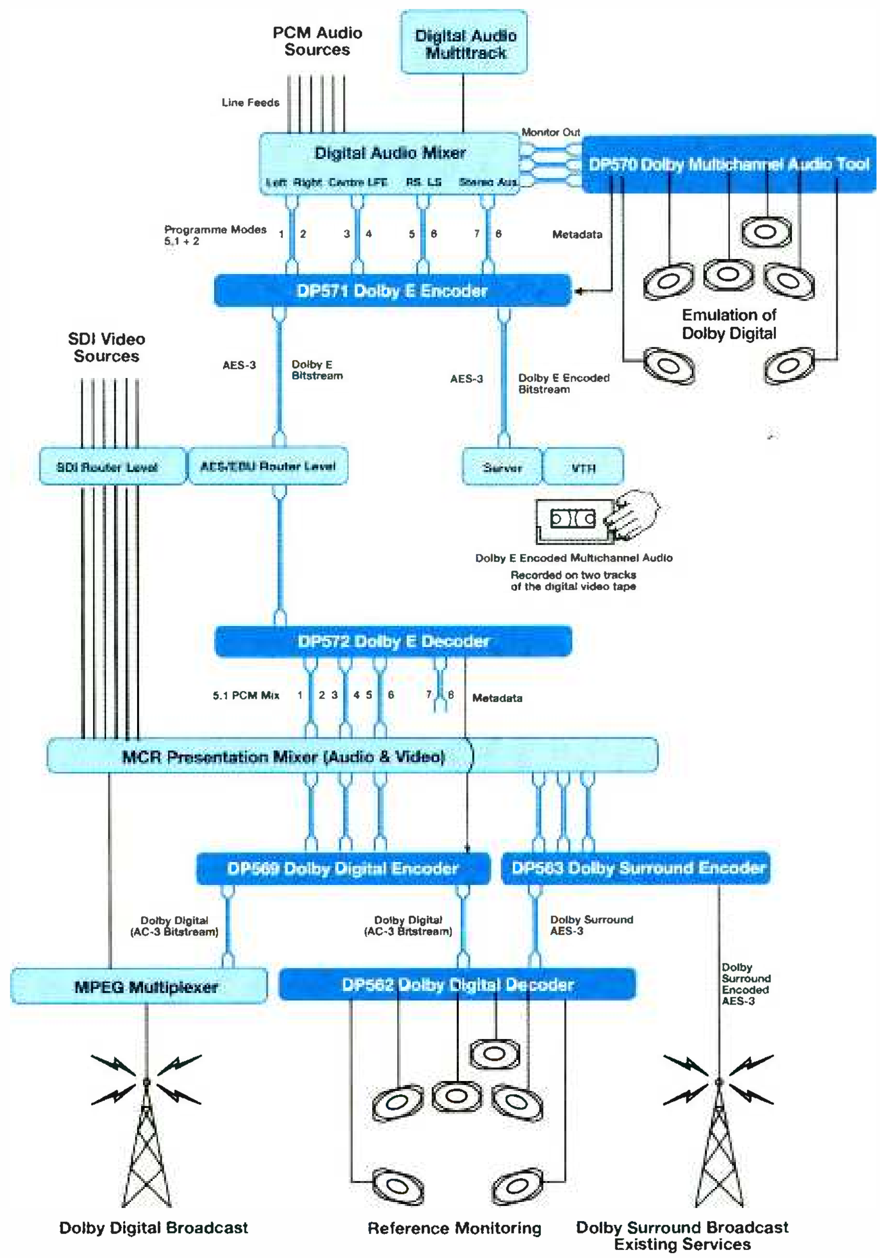
There are buzz words and keywords; some count and some don't. Dolby Laboratories' **Peter Cole** explains why, in the burgeoning world of multichannel sound and all the opportunities it offers creators and listeners, metadata counts

IN EARLIER ARTICLES, *Studio Sound* has looked at Dolby Digital and Dolby E, explaining how the two technologies are integrated to carry the 'metadata' required by the consumer decoder from the postproduction environment to the home. Metadata is basically 'helper' information that the home decoder uses to optimise the incoming audio feed for the playback audio system. The same Dolby digital bitstream can be decoded in the home to provide 5.1-channel, Dolby Surround Pro Logic, stereo and mono versions of the original material.

This article will explore how metadata can be created within the professional mixing environment by the audio professional. However, before we dive into metadata creation, here is a quick recap:

Dolby Digital is an advanced form of digital audio coding enabling the storage and transmission of high-quality digital sound. This technology can provide 5.1 channels (five discrete channels, each with a range of 20Hz–20kHz). Speaker configuration is typically positioned Left, Centre, Right, Left surround, and Right surround. The '.1' refers to a bandwidth-limited LFE (Low Frequency Effects) channel (100Hz–7kHz). Dolby Digital is widely available on the majority of DVD titles and is the only universal multichannel audio standard available. All currently available consumer DVD players contain a Dolby Digital decoder as standard.

For broadcasters, Dolby Digital is the mandatory audio transmission standard for the ATSC standard and a legitimate option within the DVB standard, where the broadcaster may elect to use Dolby Digital as the only audio, in preference to MPEG audio. Dolby E encodes up to eight PCM channels with associated consumer metadata onto a single AES-EBU pair, can be recorded on broadcast VTRs, servers and routed via 2-channel infrastructure such as routers, telecom and satellite systems. Dolby E can also be encoded into an MPEG transport stream for contribution applications. It also uses higher bit rates, allowing for up to ten encode-decode cycles at 20-bit depth and has frame lengths that match the video frames. This means that Dolby E sources can be cut and edited on frame boundaries without any artefacts and in sync with the associated video feed. Dolby E in postproduction allows metadata parameters to be included along with the mix. This metadata is then carried within the Dolby E bitstream all the way through the distribution chain, decoded back to PCM, and then encoded to Dolby Digital for final emission. The metadata is connected from the Dolby E decoder to the Dolby Digital encoder via RS485 protocol.



Creation of metadata from post to home

Dolby E has been designed for professional use only and will therefore not appear on consumer equipment.

With metadata, a single high-quality, wide dynamic range 5.1 audio bitstream can be adapted to mono, stereo, Surround; and, if required, at reduced dynamic range, as clearly a loud cinema soundtrack will not replay from a single mono TV speaker-at least not for very long. Metadata also brings about the end of having to control programme levels with a compressor. This is because the ideal actual replay level of the programme can be transmitted to the decoder. The result is simply better audio. The key metadata parameters which are required for transmission to the decoder are: Dialnorm (a dialogue-average level indication used for level setting, that enables different programme material to be replayed at the same average level, without compromising the dynamic range—particularly useful for minimising the loudness in some commercials). Downmix parameters (for setting centre and surround levels in a stereo downmix or other output configuration). Dynamic range control (information the decoders can use or ignore, depending on the capabilities of the audio system being used in the home).

The creation of metadata is the big question for all broadcast professionals, producers, sound engineers, and systems designers. Dolby has developed a product called the DP570 Multichannel Audio Tool. This enables the audio professional to create the metadata at the key point in time, during the final mix.

The DP570 provides two main functions: metadata selection and receiver emulation. These functions allow creators of multichannel Dolby Digital audio programmes to create the metadata for that programme and to monitor the effect of their choices.

The product takes eight channels of audio input and modulates the input according to the metadata values, compression and downmix settings chosen by the user. The modulated audio is available as either a digital or analogue output, and metadata is chosen via the user-interface. The metadata choices are then sent to

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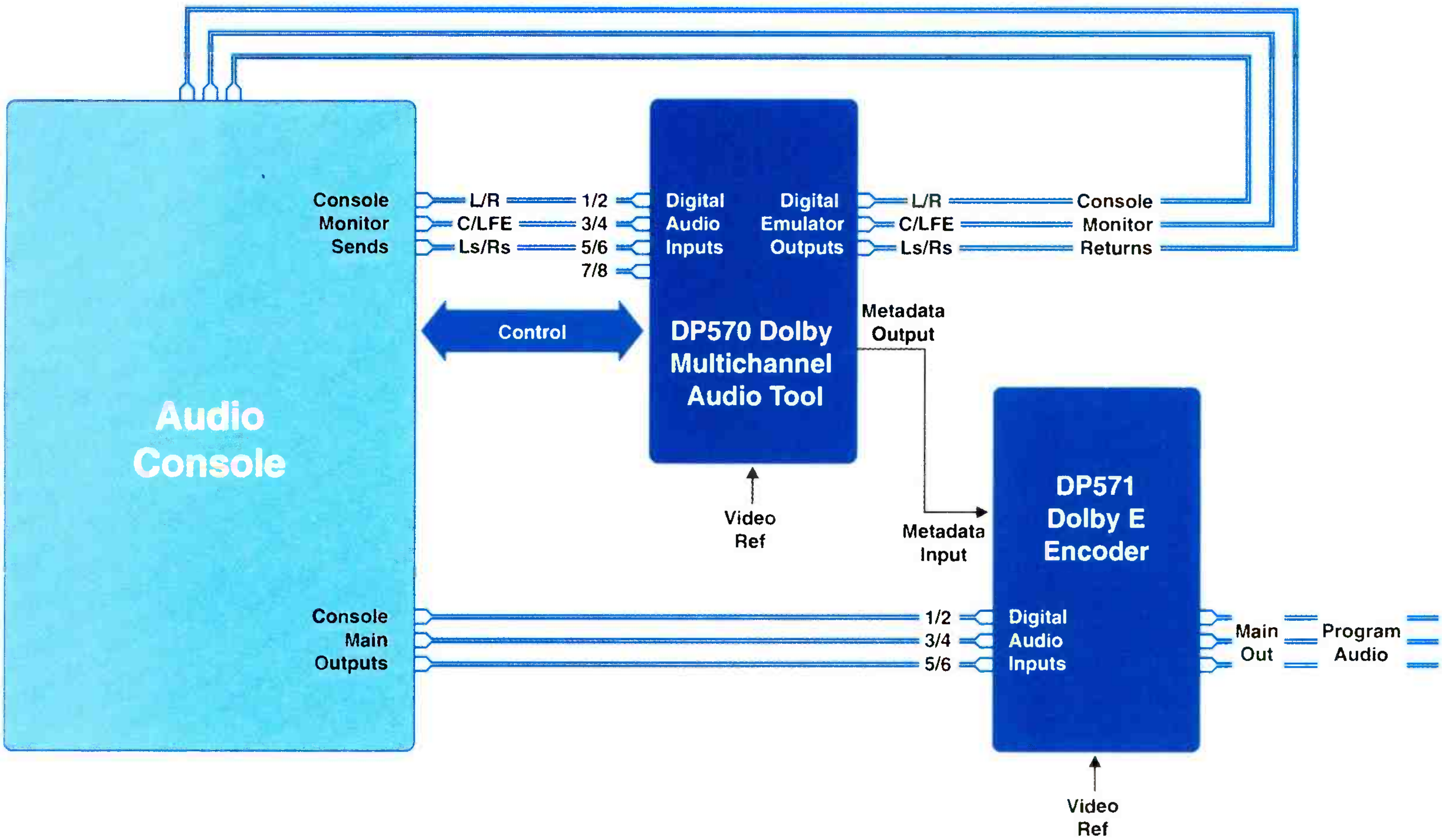

A typical integration of a console and DP570 Dolby Multichannel Audio Tool

the Dolby E or Dolby Digital encoder as a complete metadata stream.

The DP570 provides several features that simplify the process of creating multichannel audio programmes. It provides audio routeing to compensate for the various channel configuration formats. It also provides metadata authoring, while allowing the user to listen to the effects of the metadata via Dolby Digital Decoder emulation. Finally, there is an extensive monitoring section that eases the setup and use of a multi-speaker monitoring system.

Examining the signal path for metadata, the obvious place for authoring is during the production of a programme. This may happen in the studio or OB truck, if the finished version of the programme is produced there, or in a postproduction facility, if the programme elements finally come together there. However, it can be very useful to have some of the metadata defined earlier in the process. The dialnorm, for example, can be used as an indication of the headroom required by a programme element. Other parameters, such as the dynamic range control profile can only be defined for the final mix.

The DP570 Multichannel Audio Tool can be integrated with a console, so a familiar control surface can be used to author metadata parameters, and monitor their effects in real time. Stereo and mono downmixes, for example, can be auditioned—and trimmed—using metadata, so the logical place to set the values is from the console. A variety of different control interfaces on the DP570 make this already possible with a number of today's consoles. Further benefits are possible using metadata for console

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hard disk recording comes of ease

MX-2424... the full power and potential of 24 track recorcing and hard disk editing in just a single unit.

record TASCAM's new MX-2424 will fly "straight out of the box"; recording and tracking like any tape based MDM — providing 24 tracks of stunning 24-bit/ 48kH sound (or 12 tracks of 24 bit/ 96kHz audio*).

comp & edit Switching to "Loop Mode" unleashes the power of non-linear editing and the ability to record and store unlimited "virtual" tracks for later comp'ing. Powerful functions are all accessible from the front panel controls, including a Jog/ Scrub wheel to emulate the "rocking" of analogue tape for locating edit points; visual editing and transport control is possible with the supplied ViewNet MX™GUI software for either Mac™ or PC.

analogue and digital recording Simultaneous digital and analog interfacing allows for 24 channels of 24-bit analog I/O and a further 24 channels of TDIF, ADAT™ or AES/EBU digital I/O.

how far do you want to go? In addition to its internal ultra fast SCSI drive, you can... use the front-panel drive bay and rear wide SCSI[1] port to extend recording time, or create backups, with up to 15 additional external drives; store your hard drive in a machine room up to 40 feet away; chain up to 32 machines for a huge integrated system with true single sample accuracy. The MX-2424 will, of course, sync and chase to whatever else is in the studio[2].

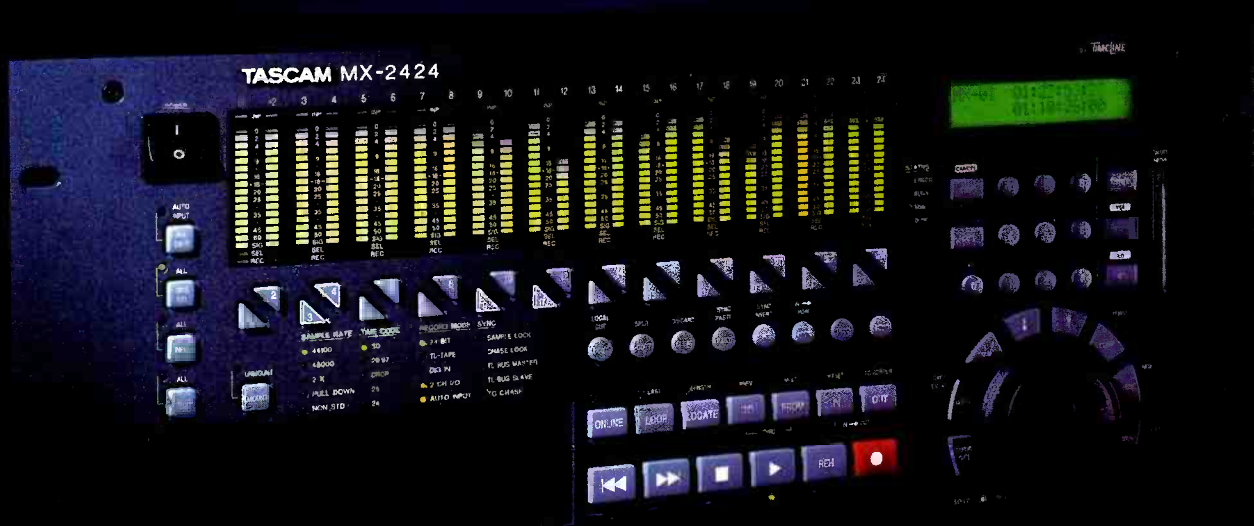
who's talking your language? The MX-2424 really shows its breeding as it reads and writes Sound Designer II files for the MAC or Broadcast WAV files for the PC* offering a multi-platform system. Moreover, it will playback, record and edit to ProTools™, Avid™ and SADIE™ format files, without even going near OMF – though it will happily work with that too.

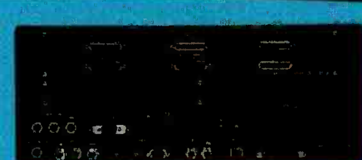
how much? And one final surprise — the price. The MX-2424 is several hundred — if not thousand — pounds less than you imagine.

[1] non SCSI bus protocol, such as IDE, might seem attractive to your accountant but no-one is ever likely to bring their session in on it.

[2] use of the optional TL Sync™ further enhances control and sync capabilities.

HARD DISK MULTITRACK RECORDER MX-2424





The MX-2424 (shown with optional IF-AN24 analog I/O) features built-in SMPTE Sync, MIDI Time Code, MIDI Clock, Video Sync, stereo AES/EBU and S/PDIF ports and much more. The MX-2424 is available today from you authorized TASCAM dealer.

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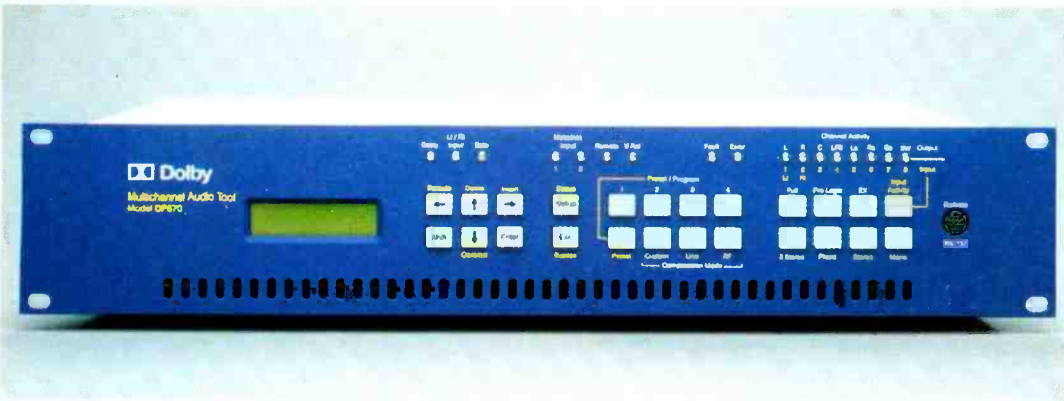

DP570 multichannel audio tool

operations. If consoles are able to read incoming metadata streams, professional metadata (such as that identifying the audio streams and individual constituent channels) might be used to autoconfigure the console channel strips appropriately, while at the same time writing ASCI text to the console fader legend strips. Dolby Laboratories will be making this know-how available under license.

In practice, a sound engineer will be able to mix for optimal 5.1 audio and establish some in-house metadata presets for the programme genre currently under production. It will be very easy to quickly monitor the mix in stereo, Dolby Surround, or mono modes by the easy selection of the monitoring button.

Between facilities—that is, in the general distribution infrastructure of OB, studio links, intra studio routeing, feeds to transmission, and so on, AES-EBU digital audio is becoming the norm. The use of Dolby E to carry either multiple audio streams or multichannel audio streams down those existing 2-channel audio paths will also provide an ideal way to carry metadata, synchronously linked to the audio to which it pertains.

The DP570 will be ideal for live DTV audio production, postproduction, network distribution and DVD authoring applications by slotting into the audio path.

Dolby has developed the DP570 with an open strategy to enable the unit to be controlled and integrated with many other broadcast products and systems. These include: the console interface as a more advanced serial interface that will function as a full remote control using Dolby remote protocol. This protocol will be made easily available to developers. Many leading console manufacturers have already planned to implement the software that would run on their platform, for seamless integration between console and the DP570. Control of the DP570 will also be available from a PC using the standard serial port. The GPIO is intended to provide a simple contact closure and tally interface that will activate user-defined presets or other key features. By providing these features via GPI, systems integrators can quickly and easily provide an integrated solution.

Dolby Digital enables a radical new approach for broadcasters to deliver high-quality multichannel audio to the home, catering for other existing audio configurations within the same Dolby Digital bitstream by the utilisation of metadata.

The creation of metadata is simple, and allows the audio professionals to achieve a level of audio compatibility across different systems and programmes that has hitherto been impossible. □

DM100 digital multichannel analyser

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Watermarking making waves

Far from clearing the air, the recent London listening sessions have added further concern to the proposed audio watermarking system writes **Barry Fox**

I'M MAKING NO APOLOGIES for writing another progress report on DVD-Audio watermarking. The audio industry will live for decades with the decisions now being taken by the music industry, and there are whopping contradictions in what we are being told.

The AES Audio Watermarking Workshop is set for the morning of 23rd September, Saturday, at the LA Convention Centre during the 109th Convention. Tony Faulkner will chair Karlheinz Brandenburg, Malcolm Hawksford, Paul Jessop (IFPI-SDMI), George Massenburg, Glenn Meadows, Joseph Winograd (Verance) and Al McPherson. There may be listening tests. But the point of it all remains as unclear as the point of the London listening tests arranged by Malcolm Davidson of Sony Music, and held at the Sony-CBS Studios in Whitfield Street.

Although no-one from 4C was present in London (indeed it is hard to see how 4C can now be an independent tester, given that 4C has formed LMI to licence Verance technology and collect royalties for Verance), Paul Jessop of the IFPI was there to represent the SDMI and David Leibowitz was there to represent Verance. As reported, some participants were very critical of the music quality. Even my cloth ears were disturbed by the sound of a laptop hard drive, which was louder than quiet passages of the music, and odd effects on decaying piano notes.

Leonardo Chiariglione, executive director of the SDMI, has since complained that the SDMI was not asked for comment on these criticisms. This is odd because during the listening test Tony Faulkner heard the dCS 954 D–A convertor muting and wondered how it was being clocked; clock irregularities would cause jitter and adversely affect sound quality, explaining some of the odd effects heard (on decay, for example), invalidating the results. One possibility is that the D–A was clocked to an older piece of Sonic Solutions kit, rather than the incoming AES audio data. Did anyone use an Audio Precision dual domain tester to compare the phase and framing of the word-clock and data? Faulkner put this to Malcolm Davidson and Paul Jessop. But at the end of July, Tony Faulkner was still waiting for comment.

Both Chiariglione and Jessop have said that there will now be further tests of watermarking at DVD-Audio level (176.4kHz and 192kHz), because the original 4C tests and London tests were at 96kHz and the material was up-converted digital or analogue. Paul Jessop has assured that nothing will be passed for use at an untested quality level. Leonard Chiariglione says that further tests will take into account 'the constructive suggestions that have been made regarding improvements to this listening experience.'

As 200 SDMI members each now pay $20,000 a year to be members, there should be no shortage of funding for these further tests. Paul Jessop has said that the further testing will embrace the London results.

But DVD Audio players are already on sale in Japan and the US, and due for launch in Europe in September. Panasonic Europe and Panasonic UK, who are launching the first players, were blissfully unaware of the London tests. David Leibowitz Chairman of Verance said in London that the DVD-Audio decision was 'done and complete'.

Koji Hase, who heads the DVD Forum in Japan, has confirmed that from 1st October all DVD-A players must integrate a Verance detector. He says the decision follows the 4C tests held last year.

'The system has been tested by more than 50 very critical "golden ears" recommended by the major music studios... the Verance WaterMark has shown the superior sound quality and robustness required by 4C and the SDMI. We at the DVD Forum wish to rely on this proof of quality inspection. Results were analysed using statistical methods. We felt very comfortable that all the people participated in this test process agreed upon the test results as credible. Verance sound quality has successfully met the critical requirements of the copyright owners and music studios as tested by the "golden ears".'

But the London tests meant so little to the SDMI and 4C that Hase asked me for information on the procedures and results! So on one hand we are being told that the choice of watermarking for DVD-A is still open and on the other hand the matter is closed.

Those who took part and were present in London, including myself, were promised the results as soon as they were available. These results have not yet been communicated. But both Ted Abe of Panasonic-Technics in Japan, and Leonardo Chiariglione, write about the results of the London tests as if they have seen then in final form. Indeed Chiariglione refers to them as 'facts'. Says Ted Abe: 'According to the latest reports regarding the London listening test, we are very confident that the watermark system is quite transparent. The result is almost identical to what was done last Summer, jointly with various golden ears of US major studios. Very few golden ears have detected the embedded WM, but they recognised it as the acceptable quality. Most of the golden years have shown no reliable detection on the basis of statistical analysis.'

So when will the participants, who were promised the results, get them? Will these results acknowledge the objections raised on the music and reproduction quality? Does the statistical analysis take these objections into account? Or, with DVD-Audio such an obvious loser, does it matter a damn?

Will these results acknowledge the objections raised on the music and reproduction quality? Or, with DVD-Audio such an obvious loser, does it matter a damn?

It started with sampling

The technologies troubling established lines of entertainment distribution have only just begun to make their effects felt writes **Dan Daley**

HERE'S A BIT OF ENTERTAINING LOGIC that won't appear on the back of your cereal box this morning: that which can be digitised can be easily stored and distributed. That which can be easily stored and distributed can be easily stolen. Virtually anything can be digitised.

Ergo sum: Oh, shit!

We have arrived at the logical conclusion of the digital audio revolution that began nearly 20 years ago. Napster and MP3 may be getting the media's attention in all the news weeklies and the Times, but these softwares are simply the evolutionary extension of the more widespread and available ability to make music. Once you make the creative process more accessible, it's just a matter of time before the distribution process catches up.

If there is any pride in having been at the forefront of this audio equivalent of the tumbling of the Berlin wall, enjoy it, because it will be quite fleeting. The real implication of all of this is actually a bit disturbing. What's happening is that we have also put into motion a process that establishes a disincentive to create music for commercial purposes.

Read that last sentence again: A disincentive to create music for commercial purposes.

The very process that has made the making of music simpler, more accessible to more people and thus democratised the music business like never before, is also up ending the economic basis for the music business. Not to sound like some kind of audio Cassandra, but a part of the world is ending, and it's the part that attracted many of us to the music business in the first place—the potential to make a living at it. People are not going to stop making music—it's too much a part of human nature to do so. But the sheer amount of it that has been created in the last 20 years—the number of new releases annually in the US has grown from around 15,000 a scant ten years ago to closer to 50,000 today—has commoditised music. That in and of itself might be enough to weaken the underpinnings of the music industry as we know it. It certainly was enough to radically alter the recording studio industry, wasn't it? But combined with the ability to store music in a very convenient form (MP3) and then to be able to make it move effortlessly and without restraint throughout the world (Napster), the incentive to capitalise the entire process on any major level diminishes dramatically.

We see the results of this already at work. Universal music, the mega-corporate scion owned (and enlarged to behemoth proportions by further acquisitions such as PolyGram's music assets) by a Canadian distiller, is now in the process of being unloaded onto a French utilities corporation turned media company. (There are so many potential jokes in that one sentence, I don't know where to start, so I'll leave it alone.)

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Another media behemoth, Sony, continues to try to quash rumours that it, too, wants to dump its entertainment media holdings, specifically its music and film businesses. Not for nothing is it an historical fact that no foreign-owned company has ever made a profit in US entertainment holdings. The handwriting, quite simply, is on the wall: it will become increasingly difficult, to the point of nearly impossible, to make a profit on intellectual property in an era in which the object of our desires has become so fluidly ethereal and—the truly pernicious aspect of all of this—as a result is conditioning an entire new generation of music lovers to the notion that the stuff is free to start with.

As much as we have ranted and railed about the mega-corporations that have run the music business for the last century, it's only now, as they begin to depart the landscape like so many dinosaurs, that we realise how critical they were, because they capitalised it. They funded it. They provided the lucre on a scale similar to that of NASA or the Ministry of Defence. They will now go off and fund other, more profitable things. Ten bucks says I'm right.

Back to pro audio. We saw this coming. The whole sampling *mishigas* foreshadowed it. Stealing sounds and riffs was a cornerstone for house and hip-hop, and it did produce a few indignant musicians complaining about being ripped off. But nothing on the scale of Metallica appearing before Congress in July explaining to lawmakers (some of whom were younger than the band members) how their livelihoods were being threatened by Napster (created by someone who is younger than all of them). They're not singing, 'Pay Bo Diddley' now. They're scrambling to cover their own assets. Maybe this is all some sort of Biblical retribution for stiffing Little Richard.

Or maybe this is all about personal responsibility, something the music business has not had a surfeit of historically. The real reason we sample other people's records is because we can. The real reason college kids use the university library T-2 server to download Metallica songs for free is because they can, too. Most college kids in the US can afford $15 for a CD. (The real irony of all this is that the price of CDs will almost certainly come down drastically in the next year or so, to compete with downloads.) We all do this because we can. The same digital technology that facilitates this new distribution model comes with technical instructions, but not any inherent moral obligations. That's for us to insert into the equation.

Because, ultimately, that's all we really control. We can set the parameters on the compressors and EQs, but the tectonic movements behind all this are beyond our abilities to direct. No one could have predicted the situation the music industry finds itself in now even five years ago. But based on the recent past, we can make some reasonable predictions about the next five years. As broadband proliferates, music isn't the only thing that's going to be stolen on a regular basis. Movies, television shows, every form of entertainment will eventually be subject to digital larceny. And no matter how sophisticated the watermarking and other digital countermeasures the industry comes up with to blunt this trend, they will all be brought to naught by someone like the bright, bored 17-year-old in Norway, who cracked the CSS2 code last year and delayed DVD-Audio by over six months.

So I guess the only thing left to try is a little personal responsibility on the part of individuals, and hope that somehow it takes on a global scale. We have to find a way to make it cool, along the lines of the 'Just say no' anti-drug campaign. Or at least cooler than stealing songs is. I'm open to suggestions.

The alchemy of broadcasting

The dynamics of language have reached such extremes that they are in danger of turning our technicians into philosophers, writes **Kevin Hilton**

LANGUAGE is important as it is how we communicate with others. But sometimes it can get in the way. JB Greenough said: 'Language is the felicitous misapplication of words.' A recent survey by a recruitment agency underlines this: it found that the biggest turn-off for potential employees was bosses spouting such inanities as 'Let's get our ducks in a row' and 'We are a global player.'

There seems to be the view that because technology is changing and that it is changing how the business of music and broadcasting is done, new words have to be invented to describe some of the activities. We are all getting used to talking in a new media, or multimedia, argot, but my jaw dropped recently when someone spoke about 'metadata and essence'.

Metadata has quickly established itself as part of the brave new lexicon of broadcasting. Regrettably, the more it is used, the less it seems to mean. At a recent conference on asset management, observers noticed delegates' eyes glazing during a discussion on the subject.

Even the seemingly helpful definition 'data about data' does not help anymore. As multimedia grows in many different directions, the data that is carried and delivered is diversifying and, by extension, so is the data that accompanies it. Part of the problem is that these matters push us dangerously into the area of semantics.

The delivery and distribution of such creative products as television and radio programmes, films and music should be practical; defining metadata is too esoteric and academic an activity. In recent months I have found myself having almost ethereal, philosophical conversations with broadcast professionals concerning the correlation between wide-screen pictures and surround sound.

In John Carpenter's flawed movie *Prince of Darkness*, which draws inspiration from quantum mechanics, a student says a professor does not want to produce scientists, he wants to teach philosophers. It is such a change of attitude that is at odds with the increasingly ruthless, fast-paced business of media today. Attempting to define a created word like metadata is something that might have sat more comfortably during the time of Lord Reith.

Getting etymological, meta comes from the Greek and means either something being in a position behind, after or beyond something else or denoting a change of position or condition. Neither of which relate to the notion of data about data. This last definition pertains to a raft of information, including copyright details, data about the video, when something was created and how many times it has been screened.

Things get complicated when a company or organisation comes up with its own definition of something. Dolby's is in relation to its Dolby E multichannel audio distribution format and is closer in spirit to the definition of denoting a change of position or condition. 'Every decoder needs metadata to determine what to do with the 5.1 audio mix,' explains Peter Cole, marketing manager for pro audio at Dolby UK. 'Metadata can be used to derive the stereo or mono version of the 5.1 if the consumer only has a 2-channel system. Metadata ultimately means that the broadcaster or production house can mix for the optimum of 5.1 and add information about how it can be folded down into stereo. It means there is no compromise.'

Dolby's concept of metadata is as control information, which is slightly at odds with everyone else, who see it as a glorified log sheet. Both are relevant, but, as the future would still appear to be built around the consumer's set-top box, being a control stream for optimum performance seems more compelling. As important as the housekeeping information is, human apathy could defeat the idea.

In the days of tape, how many contained comprehensive session sheets? Many survived with hastily scribbled notes in chinagraph pencil on the reel itself. It may be that typing the information into a special computer text box will be easier and more attractive but, honestly, how many name diskettes when formatting them?

Ultimately, studio operators merely want the system to work. They do not care how it works or what it is called, which is why metadata is becoming a multi-meaning, meaningless term. As for 'essence', this is an example of thinking that because times have moved on, the words used to describe a particular thing are outmoded. 'Essence' means the intrinsic nature or indispensable quality of something; I can see the logic but applied to a recording of whatever kind, it is senseless and unnecessary.

Railing against changes to the language is usually the preserve of deeply conservative, intransigent people. I swore I would not do it myself. But Essence is a buzz word too far. Programme material may be two words, but at least they mean exactly what they say. Technology has introduced many new words to the dictionary and this can only be a good thing, as it enriches the language. But sometimes what we call something is not as important as what it is about or what it does.

Railing against changes to the language is usually the preserve of intransigent people. But Essence is a buzz word too far. Programme material may be two words, but at least they mean exactly what they say

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DR JOHN

SIDEBANDS

Audio sidebands show up in a surprising number of places in audio working, but where do they come from and how are they used? **John Watkinson** explains

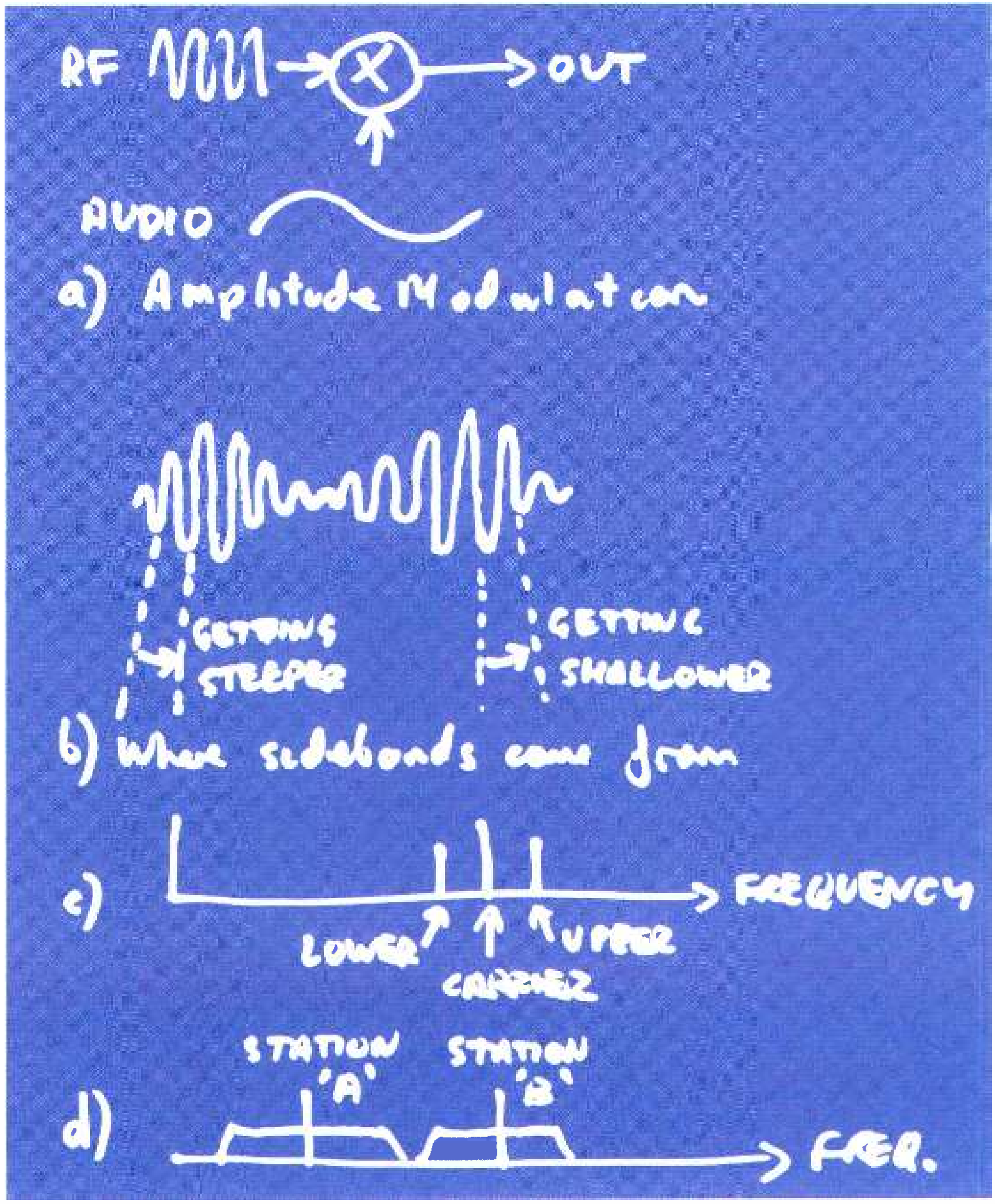

Fig.1 Sidebands in AM radio transmission

SIDEBANDS ARE A MODIFICATION of the spectrum which result when two signals interact. This can be deliberate and beneficial, or accidental and detrimental depending on the application and can take place in the analogue or digital domains. Whether the goal is to create sidebands or get rid of them, knowing something about them is useful.

Fig.1a shows an AM radio transmitter. A pure sine wave or carrier at the nominal station frequency is multiplied by the audio waveform so that the envelope of the carrier is the same shape as the audio waveform. After modulation, the carrier is no longer pure. Fig.1b shows the example of modulating the carrier with an audio frequency sine wave. During the increasing part of the carrier envelope, successive cycles must get larger and this means the slope of the waveform must get steeper. This has the effect of increasing the frequency. On the other hand, during the decreasing part of the envelope, successive cycles get smaller and the slope of the waveform must reduce, lowering the frequency. As a result the spectrum shown in Fig.1c is obtained. The carrier is still present, but some energy has moved to a pair of sidebands, equally spaced above and below the carrier frequency.

Fig.1d shows that with real audio programme material, the baseband spectrum is repeated mirror fashion above and below the carrier frequency. The carrier of the next station has to be far enough away in frequency so that the sidebands don't overlap. This is the reason AM radio sounds are so dull: the audio bandwidth is deliberately reduced at the transmitter to allow the stations to be placed closer together in frequency.

Fig.2 shows an AM signal being demodulated. The carrier frequency disappears and the spectrum shifts down to the audio band. The upper sideband becomes the conventional audio spectrum and the lower sideband becomes the negative audio spectrum in which all frequencies are negative.

What, then, is a negative frequency? There are a number of ways of considering negative frequency. One way is to argue that it really exists, but that to human observers negative and positive frequencies appear the same. The other is that it is a quirk of the mathematical models we use to predict what systems will do. As long as we get the required result does it matter?

Fig.3 shows a rotating wheel with a mark on the perimeter. Viewed along the axis, it is obvious whether the rotation is clockwise or anticlockwise. If the wheel were driven with a DC motor, we would be quite happy with the concept of positive and negative terminals and with the motor having negative rotation if the terminals were reversed. However, viewed in the plane of the wheel, the mark just rises and falls with a sinusoidal waveform and the same result is obtained whether the motor runs forward or reversed.

This means that a sine wave could have a negative frequency or a positive frequency and we wouldn't know which. It is a useful concept to assume that a sine wave contains equal amounts of positive and negative frequency. This is consistent with transform duality (Dr John, *Studio Sound*, August 2000). Fig.4 shows that in the time domain we are comfortable with the concept of a signal having a history and a future because this is what happens if we put T=0 in the middle of the waveform which is then symmetrical. In the frequency domain we would also expect symmetry, and if we put F=0 in the centre, we get positive and negative frequencies.

Thus a sine wave can be considered to be the visible result of a pair of contra-rotating processes. Fig.5 shows that if two wheels are geared together and the motion of two marks on the peripheries are summed, the horizontal motions always oppose and cancel whereas the vertical motions add to produce a sine wave.

If the entire contra-rotating assembly were to be itself rotated, the frequency of one wheel would rise and the other would fall, producing an upper and lower sideband pair. This is exactly what happens in helicopters. The blades twang like guitar strings and this vibration alters the tension at the blade root. However,

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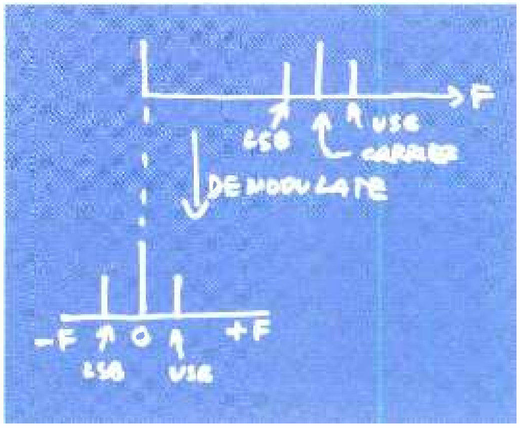

Fig.2: Negative frequency

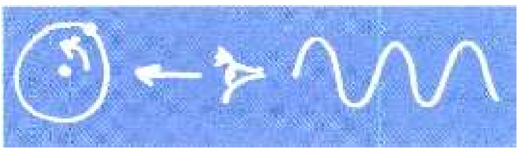

Fig.3: Sine wave is one dimension of a rotation

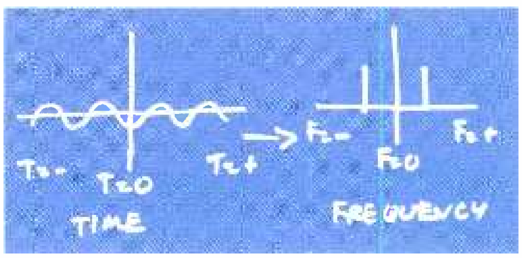

Fig.4: Where negative frequency comes from

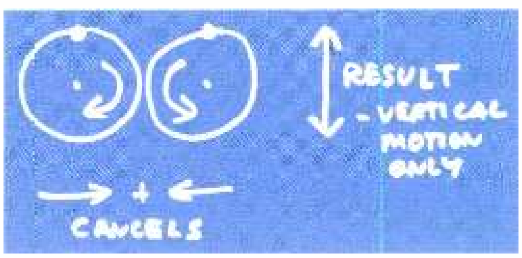

Fig.5: Frequency can be considered as opposed rotations, one positive and one negative

the vibration frequencies seen in the hull are not the frequencies at which the blades are vibrating. The rotational frequency of the rotor adds to the positive frequencies of the vibration and subtracts from the negative frequencies so that a pair of sidebands above and below the rotor frequency are created.

Suppose the entire assembly of Fig.5 were to be rotated at its own frequency. One of the rotations would be cancelled out and the other would double in frequency. This is how a stroboscope works. Flashing a light once per revolution will appear to halt rotation, and that is a time domain explanation. In the frequency domain, the lower sideband is brought to zero frequency.

In AM radio and in the stroboscope, the creation of sidebands is a requirement, however in many cases the sidebands are something we would rather not have. In analogue tape recorders (remember those?) it is very difficult to produce tapes in which the density of the magnetic coating is perfectly uniform and it is also difficult to get absolutely uniform contact with the heads. The result is that the gain of an analogue recorder varies dynamically. A reduction in coating density or head contact pressure will cause the amplitude to decrease. These amplitude modulations interact with the audio waveform to produce sidebands above and below every frequency in the audio signal. The phenomenon is called modulation noise.

Modulation noise still occurs in digital magnetic recording, but the effect isn't large enough to change the value of the numbers recorded and so it has no effect on the sound quality. Modulation noise also occurs in loudspeakers. The force applied to the diaphragm is given by the product of the coil current and the flux density. The flux density should be absolutely constant. In practice it isn't because it is modulated by the audio signal and the result is the generation of sidebands.

In order to generate a force on the coil, the lines of flux in the magnetic circuit must distort so that their distribution is asymmetric. This results in the flux trying to move with respect to the magnetic circuit of the speaker, which is how the Newtonian reaction to accelerating the cone is able to act on the chassis. Unfortunately the flux cannot move with respect to the pole structure in a linear fashion. Flux movement can only take place by movement of domain walls and this is fundamentally nonlinear, which is why analogue recorders need bias. Flux movement in a loudspeaker proceeds as a series of jumps which modulate the field strength which in turn modulates the audio. One of the reasons for the superb quality of electrostatic speakers is the absence of this distortion mechanism.

The best solution to flux modulation in moving coil loudspeakers is to make all parts of the magnetic circuit from electrically conductive materials. Magnetic fields find it hard to move through a conductor because massive short circuited currents are generated. At one time all loudspeaker magnets were conductive, using materials such as alnico and alcomax. The 'co' in these names indicates that they contain cobalt. When the price of this element went sky high because of politics, speaker manufacturers turned to ferrite magnets. Ferrite is an electrical insulator and cannot resist flux modulation. As a result the quality of loudspeakers actually went down, and because of the incredibly low cost of ferrite, it has largely stayed down. High-quality loudspeakers incorporate conductive magnets using elements such as Neodymium.

Digital audio works completely on sidebands. The baseband spectrum of the audio is multiplied by the sampling clock which is like the carrier in AM radio except that, being a pulse train, it contains harmonics. The upper and lower sidebands are replicas of the audio baseband above and below each harmonic of the sampling rate. The carrier spacing issue of AM radio is solved with an audio low-pass filter which prevents the sidebands of one carrier reaching more than half way to the next carrier. The same problem exists in digital audio and the solution is the same: a low-pass filter is used. This time we call it an anti-aliasing filter.

In audio compressors used for bit rate reduction, it is common to split the audio into a number of different frequency bands. After this has been done, each frequency band is then downshifted in frequency by modulating it with a carrier so that the lower sideband extends from zero upwards. The result is that, for example in a 32-band system with 32kHz sampling, all of the sub-bands occupy a spectrum of 0–500Hz and so the overall sampling rate can remain the same. □

Ad Index

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DTRS

HHB's service team pool knowledge to give a service overview of DTRS machines. Service co-ordinator **Gerry Glancy** assembles their thoughts

THE EVOLUTION OF DIGITAL multitrack recordings over the last 10–15 years has involved a wide and varied range of recorders from the more sophisticated Sony and Mitsibushi reel-to-reels to the VHS-video-based Alesis ADAT format.

In the early nineties there was a strong interest in smaller, modular digital multitracks. Tascam developed the DA-88 using the Hi-8 video format that possessed the bandwidth required for recording eight channels of digital audio on a standard Hi-8 videotape, and launched it in 1992. DTRS is a rotary head system using four heads each recording a track in turn on the tape. The success of the format was based on several factors including the ability to lock recorders together creating up to 128 tracks; relatively low cost media and hardware; solid construction; high-quality transport; remote control of up to 16 recorders; 9-pin port for video machine control; easy access for servicing; synchronisation capability; individually variable track delay; varispeed (±6%); seamless punch-in and punch-out; excellent interface accessories; and reliable electronics.

Fig.1 shows the track format for the DA-88 and how this relates to the RF waveform recorded on the tape. Sub-code data (ABS time) is recorded at the lead-in side of the RF waveform. Head-drum wear is most likely to occur on the lead-in side where the sub code is recorded (this is indicated by the reduced amplitude, A as compared to B in Fig.1). Hence, the first sign of head-wear is usually that the ABS time is lost or becomes intermittent. When ABS time is lost, it is difficult for the machine to chase and lock to external time code. When these symptoms occur, it is usually a good indication that the head needs replacing or that tracking alignment is wrong. As the head drum wears further, the audio data will be affected.

DA-88s are more susceptible to losing sub-code than DAT machines because there is only one sub-code region per RF waveform, whereas in DAT there are two—one on each side of the RF waveform.

The DA-88 and Sony PCM800 are essentially the same recorder. They share the same servo loading and transport mechanism they have the same head assembly record and playback boards and DSP. The part numbers are different for the Sony and Tascam parts, but they are completely interchangeable. The only difference besides the cosmetic appearance of the front panel is that the PCM800 has the AES-EBU interface not the Tascam TDIF card. The DA-88/98 and DA-38 share the same servo, load and transport mechanism, however DA-98 and DA-38 have an enclosed head.

More recently Tascam has introduced the DA-78HR and DA-98HR (High Resolution). These recorders offer 24-bit recording and replay. The DA-98HR in addition is capable of sampling at 96kHz, 24-bit in 4-track mode or 192kHz sampling 24-bit on two tracks. These higher sampling and word bits increase the data density on the tape. This means that the data read and write processes are far more susceptible to data errors induced by a dirty head drum, poor alignment and tape imperfections.

DTRS machines should be serviced every 500 hours. Over a period of time there will be a build up of tape dust on the head, guides, rollers and pinch wheel and so on. These deposits of dirt and dust in turn will effect the mechanical performance and therefore the recorded data will gradually suffer from higher error rates being recorded. Interchange ability of tapes and recorders becomes more difficult as the recorder ages, variations in tape tension and tape stretch all have an effect on tracking accuracy. Problems often arise in postproduction when the tape is further degraded by multiple passes and problems can occur in chase mode before audio drop-outs start to occur. If the original recording is made on a machine that is in good condition and well maintained then the chances of a bad recording are reduced. You cannot improve a bad recording and because there is no confidence monitoring on the DA-88 often the first time you realise there is a problem is in front of your client.

There have been two versions of the software for the DA-88 servo and four for the system control, the changes up to v3 on Syscon were made fairly quickly after the DA-88 was introduced.

The current software is Servo 2.02 and Syscon v4.00. Some recorders will still be running older versions of Servo and Syscon software and these can be upgraded free of charge to Syscon v3, but v4 is chargeable. The original Servo 1.3 only needs to be upgraded to v2.02 if there are problems. From Syscon 2.00 on product serial numbered 200001 upwards various error messages and improvements were added.

There have been several upgrades on the Syscon software from the first production model, v2.01, to the latest version, v4.00. The main upgrades from v2 to v3 were: Multiple formatting for locked DA-88s; enter rehearsal mode or auto I-O from any mode; punch point verification; auto play-rewind-play loop and peak hold on-off in the level meter.

The main differences between v3 and v4 are that changes can be made in the menu display and that you no longer have to use the dip switches. You can indicate the version of the Syscon microcomputer by pressing the STOP key, PLAY key, and RECORD key simultaneously, and then pressing the POWER switch.

To indicate the version of the Servo microcomputer: while pressing the REWIND key, F-FORWARD key and STOP key simultaneously, press the POWER switch.

To indicate the drum's accumulated time: while pressing the STOP key and PLAY key simultaneously, press the POWER switch.

The recorder should as far as possible be kept in a dust and smoke-free environment. The DA-88 uses a fan that pulls air in through the right-hand side of the recorder and across the tape transport and exhausts on the left-hand side of the unit. When there is a

Code	Meaning	Action
Error 1	Mechanical defect, general tape thread or path problem	Clear tape path, guides, drum, incomplete
Error 2	Incorrect drum speed servo not synchronised to tape speed or drum not rotating	Defective drum motor, drive or cleaning required, service required
Error 3	Errors 1&2, sometimes indicates tape wrapped around the head	See 1&2 above
Error 4	Capstan problem, not coming up to speed within 500 ms	Capstan motor spin undetected
Error 5	Errors 1&4, possibly tape wrapped around capstan/pinch roller	See errors 1&4
Error 6	Errors 2&4, general servo defect- neither drum nor capstan operating at correct speed	See 2&4
Error 7	Errors1,2&4	See errors 1,2&4
Error 8	Reel problems, reel move pulses not detected within time window	One or both reels not turning parts of mech require replacing
Error 9	Errors 1&8	See errors 1&8
Error 10	See errors 2&8	See errors 2&8
Error 11*	See errors 1,2&8	Mech not working unable to load or unload. Most often caused by damaged logic cam left-replace it and sector gear assembly
Errors 12,13,14,15	12 See errors 4,8 13 See errors 1,4,8 14 See errors 2,4,8 15 See errors 1,2,4,8	
Error 21*	Tape is not wound into cassette even when eject is pressed	
Error 31*	Tape slack in FFWD/REW and STOP operated	Dust accumulation on chassis? Replace slide reel cam with new type. This reduces mechanical load on slide reel cam?
Error 41	Solenoid not engaging correctly, recorder will not go into 2nd stage of FFWD/REW	Bent slide lever assembly Replace faulty part
Error 59	Reel motor not rotating or reel servo not locked within 1.5s after eject command	
Error 68	Reel motor not rotating or reel servo not locked within 1.5s after play/shuttle/wind commands	

Table 1

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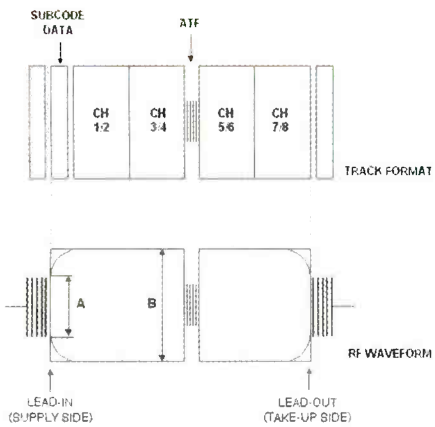

Fig. 1: relating track format and RF waveform

tape loaded in the mech then an air current also moves through the loading door and exhausts on the left-hand side. Hence any dust in the atmosphere is recycled through the tape path and transport mechanism.

Always take care when loading a tape into the recorder. Let the mech accept and load the tape never push the tape or exert pressure as you are likely to damage the loading mech and probably end up with the tape jammed in the mech.

Tape cleaners should be avoided if possible, they only remove light surface dust, but not tape dust that has stuck to the surface. HHB recommends that if the customer experiences high error light flashes and degraded audio then the recorder should be returned to a service centre for examination and cleaning by qualified personnel. If it is necessary to use a cleaning tape then we would recommend that it is only used once and only as an emergency not as a matter of routine. Cleaning tapes are abrasive and cause wear on the head every time that they are used. The more they are used the shorter the life of the head. Every time a cleaning operation is performed it is logged and can be checked .The hours of head use are also automatically clocked. Dirt and dust are the cause of most of the mechanical problems in DTRS and DAT recorders.

To use a cleaning tape proceed as follows: Eject the tape in the recorder and turn the DA-88 off. Power up the DA-88 while holding the Up and Down arrows, this puts the DA-88 into the cleaning mode. Next insert HC-88 tape and the cleaning process will start and stop automatically. Upon completion the tape will be automatically ejected and the DA-88 will be ready to use.

It is vitally important after servicing a DA-88 recorder to ensure that that RF amp is correctly aligned for metal particle and metal evaporated tape, there are different adjustments for each type of tape.

This is checked by putting all tracks into record and formatting both kinds of tape and then playing back the tapes ensuring that there is no red light for high errors

It is essential that DTRS tapes are used with these recorders as they require a tape formulation that will withstand the harsh conditions that are demanded of the medium for example continuous fast forward and rewind sequences in postproduction and multiple track laying.

There are two types of tape used in DTRS recorders and these are for different uses. They are metal particle tape and metal evaporated tape. The metal particle tape is thicker and stronger this makes it suitable for repeated editing cycles. Metal evaporated tape is suitable for one pass recording and for work that requires scrubbing, however it generally reduces head wear.

Due to the nature of the work in film and video post-production most major services may require some or all of the following parts to be replaced: cam logic and gear assembly sector which are always replaced together, pinch roller assembly, slide reel cam, reel tables supply S and T take up are again replaced at the same time and finally the head drum assembly. Head failure per 100 recorders serviced is approximately 25%

The most common faults are error 1, 2, 8, 11, 31 and 41.

This is only a guide to where a service engineer might find the cause of a problem. It is NOT a 100% guarantee that it is the exact description of the fault. (See Table 1.)

The DA-88 is a robustly built recorder with a good record for electronic reliability. The recorder rarely chews tapes, but early models were prone to having loading and unloading problems caused by the logic cam that were cured in later production models.

HHB's main area of input to the DA-88 has been the 9-pin RS422 interface. In the early days the sync boards 9-pin interface were rather basic and we helped in bringing them up to the level required for professional postproduction work. As important, perhaps, as the recorder itself is the way that the DA-88 and its accessories come together as a system. The SY88 chase synchroniser board slots easily into place and the RC848 remote control carries the 9-pin port that can drive video machines and interfacing to other digital audio equipment is easily achieved. It is advisable when using multi-recorder setups that all the recorders have compatible software including the RC848 and SY88.

If you are looking to buy a secondhand DA-88 a good test would be to use a known good recorded tape and put the recorder through a chase-lock sequence. This would give a good indication as to the state of the head. If this is not possible, then ensure that ABS time is not intermittent or audio drop-outs are not present. □

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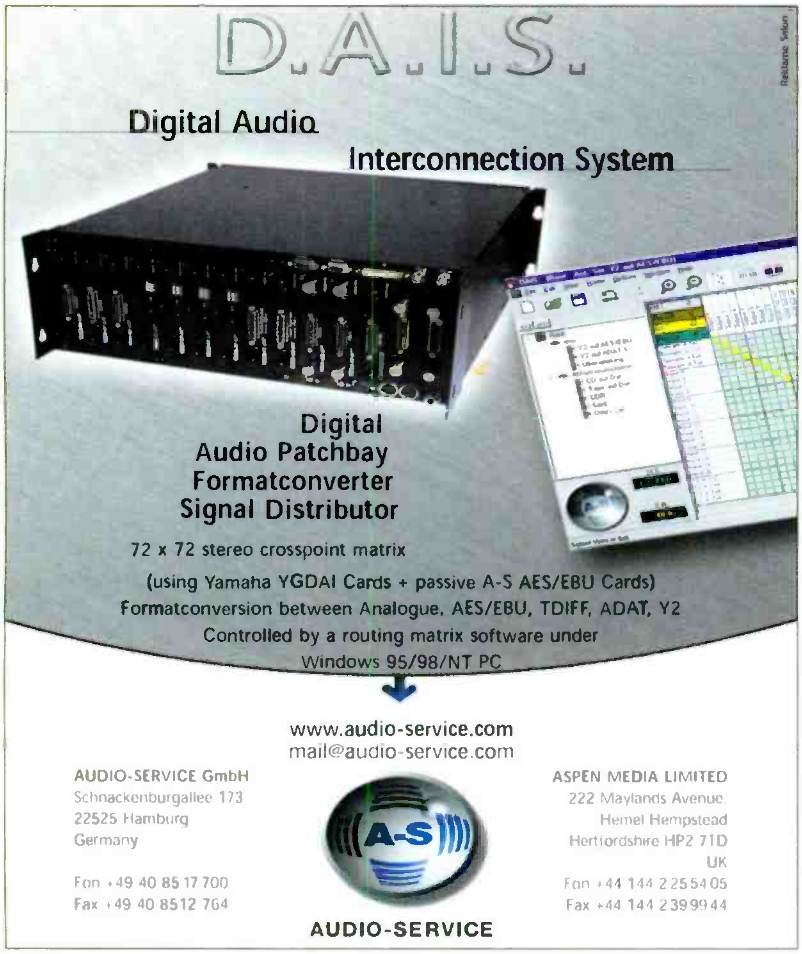

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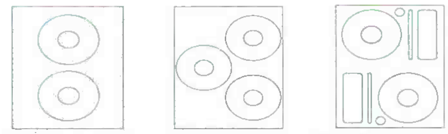

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Contact: Overseas Exhibition Services
Tel: +44 (0)20 7862 2090.
Email: philippines@montnet.com

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

10–13
Plasa 2000
Earls Court, London, UK.
Contact: PLASA
Tel: +44 (0) 8831 7607
Fax: +44 (0) 8891 6994
Email: jparmenter@golleyslater.co.uk

16–20
Cinec 2000
MOC Events Centre, Munich-Freimann, Germany.
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Email: info@messe-muenchen.de
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22–25
109th AES
Los Angeles Convention Centre, Los Angeles, California, US.
Contact: Chris Plunkett.
Tel: +1 212 661 8528.
Email: 109th_exhibits@aes.org
Net: www.aes.org

October

17–18
Broadcast India 2000
Centrum Centre 1, World Trade Centre, Mumbai, India.
Contact: Kavita Meer, Saicom Trade Fairs and Exhibitions.
Tel: +91 22 215 1396
Email: saicom@bom2.vsnl.net.in
Net: www.saicom.com/broadcastindia.

November

7–10
Satis
Paris, France.

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

ABTT Theatre Shows Points New Direction

IN A STRATEGIC MOVE to further promote and position the ABTT Theatre Show, the Association of British Theatre Technicians committee has appointed Point Promotions, organisers of the Sound Broadcasting Equipment Show, to manage its show operations, promotion and logistics. This comes at a time when the physical size and importance of the Show has surpassed the ABTT committee with the organiser, Roger Fox conceding that it is now "very much a full time job" and it can only meet the demands of the industry with professional exhibition management. In its 20-year history the ABTT show has become an industry favourite for theatrical service and equipment providers, being able to meet friends and customers in a convenient and pleasant location in the Royal Horticultural Halls, Central London.

The show will be significantly bigger next year with the newly refurbished Old Hall now being used to increase the number of stands and to provide improved visitor facilities. The original show location, the New Hall, will be dedicated to backstage equipment and services. "We are delighted to work with the ABTT. It is very much in line with our SBES operations, albeit to a completely different sector of the market" stated Dave McVittie, Partner of Point Promotions. 'We see great potential for this show which is constructed specifically for the theatre services market and, like the SBES, it is an exceptionally well-targeted event.'

The ABTT Theatre Show 2001 will be held at the Royal Horticultural Halls, London on the 4th and 5th April 2001. For further information, stand availability and general information please call 01398 323700 or fax 01398 323780.

8–10
Replitech
Hong Kong, China.

15–17
36th Inter BEE 2000
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Fax: +81 3 5402 7605.
Email: bee@jesa.or.jp
Net: http://bee.jesa.or.jp/

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Net: www.ioa.org.uk

24–27
21st Tonmeistertagung VDT International Audio Convention
Hannover Congress Centre, Hannover, Germany.
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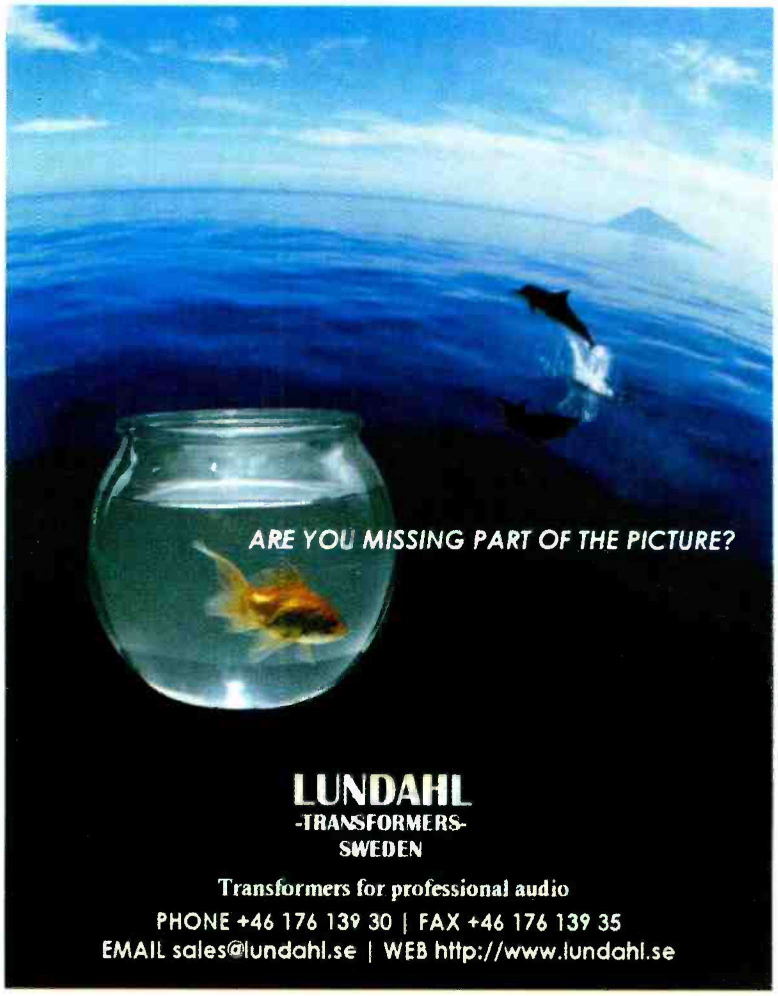

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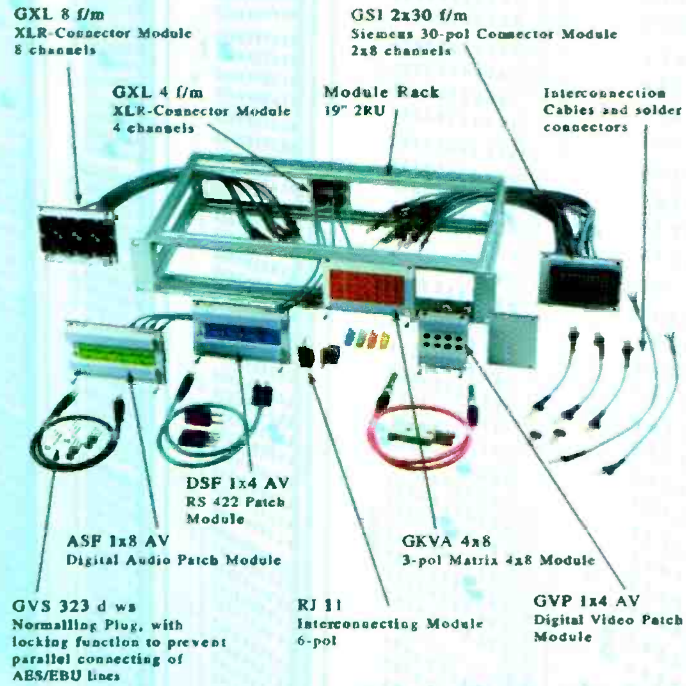

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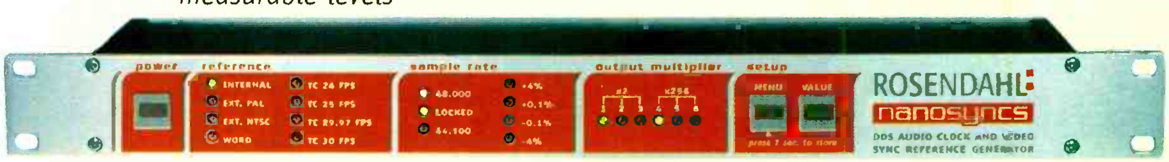

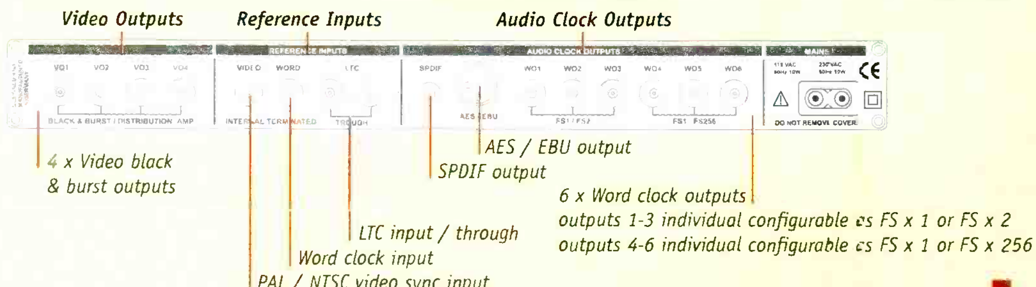

ROSENDAHL

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< Continued from page 122

be relevant to the song that I am working on, so I give it a shot.'

Radio simulator with radio compression

'In other words, checking the mix through your car radio. At the old A&M Studios in Hollywood (which I think is now *The Muppet Show* headquarters) they had a system set up so that you could send your mix through a selected frequency—with all of the relevant limiting—to your car radio. That was a lot of fun and quite helpful. I think I would like one of those.

A very comfortable control room chair, as well as two uncomfortable chairs for anyone else.

'A good chair is a must for mixing and the long hours that come with it. If not, the back is knackered by the end of the day. A Brookstone massage chair would be especially nice. At the same time, the two uncomfortable chairs should serve to keep everyone else away.'

Table tennis table

'A good game of table tennis is an important way to get a break for a moment! It gets the muscles moving after sitting over the desk for hours at a time.'

Cappucino machine

'Good coffee is a valuable mix tool. Administer when required.'

Installation

'I would have to rely on expert advice, as I am not particularly technically minded (and I am quite happy that way).

THE BALANCE SHEET

Total expenditure: < £5m

Well, Tim hasn't gone mad in terms of the equipment, so he should be able to secure a decent commercial property overlooking the Ocean. And, who knows, possibly he'll even have enough left to buy a one-room 1930s condo in Pacific Palisades.

Historical outburst

AFTER GOING THROUGH your most interesting *Studio Sound* special—Millennium Edition (December 1999)—I became sorely upset at not finding any coverage of my past mentor and dear friend, Dr Marvin Camras, who passed away June 1995.

Dr Camras' development breakthrough patent was to inject a high-frequency AC bias signal—which was modulated by audio record signals—into his special patent-designed, magnetic-record-play heads. Dr Poulsen's patent in the late 1800s on magnetic wire recording had high noise, high distortion,and a low signal level, making it unusable. Dr Camras' highest achievement was developing magnetic (wire-tape) audio recordings to produce 40dB signal-to-noise ratio, low distortion (3%), and wide frequency range 50Htz to 5kHz (±3dB) during 1940–50.

In 1945 I went to Illinois Institute of Technology, called IIT. There I was privileged to work with Dr Camras on his inventions. During this interim, he became my mentor and best friend. After obtaining patents on circuitry innovations using high-frequency AC bias in magnetic audio recording, and advanced magnetic head design—he began work on formulating experimental 'magnetic oxide' slurries to deposit onto paper and plastic tape ribbons, wound on a reel. This opened up the way for his magnetic record-play 'tape' patents, and also patents on magnetic record-play tape heads. It was at this point that he started experimenting with Multi Multi Magnetic Tape Audio Record-Play, systems, upon which he obtained patents. This was in the late 1950s

I'm sorely upset and surprised that Dr Camras' breakthrough patents on magnetic tape audio recording were not mentioned in the *Studio Sound* Millennium Anthology on audio achievements, from 1917 to 2000.

Enclosed is a complete biography and genealogy on Dr Marvin Camras—his breakthrough magnetic recording patents & acknowledgments.

I think it would be 'ethical and upright' —for *Studio Sound* to print some explanation for the oversight, together with excerpts from enclosed Biography and Genealogy Master Index depicting Marvin Camras' magnetic tape recording inventions & acknowledgments (dating from 1945 to his demise in 1995). Perhaps it could be in the next issue of *Studio Sound*.

Philip D Pavda, New Jersey, US.
vidious.pal@juno.com

Tim Goodyer replies

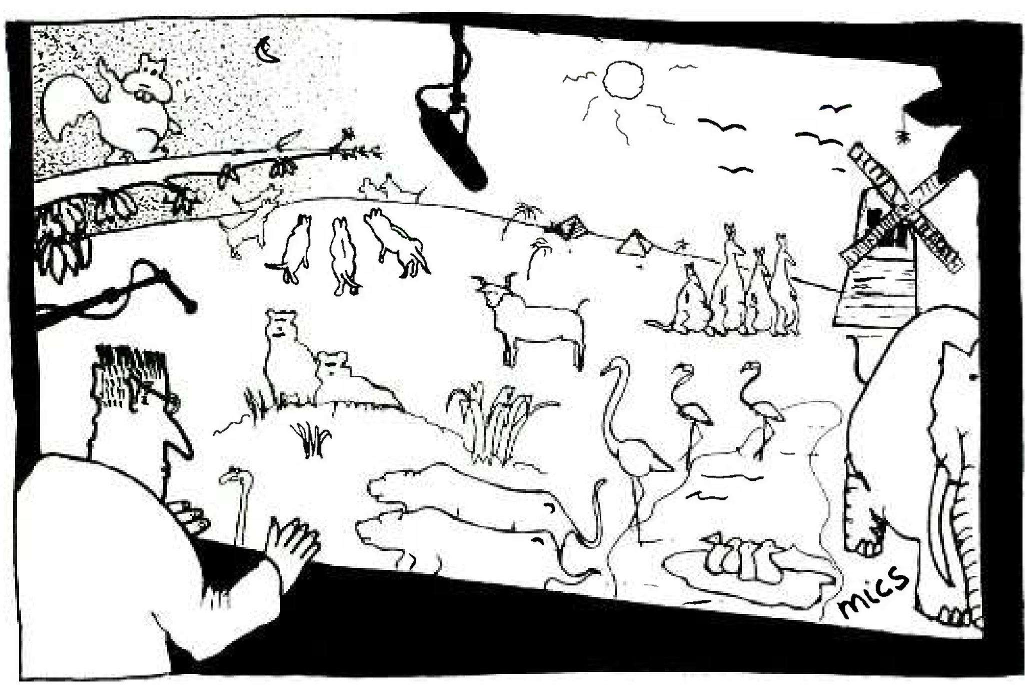
Your comments, and Dr Camras' work, are duly noted.

Photographic memory

OUTSTANDING! I commend your magazine for the excellent tribute to Richard Swettenham. It was the picture that really brought back the memory. I haven't seen him since 1976 if I remember right, when I had the chance to spend the best part of a day with him in Chicago. I was chief engineer for Brunswick Records at the time. The name rang a bell, but that picture, wow, like a flashback. What a wonderful and charismatic fellow he was. Thank you again for the memory and excellent tribute.

Robert G. Kachur, Charlotte, Michigan, US.

ROCK 'N' ROLL ANIMAS



Cedric marvelled at the rich, vibrant detail of the studio 2 eco system

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Dansak on the Ceiling
Tears on my Pilao
Living Dahl
Tikka Chance on Me
Paperback Raita
Korma Chameleon
Vindaloo Sunset
Mulligatawny of Kintyre
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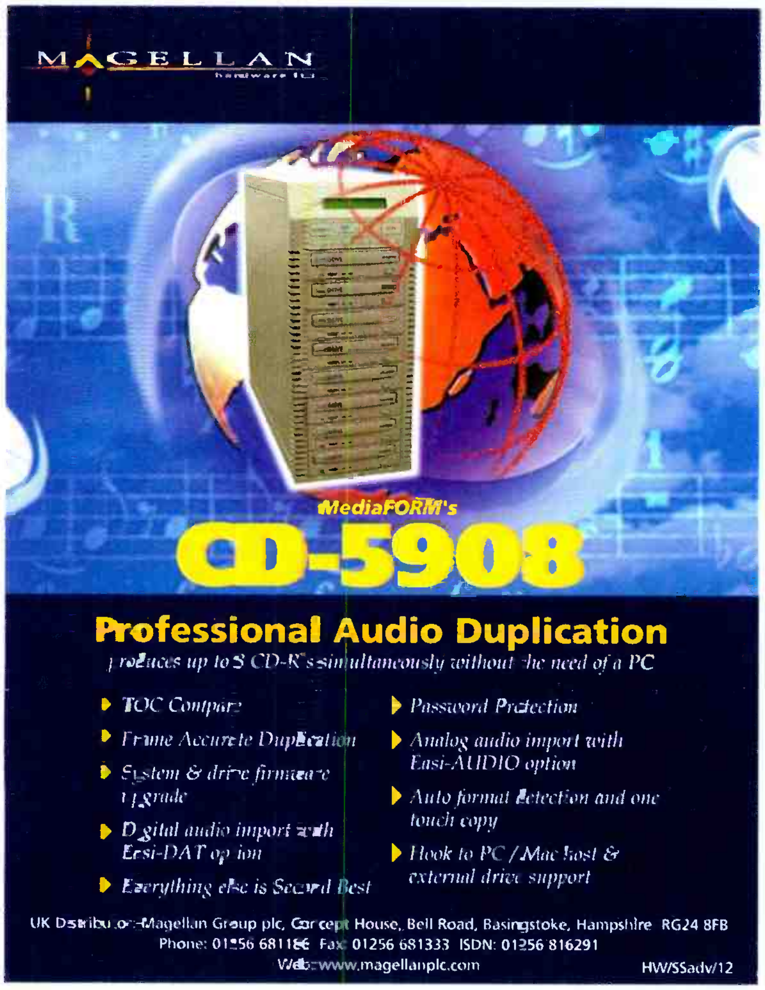

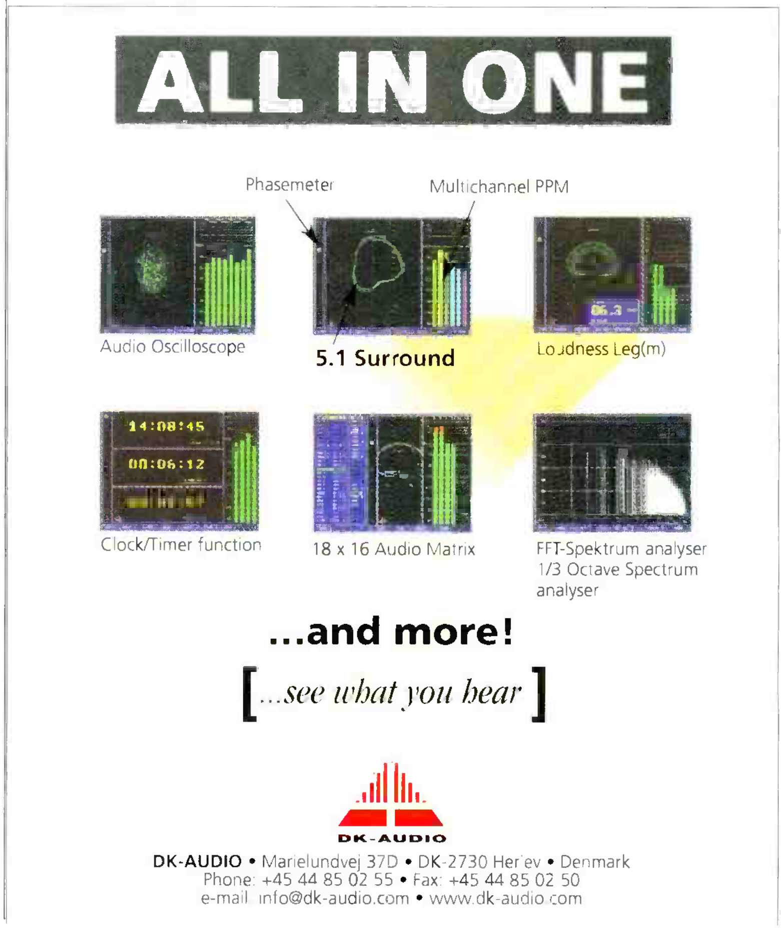

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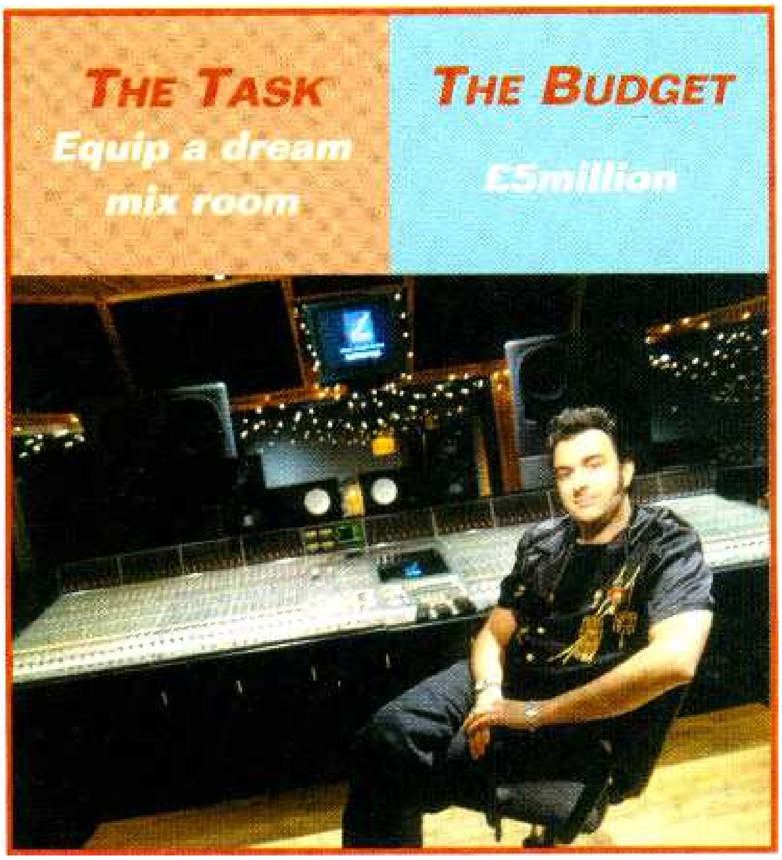
TIM PALMER'S MIX ROOM

Richard Buskin introduces a new series in which we offer a producer or engineer a (sometimes) generous sum of money with which to equip a given task



PRODUCER, ENGINEER AND REMIXER Tim Palmer started his career as an assistant engineer at London's Utopia Studios during the early 1980s. Here he worked with artists such as Mark Knopfler and Dead or Alive before becoming a fully-fledged producer on projects from the likes of Robert Plant, The Mission, The Mighty Lemon Drops, House of Love, Texas, Tin Machine, Tears For Fears, Gene Loves Jezebel and Ned's Atomic Dustbin. During the early 1990s Palmer made another transition, devoting most of his professional time to mixing and remixing —Pearl Jam, Catherine Wheel, Concrete Blonde, The Cure, James, Michael Hutchence, The Dance Hall Crashers, Reel Big Fish, Soak and Sepultura are among his credits in this latter genre.

Now relocated to Los Angeles, it is unsurprising that for his dream mixing facility he would like a Malibu, California location, with a control room that overlooks the Pacific Ocean. 'It would be quite inspiring to work in an environment like that,' he says with a tendency towards understatement. 'In a dream world I would live very close to the studio, although the house prices in that area might be a little out of my reach at this moment in time.'

Well, £5m is a lot of money to spend, Tim. However, before you go and blow most of it on a highly desirable but totally unnecessary beachside property, you would be best advised to invest your huge windfall in some state-of-the-art mix room technology.

Console:
SSL 9000J

'In the last few years I have been lucky enough to use the SSL J-Series, and in my opinion it is the best sounding and most creative console I have ever worked on. In the past, people seemed to favour the sound of the Neve and the ease of operation of the SSL. Well, with this console you have both. I love the automated panning that is available to every channel, and the automated switching makes life very simple.

'When I was mixing Pearl Jam's first album, *Ten*, at Ridge Farm, I would often split the vocal four or five times to get the variation in vocal sounds that I needed. Now, with all of the automated sends and inserts, I am usually down to just the one.'

Monitors:
Genelec 1031A;Yamaha NS10

'The monitors that I mix mainly on are the Genelec 1031As. Nowadays I find that I rarely use the big monitors in a control room. I have a listen to check what's going on in the very low end, but apart from that I work mainly on the close-field 1031A. Any speaker is just a personal point of reference, and this is mine. With this speaker you can hear what's going on in the high frequencies right through to the lows. The NS10 is also a valuable tool, and it is a great speaker for balance. You get less distracted by fidelity and simply ask yourself, "Does the mix feel good?".'

Editor:
ProTools 24 Mix-Plus with plug-ins and massive hard drives

'The Pro Tools has improved my ability immensely to be creative at the mix stage. I use the Mix-Plus system—alongside the format of the music I am mixing on—as a kind of slave, and I use Pro Tools to try new arrangement ideas, add new parts, affect guitars and vocals with plug-ins, and generally move stuff about. The Amp farm plug-in is great for vocal distortions. I used this a lot under the vocals on U2's song, 'The Ground Beneath Her Feet'. Thanks to this system everything is done very quickly and with ease, and it is of course all non-destructive.'

Tape machines:
Studer A800; Ampex half-inch

'Studer multitracks are like tanks; the best machines ever. Can you believe they don't make multitrack recorders anymore? There again, when I print the mix I love the sound of the Ampex half-inch tape machines.

Outboard:
GML stereo EQ (2); rack of API EQs; rack of Focusrite EQs; teletronix LA2A (2); Distresser; DMX stereo harmoniser; Eventide H3000 (2); Drawmer gates (20); Lexicon PCM42 (8); Lexicon 480 reverb; Lexicon 224XL reverb; AMS RMX reverb; dbx 902 de-esser (4);

'I have many favourite pieces of outboard equipment. The Teletronix LA2A compressors are great for a transparent, solid sound. On the other hand, when I want to hear the compressor working I sometimes like to use the Distresser, although the compressors on the SSL J are usually sufficient for most of the processing I need in a normal mix situation.

'As far as extra EQ, not a lot is required really as the SSL EQ is so good, and of course on the J-series you can choose between two different EQ curves. I often use the GML stereo EQ over the mix bus, and it is a unit that I think sounds really good. Some EQ sounds so angular and unnatural, whereas I find the GML to be smooth and—dare I say it—very musical. In addition, I really like the API EQs, which I fell in love with when I was producing bands like Texas and The Mission at RAK Studios in London.

DAT Machine and CD Burner

'A DAT player is an obvious must-have. When mixing, I still print the "master" version of the song to half inch, but all variations I print to DAT only. In fact, some mixers don't even bother with half inch at all these days, so a good DAT machine is important.

'CD is likely to be the medium on which the band and A&R listen to your mix, so it is worth taking time to get a good CD burner and put plenty of level on it. It seems to me that some members of our industry just nudge a mix according to how loud it is. They cannot wait until it is correctly mastered, whereas I have never listened to a CD and not been able to reach over and set the volume to my desired level. Some mixers actually master their CDs before sending them out, and this prevents their work from being judged on this ridiculous premise.

Big TV Monitor

'I like a big TV in the control room, with the sound fed through the monitors. I take short breaks to clear my head, and also, often when I switch the TV to MTV, M2 or a movie, I may hear a sound or effect that could

Continued on page 120 >

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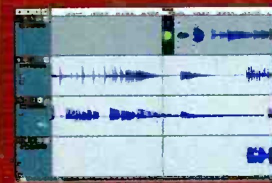

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