Studio
sound

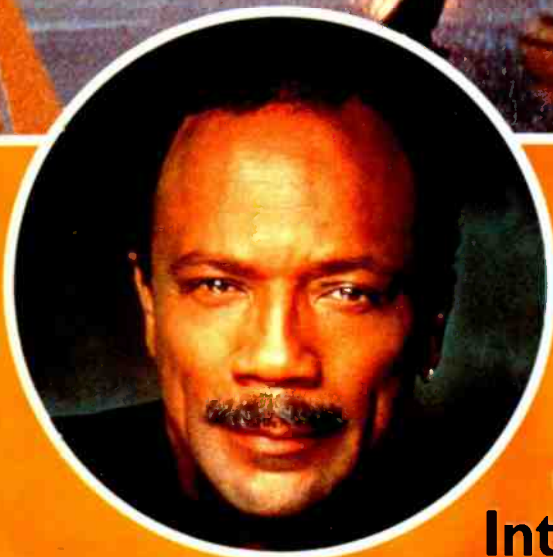**THE INTERNATIONAL PROFESSIONAL AUDIO MAGAZINE
FOR RECORDING, POSTPRODUCTION AND BROADCAST**

LA Audio DigEQ
Sonic Foundry Soft Encode
Liquid Audio Liquifier Pro
Allen & Heath GS3000
Digidesign ProControl
Focusrite Platinums
Quested VS2108
Lexicon PCM 91
tc MasterX
Studer V-Eight
Alesis M20

GODZILLA
Live in Manhattan

**The
QUINCY
JONES
Interview**

A QUESTION OF DESKTOP AUDIO
META SOUND'S D&R OCTAGON
RADIO MICS ON THE HOOF
WORLD CUP FINALS

9 770144 594031   07

Studio Sound July 1998

12

66

73

76

Soft fixtures and wire

A CONVERSATION that will always make me sit up and listen is that old chestnut of what constitutes a professional product. While tedious in the extreme, some still feel moved to pontificate upon the subject with pseudophilosophical gravity, and therein lies the sport. To understand you must first dismiss, for reasons of clarity, the precise meaning of the word 'professional' because you are entering the realm of those who see the world in monochromatic simplicity, or who are playing the provocative mock-intellectual.

The former camp likes to see XLRs, transformers, and a hefty price on its boxes, and will sing to you the you-get-what-you-pay-for ballad. The latter will try to shock you with the revelation that the vocals on the last Goat Throttle single were recorded on a portastudio in Trap 3 of the gentleman's toilets in Camden Town—Traps 1 and 2 were used as drum and brass isos respectively. And did you know that it is the fastest selling single since Bell End's *Chunder like Thunder* which itself was recorded in a large trunk? Conclusion: anything can be deemed professional if it is used in the creation of commercial product.

By extrapolation we are quickly within a gnat's whisker of discussing the professional merits of bedroom soft-fixtures and wire. But the attitudes get to you. You are forced to use that old flanger box even though you know that the atrocious background noise will be a real problem. Your hand is stayed from screwing the o200 dual compressor into the rack by someone who asks why you are so prepared to risk the integrity of your entire signal path with such a cheap unit. Pro is a matter of attitude, and it's why you can do a perfectly adequate 'to a standard' job with budget gear and foul up big time in Studio A. Acknowledging that there are differences between items of gear is a major achievement because it will allow you to make decisions about what you intend to use from an enlightened and experienced standpoint. It is all about playing to the strengths of what you use and not what you use it on. Cassette can still cut it in radio reporter acquisition, but finds few friends in purist classical recordings; although, faced with the challenge, there are those who would produce markedly superior results with it than the rest of us. Attitude is the key, and it should never be so narrow that it precludes you from working.

Zenon Schoepe, executive editor

Standard Issue

IT WOULD BE EASY to lend weight to what follows by dropping the names of the industry heavies who have sought to discuss it with me. And weight is what is required, given what is at stake. The problem with name-dropping, however, is that we too readily tend to respond to the personalities rather than the issues they proffer, so instead, I will open by pointing out that the April issue of the *Journal of the Audio Engineering Society* devotes a considerable amount of space to surround-sound-related matters.

Not that the names in the *JAES* are lightweight, of course, but the papers confine themselves to technical issues and back up their positions with the usual fare of AES papers—loquacious English, arcane equations, sweeping graphics, copious references… It is the essential stuff of industry foundation, but it stops short of championing issues—and surround is an issue of epic proportions for the pro-audio industry.

Although my own recollections of the seventies quadraphony fiasco are restricted to touring hi-fi shows with adolescent awe, it is this precedent that forms one of the tenets of the New Surround Activists' agenda. Technologically inferior to today's systems, the SQ, QS, and even CD4 formats exposed many of surround's fundamental obstacles (from acoustic to domestic), ultimately failing through lack of consumer acceptance. Since then we have been through the so-called format wars waged between early domestic VTR systems, the introduction of CD, and the chaos that surrounded MiniDisc and DCC. Further issues have arisen through interim advances in technology—compatibility between the Dolby Stereo carried by so many films, and the audio-compression systems being lined up to deliver them, for example. Today, the climate appears more ready to favour surround's success than it did 25 to 30 years ago, but only if we have learned the lessons offered.

What is desperately needed is serious and open discussion—and ultimately agreement—on a variety of key issues. Potentially, surround sound represents unprecedented opportunities for pro-audio. But unless we organise ourselves it will be a missed opportunity—at best.

Tim Goodyer, editor

Avid alliance

Avid Technology has formed a strategic alliance with Microsoft through its acquisition of Softimage. Avid's nonlinear video and audio systems will benefit from the association of Softimage's 3D and 2D animation and compositing software systems.

The move is accompanied by the appointment of Softimage founder, Daniel Langlois, to Avid's board of directors. Softimage will cost Avid US$285m in consideration of stock options held by Microsoft and Softimage employees.

Microsoft will receive $79m in cash, a subordinate note in the amount of $5m, Avid common stock valued at $93m and a warrant valued at $32m. The Avid common stock in the transition will consist of approximately 2.344 million shares valued at $39.71 per share. The warrant, that has a life of ten years and is exercisable beginning in the third year, will permit Microsoft to purchase 1.155 million shares of Avid common stock at $47.65 per share.

On completion, Microsoft will own approximately 9.1% of Avid outstanding stock, excluding the unexercised warrant. Microsoft has agreed not to purchase any additional Avid common stock for a period of five years and to maintain its investment for a minimum of three years.

Additionally, Avid will issue options valued at $76m to purchase 1.912 million shares of Avid common stock at a nominal price of $0.1 per share, to Softimage employees in consideration of their unvested Microsoft stock options which will be forfeited in the transaction.
Net: www.avid.com

▲ Austria: Vienna's famous Opera House has been renovated and includes a new sound system based on the JBL HLA, Venue and Sound Power sound reinforcement systems. The venue will accommodate over 2,000 people and was rebuilt following bombing during WWII to reopen in 1955. The new renovation involved the use of room modelling using JBL's CAD P2 simulation system and has resulted in the use of a 'surround type' installation involving 12 VS3218 Venue cabinets, four HLA 4895 cabinets and a pair of Sound Power SP128 LF cabinets. Extensive use has also been made of DSC280 digital controllers.

▼ US: Sci-fi epic *Deep Impact* involves a newsroom scene designed around a Telex intercom setup. The film, which charts the events of a comet on a collision course with planet Earth, involves an MSNBC news reporter breaking the story and requires that the newsroom scenes be convincing—especially to those in the know. Telex, US. Tel: +1 612 884 4051
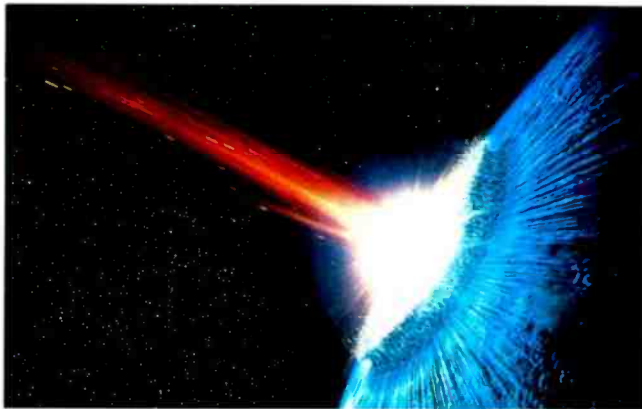
On-line copyright

US: An American initiative to protect the interests of copyright holders in the Internet has been formed by an 'ad hoc coalition of high technology, telecommunications and creative industries' whose aim is to urge the US Congress to approve protective legislation.

The WIPO Copyright Treaties Implementation Act (HR 2281) is intended to allow the implementation of two international treaties dating from 1996 and is claimed not to 'fundamentally change' US copyright law, but should prohibit business and activities regarded by bits supporters as designed to undermine technologies protecting on-line material. It would also clarify legal responsibilities on the Net and make service providers more readily accountable for piracy conducted over their networks.

Among those forming the alliance are AT&T, Bell Atlantic, the Motion Picture Association of America, the Recording Industry of America, Sony Pictures, Time Warner, Walt Disney Company, Twentieth Century Fox and Universal Studios.
RIAA, US. Tel: +1 202 775 0101

▲ Italy: The D&R Cinemix console at Rome's Cine-AudioEffects facility (above) was recently involved in the awarding of a special prize at the Cannes Film Festival. Four similar consoles have been taken by Rome's post facilities following the Amsterdam AES Convention—two will go to Fono Roma for use on Dolby Digital dubbing, a 64-input Cinemix console will go into foreign language specialist dubber Sound Art, and Studio Sound Design will take a 32-input Cinemix for Dolby Surround work, notably for RAI.
D&R, Netherlands.
Tel: +31 294 418014

◀ Japan: The live hall of Japanese national TV station NHK has seen a custom 70-input Midas XL3 console installed. With both facilities and sound quality being quoted by NHK's Productive Engineering Centre staff, the most immediate aspect of the customisation is the console's ability to be broken down into five smaller sections (as demonstrated by Midas' David Cooper, left) enabling a single sound engineer to handle small productions without sacrificing its ability to handle massive events.
EVI, UK.
Tel: +44 1562 741515

UK: Two industry figures have died unexpectedly in recent months—co-founder of The Professional Monitor Company, Adrian Loader, and Sarner International's Peter Sarner. The unrelated events saw Loader pass away on 17th May after a short illness and Sarner die suddenly on his 50th birthday, 10th April. Loader was well-known for his work on loudspeaker design having begun in the industry with the BBC and spent time with FWO Bauch before establishing PMC with Pete Thomas. Sarner made his name in Britain's audio-visual visitor attractions industry in which he had been active for some 30 years.

▼ **UK:** London's Imperial War Museum is to archive more than 27,000 hours of audio recordings of sounds and interviews documenting events from the Boer War to more recent events in the former Yugoslavia. With the collection expanding at more than 1,000 per year, the task was becoming progressively more urgent when the Museum opted for the CD-R medium and chose three Marantz CD620 CD recorders for the job. The new CD archive will replace the duplicated ¼-inch tape archive. **Marantz, The Netherlands. Tel: +31 40 273 2241**
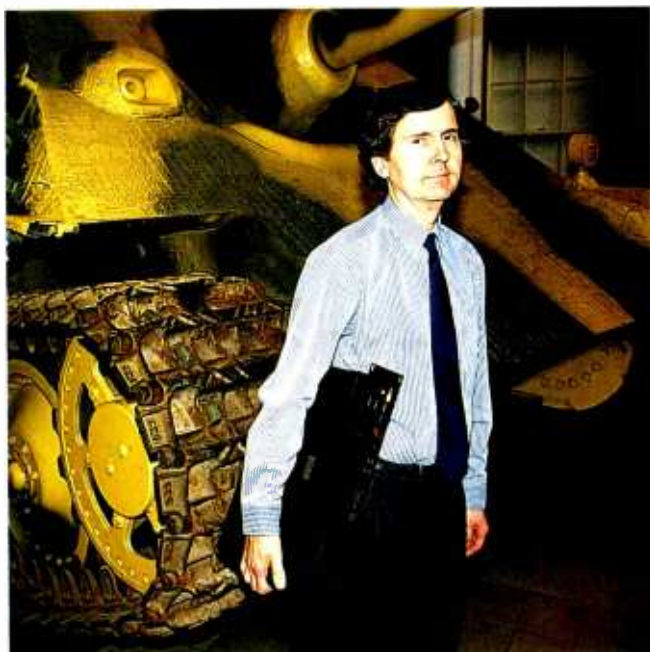
Tape awards

US: Both Quantegy and Emtec have been handing out recording awards recently.

Quantegy's Golden Reel was presented to the husband and wife partnership of jazz guitarist Larry Carlton and singer Michelle Pillar Carlton for use of Quantegy tape in both recording and production successes. The evening at Nashville's Loews Vanderbilt Plaza was furthered by a performance from Michelle Pillar Carlton in front of a selection of Nashville's many producers and artists.

Emtec's Master Award was presented to producer Tommy LiPuma, engineer Al Schmitt and New York's Avatar Studios for their parts in jazz singer Diana Krall's No.1 album, *Love Scenes*, and the choice of BASF tape.

Both media manufacturers were also sponsors of the first Audio Masters Golf Tournament held in Nashville to benefit the AES Nashville Section Engineers Relief Fund, along with the likes of Audio Technica, Studer, Otari, Amek and a number of recording studios. The Tournament was won by the team from Nashville's Starstruck Studios lead by David Malloy.

◄ **Australia:** A new Australian touring production the musical *Grease* will make particular use of the automation of a pair of Soundcraft console for its demanding monitoring setup. Key cast members will use in-ear monitoring while the musicians use headphones, the mixes for all of which are delivered by Series Five and SM24 consoles. All inputs arrive at the Series Five from which the automated scene muting is used to enable required monitor mixes and disable those associated with scene changes. **Soundcraft, UK. Tel: +44 1707 665000**

■ British independent broadcaster, Granada Television, has replaced mag film with an 80-track dubbing system based around ten DAR OMR8 MO recorders. The system will be accessible from both of Granada's dubbing theatres with transfer and copying duties taking up any spare capacity. BBC Television, meanwhile, is to build a new dubbing suite at Elstree Studios, based around a 24-fader, 64-input Soundtracs Virtua console and Akai DD1500 hard disk recorder. The facility will be dedicated to dubbing of the EastEnders soap. West Country Television is also to use an 8-fader, 64-input Virtua with a DD1500 for regional programme production. **DAR, UK. Tel: +44 1372 742848. Soundtracs, UK. Tel: +44 181 388 5000. Akai, UK. Tel +44 181 897 6388.**

■ Nashville's Seventeen Grand Recording has opened a second room based around a Euphonix CS3000 desk. The 104-fader console is 5.1-surround capable and is part of an expansion programme that involves new control, live and overdub-lounge rooms designed by Seventeen's Jake Niceley and Michael Cronin Acoustic Construction. Studio 2 complements the Neve VR60-equipped Studio 1. Elsewhere in Nashville's recording community, Al Schmitt has bought a Studer D19 MicValve preamp-convertor. **Seventeen Grand, US. Tel: +1 615 327 9040. Euphonix, US. Tel: +1 818 766 1666. Studer, US. Tel: +1 615 391 3399/+1 415 326 7030.**

■ The Japanese studios of Tokyo Broadcasting System Inc have recently been refurbished to include two Amek 9098 analogue consoles. Occupying two of the four studios at TBS Tokyo Midoriyama complex, the 48-input consoles will be used primarily for spoken word and FX recording to D2 digital VTR. **TBS, Japan. Tel: +81 3 3224 2925. Amek, UK. Tel: +44 161 834 6747.**

■ Canadian film and TV post facility PFA Medallion has bought a second Lafont Panoramix film console identical to the 68-input desk installed last year. The new console is at the heart of a new room currently under construction, and will be equipped wit Flying Faders, 1,024-point TT patchbay permitting it to process 212 simultaneous audio channels. **Lafont Audio Labs, France. Tel: +33 1 3473 6539.**

■ Malaysian concert hall, Dewan Filharmonik Petronas, will open in August with a 48-fader AMS Neve Capricorn console. Served by a custom video monitoring and comms system, the desk will be linked to the hall's stage from its Abbey Road Studios-ADG designed studio. Down the peninsula, Singapore's new Green Room music and post facility has opened with a Euphonix CS2000 console. Otari Radar hard-disk recorder, Studer Dyaxis II and Genelec 1031 monitoring. **AMS Neve, UK. Tel: +44 1282 457011. Genelec, Finland. Tel: +358 17 813311.**

■ British music mecca, Real World Studios, has bought a second Sony Oxford digital console, making it the first European studio to do so. John Oram has taken his first UK order for the digitally-controlled analogue Series-48 desk, that is bound for a new custom recording facility in East London. Elsewhere in London, Chiswick Reach studio has been shopping for Lexicon, Focusrite and tc electronic outboard, KRK monitors and an HHB CD recorder, while Abbey Road has taken two further Total-systems SPA2s for its interactive and DVD suites. **Real World, UK. Tel: +44 225 743188. Chiswick Reach, UK. Tel: +44 181 995 2647. Abbey Road, UK.**

Tel: +44 171 266 7000. Sony Broadcast, Europe. Tel: +44 1932 816000. Oram Professional Audio. Tel: +44 1474 815 300.

■ New York's NMT is to be the first customer for SSL's new Axiom-MT digital console. National Mobile Television is to install the 96-channel, 48-fader desk in its all-digital, 5.1 capable HDTV mobile slated for service at Madison Square Garden including extensive sports programming. **NMT, US. Tel: +1 206 242 0642. SSL, UK. Tel: +44 1865 842300.**

■ China, Tibet, Burma, Vietnam, North Korea, Laos and Cambodia may all benefit from Radio Free Asia's investment in 15 further Orban Audicy radio workstations. The total of 36 v2 Audicys spread around 5 Asian Bureaux and Washington DC are networked to serve the 85 staff delivering news and information broadcasts in seven languages. **Orban, US. Tel: +1 510 351 3500.**

■ San Francisco's Smooth Jazz radio station has favoured Microtech Gefell's UM92.1S microphone for its new studio. Eight UM92s have been purchased for use in all on-air and production applications. **G Prime, US. Tel: +1 212 581 8938.**

■ London post facility TSI has chosen Fairlight's MFX3 Plus workstation and FAME production system for its expanding long-form post work. TSI serves major UK TV networks including the BBC, and is looking to new cable and satellite networks for its future. London's M2 post house has purchased 5 MFX3 Plus workstations and a FAME replacing SSL systems at its existing Camden facility and to equip a new White Mark-designed £3m Soho complex. Elsewhere in London, SD Post has purchased a 32-fader Soundtracs DPC-II digital desk for a new 5.1 surround room. Back in Soho, 2 160-channel, 64-fader DPC-IIs are to be installed at Magmasters Studios for TV and feature film work, which recently included Godzilla. **Magmasters, UK. Tel: +44 171 437 8273. Fairlight, Europe. Tel: +44 171 267 3323. White Mark, UK. Tel: +44 7071 222543.**

■ The London Festival Orchestra is to build a recording studio as part of the redevelopment of its South Bank HQ. The facility will include a Mackie 32:8:2 desk with Ultramix automation serving a Tascam DA-88/DA-38 recording setup and Mac-based Cubase Score with Audio VST. **London Festival Orchestra, UK. Tel: +44 171 928 9251. Mackie Designs, US. Tel: +1 206 487 4333. Tascam, UK. Tel: +44 1923 819630.**

■ California-based Skywalker Sound has purchased 2 Stage Tec Nexus audio routing systems, one for Stage A to provide 48 tracks of busing and playback, and 8-speaker formatting, the second for Stage B. Both stages use AMS Neve Capricorn consoles and the routers will be linked via fibre optics to allow complete interconnection between the facilities. **Stage Tec, Germany. Tel: +49 951 97 22525**

■ Hong Kong's Avon Recording Studios has joined Taiwan's Formosa TV in installing SSL 400-series consoles. Avon's 4048 G+ joins an SL9000 and sits in a Toyashima control room. It is intended to broaden the appeal of the facility to the HK and Chinese markets. Currently nearing completion, Taipei-based Formosa is intended to serve a variety of sound for TV roles including orchestral recording. A further 4040 G+ has been ordered by Sydney-based Australian Institute of Music for its Q Recording Studio. **Avon, Hong Kong. Tel: +852 2332 1620. SSL, UK. Tel: +44 1865 842300.**

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July

9
Biz Tech 98
Loew's Vanderbilt Hotel, Nashville, Tennessee, US.
Contact: SPARS National Office.
Tel: +1 800 771 7727.
Email: spars@spars.com

14
AES Lecture:
Controlled Subjective Assessments of 2–5 channel Surround Sound Processing
by Francis Rumsey, University of Surrey. The Conference Room, Baden Powell House, South Kensington, London, SW7, UK.
Contact: AES.
Tel: +44 1628 663725.
Fax: +44 1628 667002.
Email: aesuk@aol.com

14
Lecture:
Sennheiser Radio Mic Seminar
by Malcolm Johnson, Orchard Audio. Bruntsfield Hotel, Edinburgh, UK.
Contact: Rosie Field, Sennheiser UK.
Tel: +44 11494 551 531.

15
Lecture:
Sennheiser Radio Mic Seminar
by Malcolm Johnson, Orchard Audio. The Liverpool Institure for the Performing Arts, Liverpool, UK
Contact: Rosie Field, Sennheiser UK.
Tel: +44 11494 551 531.

17–19
Pro Audio and Light
World Trade Centre. Singapore.
Contact: IIR Exhibitions.
Tel: +65 227 0688.
Fax: +65 227 0913.
Email: ruohyi@iirx.com.sg
Net: www.mediav.com.sg/pala

22
Lecture:
Sennheiser Radio Mic Seminar
by Malcolm Johnson, Orchard Audio. Holiday Inn, Kensington, London, UK
Contact: Rosie Field, Sennheiser UK.
Tel: +44 11494 551 531.

August

11
AES Lecture:
Design of a 3-way Loudspeaker system for Cinema Sound
by Bill Webb, Martin Audio. The Conference Room, Baden Powell House, South Kensington, London, SW7, UK.
Contact: AES.
Tel: +44 1628 663725.
Fax: +44 1628 667002.
Email: aesuk@aol.com
Net: www.aes.org

September

11–15
IBC 1998
RAI Exhibition and conference centre, Amsterdam, The Netherlands
Tel: +44 171 240 3839.
Fax: +44 171 240 3724.
Email: show@ibc.org.uk
Net: www.ibc.org.uk/ibc/

16–19
Duplitech 98
Duplication and Replication Equipment & Technology for Magnetic and Optical Media
Beijing International Conference Centre, Beijing, China.
Contact: China Exhibitions
Tel: +44 171 546 8775.
Fax: +44 171 546 8570.
Email: nfo@chinex.com

19–21
2nd Cinec 98
2nd Internationmal Trade Fair for Motion Picture Technology and Postproduction
München, Germany
Contact: MesseMünchen Gmbh
Tel: +89 9 49 01.
Fax: +89 9 49 09.
Email: info@messe-muenchen.de
Net: www.cinec.de

26–29
105th AES Convention
Moscone Convention Centre, San Francisco, California, US.
Tel: +1 415 558 0200.
Fax: +1 415 558 0144.
Email: 105th-chairman@aes.org
Net: www.aes.org

October 12
--November 6
ITU Plenipotentiary Conference
Minneapolis, Minnesota, US.
Tel: +41 22 730 5969.

October

20–24
Tele-, Kino-, Radio Technologies
Exhibition Centre, Sokolnik. Moscow, Russia.
Contact: Ekaterina Zotova
Email: main@admt.ru

November

4–5
23rd Sound Broadcasting Equipment Show (SBES)
Hall 19, National Exhibition Centre, Birmingham, UK.
Tel: +44 1491 838575.
Fax: +44 1491 832575.
Email: dmcv@pointproms.co.uk
Net: www.i-way.co.uk/~dmcv/sbes.htm

4–8
News World 98
Fira Palace Hotel, Barcelona, Spain.
Contact: News World Ltd
Tel: +44 171 491 0880.
Fax: +44 171 491 0990.
Net: www.newsworld.co.uk

3–6
SATIS 98
Paris-Expo,-Porte de Versailles, Paris, France.
Contact: Alexandra Tholance Conseil.
Tel: +33 1 45 45 65 25.
Fax: +33 1 45 45 65 35.

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

December

9–11
5th Broadcast Cable & Satellite India 98
6th Comms India 98
Pragati Maidan, New Delhi, India.
Contact: The Federation of the Electronics Irdustry
Tel: +44 171 331 2000.
Fax: +44 171 331 2040.
Email: feedback@ fei.org.uk
Net: www.exhibitionsindia.com

9-11
Cable & Satellite Asia 98
Singapore International Convention & Exhibition Centre, Level 4, Suntec City, Singapore.
Contact: Reed Exhibitions
Tel: +65 434 3675.
Fax: +65 334 4119.
Email: chuilan.chia@reedexpo.com.sg

June 99

10–15
21st Montruex International Television Symposium and Technical Exhibition
Montruex, Switzerland.
Contact: Ms Patricia Savioz.
Tel: +41 21 963 32 20.
Fax: +41 21 963 88 51.
Email: message@symposia.ch
Net:wwwmontreux.ch/symposia

October 99

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

Proximity alarm

MAY I TAKE very gentle and respectful issue on one small point John Watkinson makes in his 'Microphone Principles' article (*Studio Sound*, May 1998, page 93). He says that 'a cardioid will have 6dB less proximity effect...' Basically I'm sure this is true, however, some years ago I got quite fascinated by proximity effect and went to quite a lot of trouble doing the maths and realised that the important thing is the effective delay (or path length, if you like) between diaphragm front and the rear via the phase shift network. The longer this is, the less the proximity effect.

There are two contrasting practical examples: one is the (in my view) seriously underrated Calrec CC01 (I think) cardioid which has a very marked proximity effect, as one might expect from its construction. The other is the AKG D202, which has a very long path length, because the rear apertures are a long way back, and the proximity in this is quite minimal. In fact, if I remember correctly, this was made a point of in the advertising.

Enough of this carping. Please keep up with your splendid work. I hope the many-headed, dwelling in the outer darkness, read and absorb what you write. The cynic in me feels that they are beyond redemption, but I may be wrong, and perhaps you have evidence to the contrary.
Michael Talbot-Smith, Worcs, UK.

Ribbon rules

AT THE RISK of showing my age (too late to worry, I suspect), I must comment on the point about ribbon microphones (*Studio Sound*, June 1998, page 53, last paragraph) in which Malcolm Addey ridicules the rule about not using ribbon mics horizontally.

Of course, modern ribbons are too small to sag, but the original BBC AXB and AXBT microphones had ribbons about 5-inch in length, and we were told at the Beeb not to use them horizontally because the ribbon sag was enough to be detectable. The ribbon didn't exactly 'pop out of the pole pieces', but if it wasn't centralised the output could be affected. We were also told that a slight tilt away from the sound source could be usefully utilised to reduce sibilance problems with some announcers, as the ribbon was so long that HF cancellation occurred along its length. (We didn't have EQ in those days!).
John L Andrews, Marketing Director, Solid State Logic, UK

Frankly speaking

I'VE JUST BEEN READING Philip Newell's piece on speaker testing (*Studio Sound*, April 1998). Phil starts well, goes on to argue himself into a corner and then, accepting that there is no sensible way to test a speaker anyway, tells us that he is going to continue the series with measurements 'that can then be correlated with the subjective opinions of the loudspeaker users'. Huh?

I'm sure that the industry will continue to try to bring about an ecumenical situation between the slide rule brigade (yes, I'm getting old) and us psychoacousticians, but in the final analysis they will continue to fail. A rubbish loudspeaker will show up on the impulse response tests as awful, but the same set of tests will never reveal the subtlety of a first class studio monitor. Of course Michael Gerzon and Richard Heyser were right about the impossibility of true fidelity, but that's simplistic; they both had much more influential things to say than that!

In a funny sort of reverse way your 'Open Mic' page is on the same sort of wavelength. Todd Wells is crying in his beer about how different and difficult digital engineering is from our old analogue systems. The world has moved on Todd!

I'm surrounded by digital goodies produced for PCs in the Joemeek Studio; and they all work, with remarkably few bugs, less in fact than we used to achieve in the early days of analogue mixers of similar complexity. (I hasten to add that they are not necessarily 'good' because of that!) We're even fitting digital backends to Joemeek compressors without the forecast doom that Todd expects.

I'm enjoying the world of audio now more than ever before. We all have better equipment, we can all make better sounds and recordings and we (theoretically) have more time to enjoy them. Let's admit a diversity between theoreticians and artists and enjoy the music.
Ted Fletcher, Joemeek, UK

Off track

I REGRET TO INFORM YOU that Richard Buskin's statement that multichannel audio did not exist until the digital formats is horribly off the mark. Four-track magnetic sound was quite common in the better theatres in the late fifties, and that's not even counting the multitrack systems that were part of more specialised formats like Vista-Vision or Cinemascope, or the special system developed for *Fantasia*.

What is worse is that I am sorry to say that some of the older formats sounded better than the some of the newer digital formats. But they died a slow and painful death due to the high cost of operating them as television killed off much of the theatres in the sixties and seventies.
Scott Dorsey, Kludge Audio, Williamsburg, VA, US.

Alesis M20 and Studer V-Eight

The ADAT format forges ahead with the release of two flagship Type II machines from two leading pro-audio lights. **Dave Foister** compares and contrasts, and explores the options

NO DOUBT ABOUT IT, Alesis is bidding for the big time. Having dominated the home-project end of the MDM market for so long (so much so that the word ADAT is to digital 8-track as Hoover is to vacuum cleaners—how many times have you heard someone refer to a Tascam ADAT?) the company is moving determinedly up-market with attacks on several fronts. The first was word length, with the XT20 (*Studio Sound*, June 98) making Alesis the first to give us 20 bits on such a format, and now we have the facilities contender: the M20 goes all out to provide all the features and functions known to professional man, in a no-compromise attempt to move on from the home studio image.

Everything about the new machine points to this, from the size to the weight, and even a casual glance at either the front or the back shows how much is on offer. Facilities which used to need additional bits, like the BRC, are clearly accessible directly, and it is evident that the M20 is much more complete and self-contained than any ADAT before. This is not a souped-up semi-pro machine, but a full-blooded industrial recorder; there's hardly a jack socket in sight, but a panel full of XLRs to gladden the heart.

This is the first striking thing on the back. The unbalanced -10dB connectors have gone, replaced by balanced +4dB ins and outs, alongside the familiar EDAC multipole—good news for those who want to slot one of these into an existing rack in place of an XT, or even an older ADAT. Those with an eye for symmetry will also quickly spot extra connectors, as the M20 has a dedicated time code track in its format, and a brand new feature giving an auxiliary analogue track, recorded linearly, to give something recognisable as audio when scrubbing. Both of these are balanced in and out on XLRs.

The front panel is a clear departure from its predecessors in having two display screens surrounded by more buttons than an ADAT has seen before. One screen is dedicated to the various formats of time display and global settings, while the other concentrates on metering and other track-specific information. The displays are the focal points for a control surface that completely abandons the austerity of the early machines; here we have a wealth of dedicated buttons, an alphanumeric keypad, and a jog-shuttle wheel, all pointing to the comprehensive nature of the machine's capabilities.

Before exploring them, it is worth making it clear that the M20 still retains at its core the familiar ADAT format, and is compatible with other machines in the same way that the XT20 is. It has full 20-bit recording alongside the older 16-bit format, and can effectively be slotted into an existing stack of machines as a direct replacement. It would appear, in fact, that with an M20 as master, all its built-in sync facilities control the whole set up, although more basic machines may struggle to keep up with the transport.

The old ADAT transport came in for a lot of stick once other faster machines came on the scene, and was one of the main improvements on the XT20. It is, after all, asking a lot to shift the mass of tape in an S-VHS E180 as quickly as that in an 8mm cassette, particularly when the thing records at three times normal speed and has therefore got three times as far to go as even a conventional VTR. The M20 goes further still with a heavy-duty, precision-controlled, direct-drive transport that does a much better job of keeping up with the demands of the studio without frus-

Alesis' M20

trating delays. Wind speed is more than twice as fast as an XT, and parking is more direct and precise—just what is needed when chasing code. The whole transport is remarkably free of mechanical noises apart from a smooth whir, and despite the speed the impression is that the tape is in fact being treated much more gently than before. The intelligence with which the machine searches is clearly improved, with virtually no overshoot and constantly changing speed control. It manages smooth transitions between Wind and Play—without stopping first—of a kind normally expected only on high-end, open-reel machines.

This fast slick tape handling is part of what the M20 is all about, but the other is its built-in sophistication; anything you have ever been able to do with an ADAT via the Big Remote Control can now be done directly on the machine. Thus there are 100 locator memories; all can be named, and there is a list of useful labels—Verse 1, Chorus 2, Bridge and so on—which can be used to save time, and even customised. The keypad provides both numeric and character entry, using the same system as the BRC where each key carries a number and three letters—much faster than data entry wheels or

The M20 has its own new remote, the CADI, able to deal with the additional facilities it offers over the XT20, but Studer has again decided to go one better with the Cockpit

nudge buttons once you get used to it. Comprehensive signal routeing is available directly using the meter screen and the track buttons; this allows input sources to be a mixture of analogue and digital, inputs routed to other tracks, and limited sources to be sensibly duplicated across tracks so the machine can be used with 4-bus or even stereo mixers. Copying tracks is set up in much the same way, and there is also access to the individual track delays. All this and more forms a data file about the project in hand which can be recorded in the Data area at the head of the tape and loaded back in next time, restoring all the locations, names and detailed settings —one of the biggest practical pluses of ADAT to my mind.

But the M20 goes considerably further than just squeezing the BRC's functionality into the recorder—useful as it is, manipulating the little buttons on an upright front panel can never be as easy as hitting the big ones on a remote control panel, so for many purposes half this stuff might never get used as there will be a remote available anyway.

An important new introduction on the M20 is the Auxiliary track, allowing a jog-shuttle wheel to be used with the same kind of precision as on an ana-

logue recorder. Scrubbing direct from digital tape is never very successful—hard drives can do it well, but tape needs some kind of RAM buffer to offer recognisable audio.

The M20 has another idea; a dedicated analogue linear track with the flexibility to record whatever you want so that it can be scrubbed directly with the wheel. It has its own input and output on the back, and will record signal from this input along with any other inputs that are selected to it. These signals are mixed on to the Aux track as well as being recorded digitally on their own tracks. During a search operation, the track appears not only at its dedicated output but also on the outputs of the tracks it was recording, giving way to the digital tracks when the machine goes into Play. An Auto function will record any armed tracks to the Aux track at any time, so that the latest version of an overdub is always available for scrubbing. The upshot is very nearly real analogue scrubbing; I say very nearly because there's still one limitation, which is the difficulty of moving the tape smoothly across the heads. The sound still tends to come out in very short bursts as the jog wheel is moved; although they are so much more recognisable than what digital normally produces, and the bursts are so short, that finding a precise position is quite easy. There seems to be some heavy-duty compression on the analogue signal so that it is always audible, however quiet the original may be; this shows up when there is nothing actually recorded on the Aux track, and the sound emerges >

Studer's V-Eight

< as a bizarre pumping hiss.

The jog-shuttle wheel is vulnerable to accidental knocking, and very light in operation, so it is just as well that its automatic activation can be turned off, forcing the use of the SEARCH button next to it. This is one of a vast menu of utilities, called up by a single editing button and clearly displayed using the unusually long text line on the screen. Here are functions for dithering the 20-bit output down to 16 bits, saving and loading the tape's data area, and adjusting a host of parameters such as drop-in crossfade times, time code output level, and the time it takes before the mechanism unthreads the tape. It is also possible here to determine whether the monitoring of an input signal comes direct through analogue ins and outs or goes via the convertors, a useful check as to the contribution of the convertors to the sound. It has to be said there is a difference, with the same slight increase in brightness via the convertors as seemed to be present on the XT20—hardly surprising as they probably use the same circuitry. It is only slight, however, and not really significant, especially compared with the excellent noise performance and overall transparency.

The recording of time code takes MDM a little further than it has been before. All time code rates are supported, and the M20 can record either incoming code or its own onboard generator. In case you were worried, the code is not recorded linearly alongside the Aux track, but digitally in among the audio and subcode in the manner of DAT. The difference from previous machines is the facility for actually recording the external code rather than
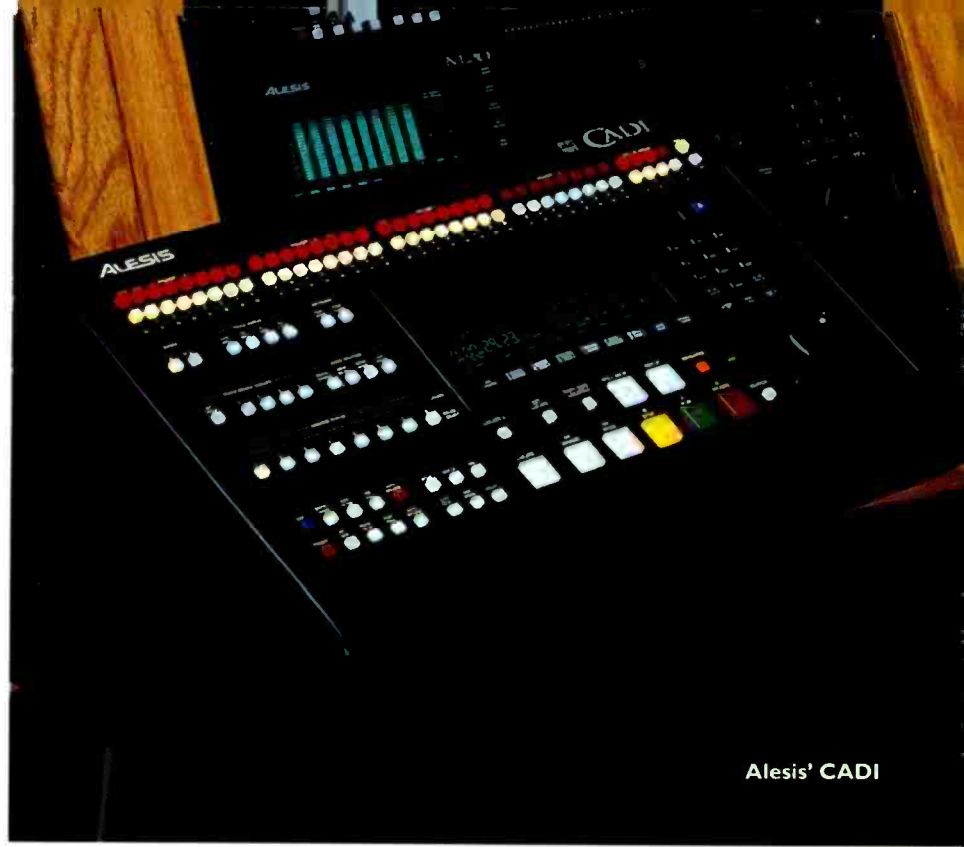

Alesis' CADI

mapping it to ADAT's internal time reference; although the ability to do this remains, so that in the absence of recorded code ADAT time can be used for chase operations with whatever offset is required.

The M20's new arrangement means that audio from elsewhere can be transferred into the M20 along with its original code and sync maintained effortlessly throughout the process. The only stipulation is that the recorded code track should be synchronous with the sample rate, and that when the machine is chasing incoming code, it too should be locked to the incoming wordclock—
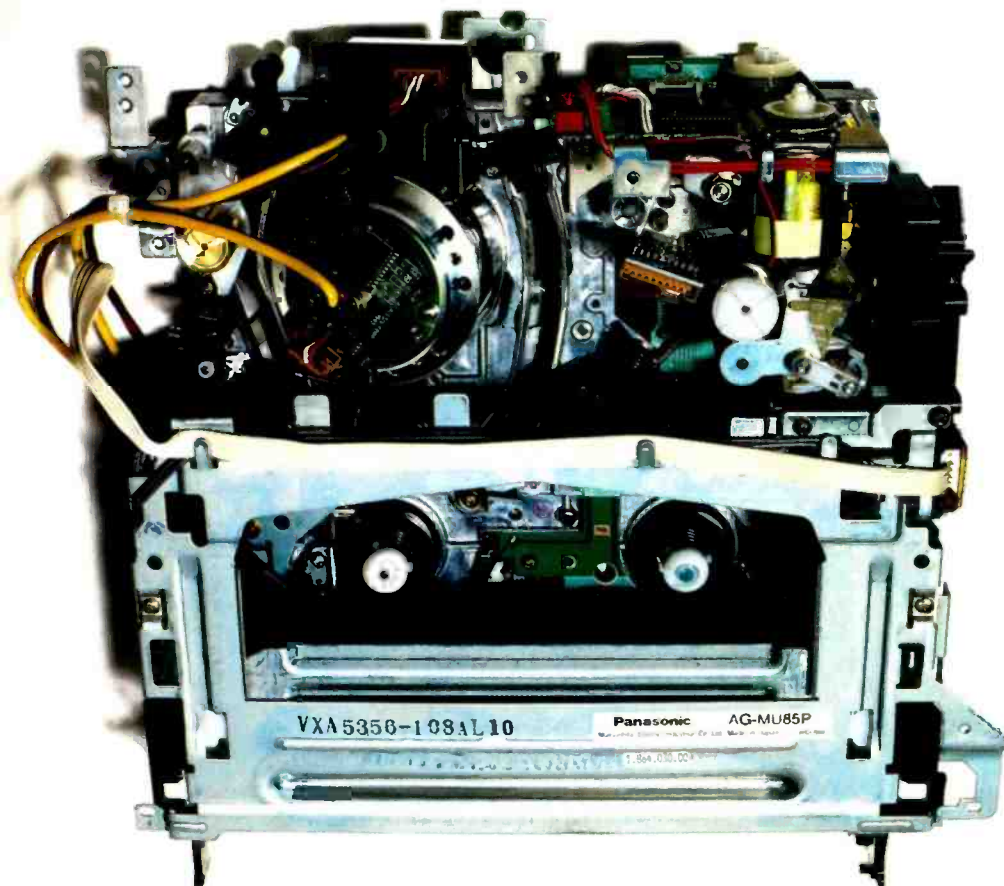
all sensible and normal if everything's set up properly. The onboard generator will, of course, always be locked to the digital reference, and besides its full palette of frame rates it also allows full control over the user bits, handy for date-stamping in the absence of an onboard real-time clock.

All of this is standard, but there is an optional upgrade: a single board with eight XLRs carrying inputs and outputs in AES-EBU format, the final touch in interfacing the M20 to the outside world in familiar pro formats. Naturally ADAT lightpipe I-Os are provided as well.

IT WON'T HAVE escaped your notice that with the new features boasted by the M20, the familiar BRC no longer has the capability to take full control of the machine. It is good not to have to get at the machine to tinker with it, partly because even with the much enlarged front panel the buttons are still small and vertical, and partly because the benefits of the quiet transport are offset by a very audible fan in the back. In an ideal world the machine(s) would not be by your elbow when mixing, so the need exists for a comprehensive remote that does the M20 justice. Just such a remote is available, of course; the CADI picks up where the BRC leaves off, with styling and layout to match the M20 and a much bigger display area, duplicating the main time and text window of the machine itself with its dedicated function selection buttons. I haven't had the opportunity to try one, but the pictures suggest a very practical control surface, and I cannot help wondering whether Alesis might consider designing a 'dumb' M20, with the new transport and the Aux and time-code facilities, but >
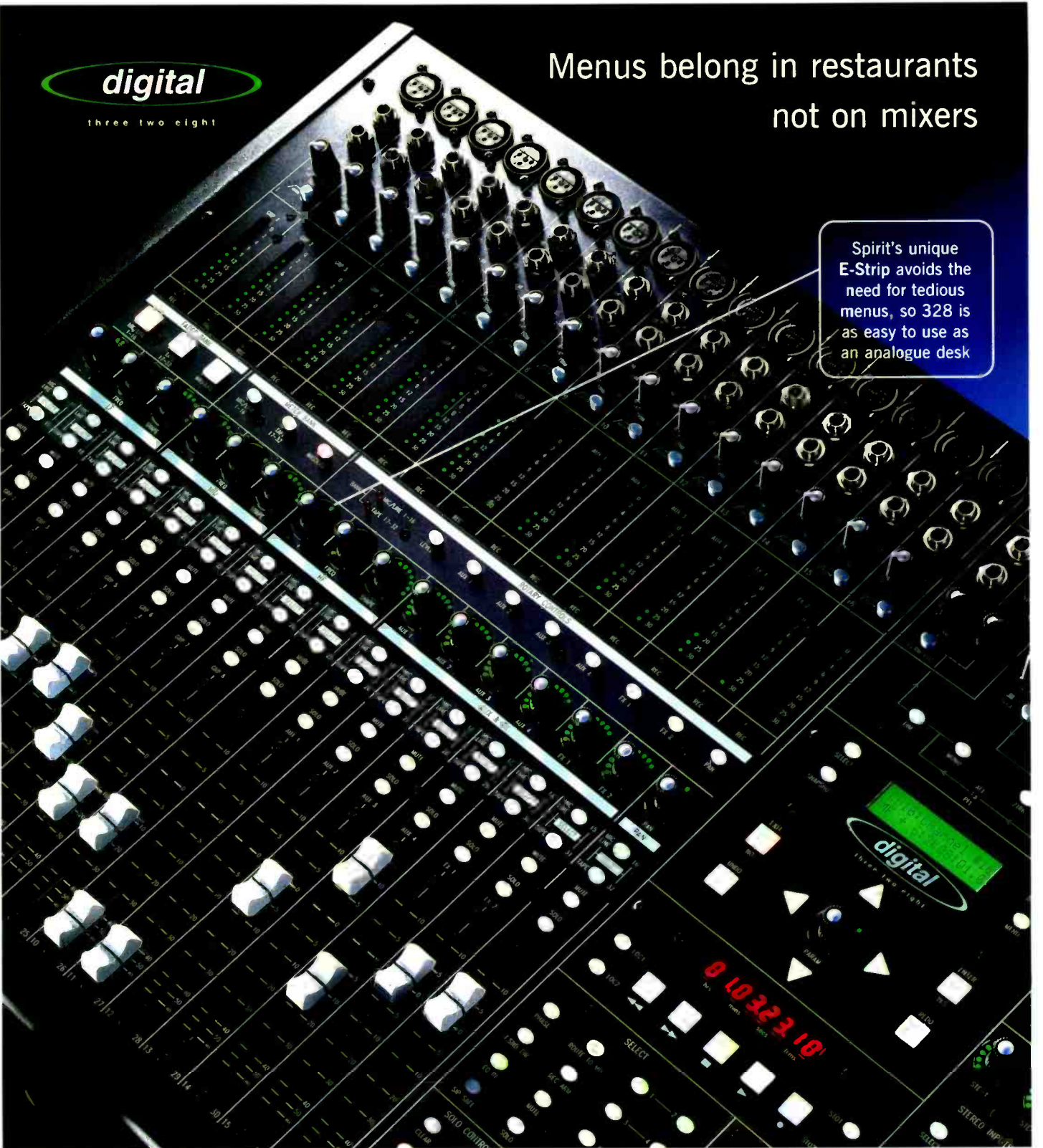

VXA5356-108AL10   Panasonic AG-MU85P

July 1998 **Studio Sound**

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< without all the expensive controls.

As it stands, there is nothing dumb about it at all. The transport alone should silence ADAT's critics, and the new features are real practical improvements that raise Alesis' game substantially. Others still need to catch up with the high bit rate, never mind the other stuff, showing once again that Alesis never stands still for long. Personally, I can take or leave the styling—it is very functional and VTR-like, and the buttons look a bit tacky and wonky here and there—but we do not buy recorders for their looks and the performance and facilities speak for themselves. It remains to be seen whether the competition is too deeply entrenched for the M20 to make a difference, or whether the major enhancements are enough to change loyalties in the target markets; it certainly looks as though Alesis has given it its best shot.

I kept feeling that if the ADAT format had appeared in this guise from the word go, even at the higher price, it might have made more headway in high-end applications than it has so far

FEW THINGS HELP a fledgling format raise its profile better than its adoption by a big third-party name. Although Alesis gained the support of Fostex a while ago, the resulting RD-8 for some reason failed to make much impact; now, however, the Rolls Royce of tape recording has stepped into the fray, lending even more credibility to the ADAT format at a time when Alesis is clearly trying to drive it up-market.

The name of Studer has an import like few others. Not many companies have produced such a high proportion of classics, industry standards and benchmarks, or managed such a happy combination of brick outhouse build-quality and sonic desirability. There can be few of us who have not used Studer tape machines and regarded them as the elite; it should say a lot for the father of MDM formats that Studer is happy to put its name on an ADAT machine. Simple rebadging is not what the V-Eight is all about, however, as Studer has re-engineered several aspects and added further features to make this very much a separate entity; although, of course, it retains full compatibility with other machines in the family. In every important respect it is a straight plug-in replacement; it shares the same sync and control interface, the same multiway analogue I–O connector, and the same increasingly familiar ADAT optical multichannel digital connections. It is in every technical sense an ADAT Type II machine; although the way it is put together means that it could only

really be a Studer.

It is immediately apparent from the front of the machine that it is not just an M20 in a Studer box, although the M20 is the comparable Alesis model upon which the V-Eight is based and with which much of the internal design is shared. The displays, transport slot and controls are all in more or less the same place, but the satin aluminium front panel, the aluminium jog-shuttle wheel and the substantial illuminated buttons all shout Studer and give the impression of the ruggedness and engineering excellence with which the Swiss company has always been associated. It does not end there, as there are obviously several features on the V-Eight which are additions to the already impressive spec of the M20. The Alesis machine brings ADAT into the full-blown professional world, but the V-Eight brings us that little extra Studer something.

Before coming to the specifics of the extras, the details of the control layout are noteworthy in many respects. The front panel is a single heavy aluminium sheet, as opposed to the contoured plastic of the M20, and has integral rack ears with heavy-duty handles; a dedicated set of rack slide rails is available for easy access. The M20 has a vast array of control buttons, some of which are on the cramped side and rather basic and anonymous; the V-Eight has all these functions, and more, on buttons which present rather a different image. All are labelled on the buttons themselves, all are illuminated, and all share a style and feel that is positive and reassuring. The illumination boasts the machine's campest feature: a light sensor in the front panel that dims the buttons when the ambient light level falls and brings the brightness up again under strong light. It is slick, it is helpful and it is subtle, and you get the feeling that only Studer would bother.

The transport buttons may not be as big as those on an A80, but they are substantial and positive, and difficult to miscue with. Likewise, even though you can tell the mechanism behind the shuttle wheel is the same as on the M20, the solid metal wheel itself has a weight and feel that make it somehow easier to use. A small detail here is a little light marked QUALITY; this flashes in various colours in its subtle way to alert the user to error correction activity even when the error display is not selected. It is reading the same data, but glowing in progressively more alarming colours as the number of errors increases. It is not a bad >

< idea to draw attention to deteriorating performance of either tape or machine, since as we all know digital error correction can work against us by concealing problems until they become terminal, and this light is the first thing I've seen that flags the errors up so clearly without having to look at any numbers or select a specific display.

But perhaps the most obvious difference is a row of dual-concentric knobs at the bottom left, which reveal themselves to be a simple but useful monitor mixer. Although it is only mono, this is a remarkably powerful addition to the system, partly because it extends beyond the single machine. Four of the dual controls handle the levels from the eight tracks; the fifth has a master control for the adjacent headphones output, plus a level adjustment for the rear-panel monitor linking facilities. This is

one of several differences found at the back that are not so immediately obvious, but are plentiful nonetheless. The front-panel monitor mix appears on its own dedicated balanced output, and there is also a pair of XLRs for daisy-chaining this feed through multiple machines. Effectively all the machines can be plugged up to feed a common bus which makes its final appearance on the last machine in the chain, either for headphone monitoring or for feeding elsewhere at line level.
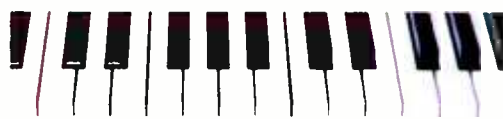
Another important addition on the V-Eight is the rows of screwdriver presets below the analogue input and output XLRs. Like the M20, the machine has only pro line level balanced analogue connections, both on XLRs and the ADAT standard EDAC, but the presence of these level adjustments suggests that the circuitry inside is different. The M20

is fixed in terms of the levels it expects to produce 0dBFS, and the levels it will produce at its output for given digital values, whereas the V-Eight can be matched to any reasonable operating level over a 20dB range. In fact Studer has swapped out the entire input and output circuitry, putting in classic Studer transformer-balanced analogue stages and its own design of 24-bit convertors both ends instead of the 20-bit on the M20. The A–Ds are sigma-delta 128x oversampling convertors directly descended from the acclaimed convertors used on the D827. The clock performance in particular has been upgraded from that in the M20 as it was felt the jitter behaviour offered scope for improvement. The resulting sound is quite stunningly good—quiet, clean, smooth and transparent, every dB the professional machine.

The same facilities for time code are presented on the V-Eight, with the subcode implementation fully compatible with the M20, and the analogue Aux track is also carried across, justifying the presence of the scrub wheel. Also common to both is the bay for the optional AES-EBU card, giving full in and out access to the machine in this common standard format as well as the ADAT fibre optic interface.

The most significant extra on the back of the V-Eight is the connector for its special remote control. The M20 has its own new remote, the CADI, able to deal with the additional facilities it offers over the XT20, but Studer has again decided to go one better with the Cockpit. Like the CADI, it incorporates the central time and function display from the machine, but like the V-Eight it reeks of Studer class and could easily be the remote for any top-flight Studer deck. Where the CADI needs its own local power supply, the Cockpit derives its power from the connected master machine, which then passes commands down the sync chain to further ADAT-compatible machines. It is lightweight and can be mounted on a stand (the same as the D827 remote) or in a rack, but at the same time is predominantly of metal construction, and very much a robust and hittable control surface. It can control up to 64 tracks' worth of ADAT, arranged in two banks of 32, duplicating all the front-panel controls of each machine and applying global control of transport and utility functions. Time code and Aux track enabling for each machine is here, along with all the locator names and track-group functions—a facility to select bundles of tracks to arm or make safe all at once, carried over from the BRC. Everything's clear, intuitive and positive, with an exemplary layout, and the arrangement of the sections follows the CADI set up less closely than the machine's front panel mimics the M20. The heavyweight wheel is exactly the same as on the

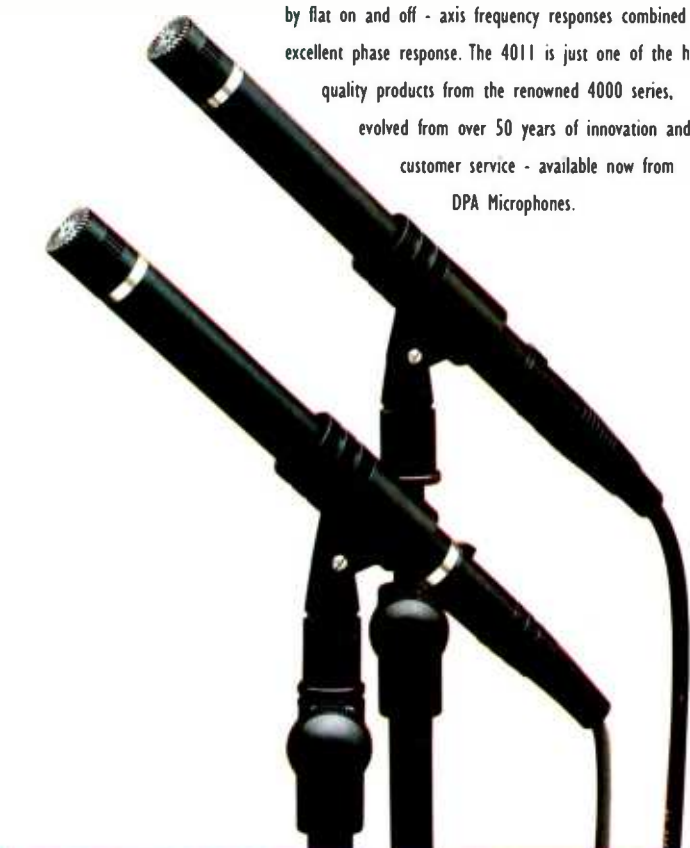# S O U N D S .

D I S T I N G U I S H E D

V-Eight, and the dimming buttons are also carried across, with their own front-panel sensor for when machine and remote are in separate rooms.

There was something in the labelling and layout of the Studer's controls, both on the machine and on the remote, that helped it become familiar very quickly, and made settling down to a session within minutes of getting the kit out of the box straightforward. This was despite the fact that the manuals for both are still in preparation, and I had neither of them, with only an M20 manual and a quick list of differences to guide me.

The Cockpit supplied to me had built into it the RLD (Remote Level Display) meterbridge, which is also available as a separate rackmount entity. Like the Cockpit, it dispenses with the onboard power supply required by the Alesis equivalent and draws its power from the connected V-Eights. It has 32 bar-graph meters, very similar to those on the machine, with the same selection of peak hold and clear facilities. Connections are provided again for 64 channels' worth of machines, so the meters are switched over in two banks. What I did miss was any calibration indication on the meters; there is some colour coding, with the bars turning from blue to green near the top and showing a red Max at full scale, but there is nothing to show what the other graduations represent. Perhaps this is a consequence of this being a preproduction version; although the brochure shows no scale markings either. The meter input connectors on the back are the same as those found on the M20, further ensuring compatibility in a multimachine system.

When not in a rack or a console, the meter bridge attaches naturally to the Cockpit to form a very compact and ergonomic control centre for the connected system. Even the angles of presentation are carefully considered to give maximum visibility from a range of operating positions.

I T SHOULD BE almost painfully obvious by now that I like the Studer set up very much indeed. I immediately felt comfortable with its operation, and its construction inspired total confidence. I have had Studers of one sort or another around me all my working life, as I am sure have most of you reading this, and the V-Eight feels and works like any of them (except that I have not yet had to replace an A101, a routine occurrence on virtually every session with one particular favourite A80). I have to say that when MDM first appeared I never expected that it could compare with what I thought of as heavy-duty professional machines; its appeal was its phenomenal performance and functionality for the price. In fact at the time the idea of someone of Studer's standing getting involved in it seemed inconceivable; as it is, far from

bringing itself down to some lowest common denominator, Studer has made the format its own with a recorder worthy of the illustrious name.

The latest generation of Type II machines proves that real industrial muscle is possible from the format. Alesis' own M20 does the job extremely well, showing us that ADAT is capable of keeping up with everything a professional can ask of it, but the Studer goes that little bit further down the road and puts the stamp of top-drawer quality on it. I kept feeling that if the ADAT format had appeared in this guise from the word go, even at the higher price, it might have made more headway in high-end applications than it so far has. This is true of the M20, but even

more so of the V-Eight, and unless it is too late for tape altogether then the two machines together could still see ADAT come to the fore. Time alone will tell whether the new transport and head drum has the durability and reliability to make it, but first impressions are good, and frankly it is hard to see what would make either company put this much effort into the format if they were not totally convinced of its mechanical staying power. As it stands, this is the only modular multitrack format to offer 20 bits on tape, and it now has a choice of machines with the facilities to compete with anything else on the market. But then I cannot believe it will have the space to itself for long… ∎

See all track and input parameters with the push of one button including routing for all 12 output busses, V-tracks and EQ settings.
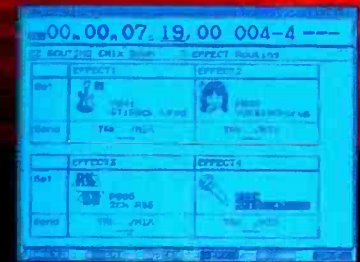
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Allen & Heath GS3000

Music recording is being offered an affordable redefinition of the analogue desk with the added innovation of two valve channels. **Zenon Schoepe** sits comfortably with the idea

IN AN AGE when so many manufacturers are turning their attention to matters digital, it is surprising to see an all-new analogue music recording console appear. The GS3000 is not just a rehash or scaling up of an existing model, it is a bold attempt at looking again at the role an analogue console can still play when it plays to its strengths. Thus we are presented with a good-old traditional 8-bus in-line desk that is available in 24-channel and 32-channel sizes with stereo inputs, MIDI muting, MMC, and the curious inclusion of two valve channels. There is little that is shared in this board with any other A&H board and for all intents and purposes it should be regarded as an all-new design with, in particular, a new mic amp and altogether improved noise performance front to back.

It is also something of a departure for a British manufacturer that, while having a history in recording console production, has more recently been applying itself most intensely to the live and 'affordable' project studio markets. Its first forays into digital, meanwhile, concerned themselves with the installation sector. It does not take transparent perception to conclude that

A&H's digital technology will eventually port across to its other areas of operation, but it has to be said that in the sort of price range where the GS3000 is operating (prices starting at around £3000, UK) there is very little in the way of new releases or attention. Allen & Heath should therefore be applauded for continuing to support an area of the market that has been rather neglected of late. If you are looking to spend this sort of money on an analogue desk with loads of mic inputs, and offers something different, then this, my friends, is it.

Strong points include new mic pres, new EQ circuitry that is remarkable for its flexibility, and that now rarest of commodities—long-throw and short-throw faders for the in-line channel strip. But what sets the GS3000 apart from just about every other console is the onboard inclusion of two channels of valve processing. While not exactly earth shattering, it is still innovative and much more than a gimmick. It is also a practical solution for its target market that adds quantifiable value. This provides a mic-line amplifier with the double triodes working in differential mode and a single-ended mode for a guitar

input to replicate the valve behaviour in guitar amps. In the mic-line mode you can work the circuit comfortably in to compression while the instrument level transcends the concept of DI and enters an area where constructive guitar distortions can be introduced.

The desk is built in blocks of 8 channels with nutted pots throughout to the industrial level of quality now expected from the Cornish company. It is big enough to command the sort of respect that analogue boards still can visually, especially when it has the optional meter bridge. I cannot tell you how nice it is to sit behind an affordable proper analogue desk again.

All I-Os, with the exception of some 2-track phonos and the TRS channel inserts, are balanced on XLR or TRS jacks. Inserts are also provided on the 8 groups. Each channel has a GAIN pot, phantom-power switching, phase reverse, LINE selector, and a 4-band EQ section (that can be split HF-LF) and twin parametric mids between the channel and monitor paths. You are offered ±15dB at 12kHz and 60Hz shelving, and 2dB better each way on the mids, which cover 300Hz–18kHz, and woofer-unseating 18Hz–1kHz. Q for these

bands is fully variable from 0.6 to a not too sharp 2. I am still not convinced about super wide ranging sweepable bands as I find resetting a problem; although I have to admit that the GS3000's arrangement also boasts immense flexibility by being fully parametric in the mids. The overlap means you can ignore part of the HMF's travel if you care to. It is decidedly modern sounding, with lots of low and high-end activity if you need it, but it is easily up to corrective work. In fact, it is a preposterously well featured EQ for this sort of money when you think how many channels there are. You would have been forgiven for expecting a band less, perhaps, but it shows that there is still room for progress in this domain. Its a class-leader in this respect.

Aux arrangement is neat in providing 6 mono auxes accessible from 4 pots. The first two are globally switched prepost for channels, while the remainder are post fader and can be flipped in pairs between channel and monitor paths with an additional switch changing the assignment of two of the pots between Auxes 3-4 and 5-6.

An XFX switch takes the source for the short fader from the long fader output which is useful in mixdown as it uses Mix B—that's right, the GS3000 adopts the now seemingly acceptable practice of providing two main stereo buses that can be split for main stereo and stereo monitor mixes—as a post-fade stereo effects send. Mix A is the main stereo bus with Mix B taking the monitor bus and able to be summed into Mix A on a switch in the main section. Each channel also has a GROUP-DIRECT switch for doing what that suggests in the context of an in-line. Paired 8-bus routeing can be switched between the monitor and channel paths, and that, basically, sums up the main features of the GS3000 input strip.

MIDI capabilities offer automated muting on channel, monitor, stereo channels and Aux masters 5 and 6.

Four mute groups are supplemented by 128 additional patches and includes Safe status for channel and monitor mutes. Operation is simple and you activate patches with UP-DOWN keys and a RECALL switch or via external MIDI command. Each mute corresponds to a MIDI note, so it would also be possible to run the muting from a sequencer. Most surprising is the inclusion of MMC transport keys, arranged unnaturally in a column, and track-record arming is performed by pressing the relevant channel MUTE.

The strip section of the desk that contains all the aforementioned MIDI command-related controls also houses the Solo-In-Place selector complete with safes, but there is no solo trim control.

All frame sizes come with 4 stereo input channels—two basic and two better featured. The more elaborate stereos

have 4-band, Q-less EQ, gain control, 6-aux access, and full routeing while the more basic stereo inputs retain only 2 auxes, input level switching, Mix A-B switching, balance, PFL, automated mute, and a short fader.

A cluster of 8 long-throw faders handle the group functions, each with individual AFL, and either mono or stereo routeing to the main stereo output and these work in conjunction with corresponding bar-graph metering. Group output levels can be switch-selected for -10dB/+4dB operation globally. The main stereo bar-graphs meters can be supplemented by a meter bridge to display channel or monitor paths and the desk undoubtedly looks its best with this fitted.

The master section is completed with aux masters, each with AFL plus a 'blending' function between Auxes 1–3 and 2–4, and two separate studio outputs. Each can draw from control room, Mix A, Mix B and an aux as its source, has a non-automated mute and AFL plus a LEVEL pot. Either of these studio sections can serve as an aux master output when the previously mentioned XFX facility is activated.

You are then into a 2-frequency oscillator routeing to the groups, and talkback through a built-in mic also going

to the groups. Three 2-tracks can be connected, with dubbing switching incorporated and the lot goes out via two pairs of monitors via MONO switch, LEVEL control and MUTE switch.

FOR ALL THIS, it is two valve channels that has drawn most attention to the GS3000—valves on any desk are rare, but their inclusion at this level is all the more remarkable. I must begin by explaining that the two identical valve channels in question should be regarded as patchable processing commodities that have to be physically plugged in to and out of. They will go nowhere on their own from the top panel. At the back of the desk you will find each valve channel's connectors for mic (XLR), line and instrument inputs plus a TRS socket for patching the output of the valve section into any channel insert. Look at the two valve channels as patchable tube front ends.

Each valve input has phantom power, a wide ranging GAIN pot, LINE selector, GUITAR MODE selector with associated single-band bell EQ, a HIGH CUT SWITCH, output LEVEL pot and PFL. However, central to it all is a chicken-headed pot controlling valve drive that works in conjunction with a tricolour LED to give an indication of how hard the bulb >
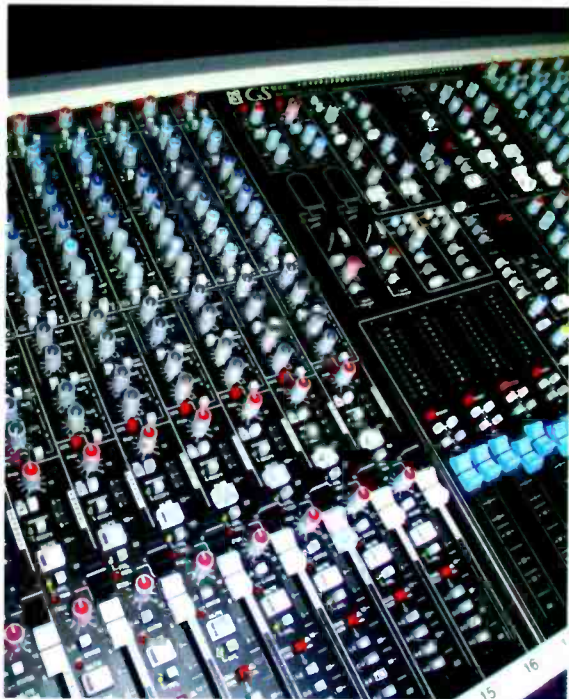
< is being made to work. What you have here is an alternative route into a channel that lessens the need to buy valve outboard. On its best behaviour for mic and line signals you can hit cliché valve compression without too much trouble as balancing the Input Gain against the valve Drive control can range all the way down from notice-ably compressed and thick to 'you wouldn't think there was a valve in it at all'. This suggests that maybe the range could have been a little hotted up to make the effect that more appar-ent at the lowliest settings, but I am not really knocking it. You can fatten a vocal or a drum track quite effectively by simply passing them through here, and beefing them up with a tad of squash, and the high-cut switch is there to smooth out the top end if you want it even more rounded.

Plug in a guitar and the personality changes. You can go to pretty hefty overdrive and benefit from the guitar mode-only swept 120Hz-6kHz EQ with ±10dB of boost to add that little extra honk, particularly as it influences before the signal hits the glass.

On super-saturated settings I would not say it rivals the best dedicated gui-tar valve processors, but it is easily on a par with any budget outboard with valves in it that takes instrument-level signals. Clean to middle sounds are the forte; although they cry out for a bit of ambience, but the all-out sound is more con-vincing on lead lines than it is on chords because it is not quite complex enough on full grunge.

To get the most out of the valve chan-nels you will have to have a patchbay well sorted if the frequent requirement to scamper around underneath the desk is to be avoided. With a bit of thought it is possible to get a valve into just about most signals that are going through the desk; although with only two valve chan-nels to play with, the track-ing stage is the place to do it. Of course, you are also be free to take keyboards and just about anything else through these sec-tions. It is a master stroke of clever thinking and gives the GS3000 a really competitive edge against anything else for the money.

It is an alarming fact that of all sectors, it should be the low end that is being spoon-fed the promise of digital most forcibly. Some-how it has always seemed natural that postproduc-tion mixers would become digital but the resistance and lack of choice has significantly kept music recording a predominantly analogue desk affair. At the self-op end of afford-able there are many instances where a digital desk offers benefits that an ana-logue equivalent could never match. However, at the same time there remain numerous instances where people just want to be able to record 24 mics in real-time cheaply, and in such cases the choice has become radically reduced. Such an intended task is largely impos-sible on any of the digital desks that weigh in anywhere near the sort of money we are talking about here.

What Allen & Heath has done is rede-fine the affordable analogue music recording console and shown what can, in fact, be achieved now with careful production engineering. Perhaps its market is not enormous, but the demand is certainly large enough to welcome a console like this.

You've really got to like it. Good solid performance front to back, dead quiet, nice mic amps, and class-lead-ing EQ. Throw in MMC, automated muting, lots of inputs, some stereo inputs and the odd valve to give you a different palette of possibilities, especially on vocals, then there's noth-ing to argue about. Most importantly, in the face of digital compactness and intelligence, it is still remarkable value for money.

They may never sell as many GS3000s as Yamaha has 02Rs, but I believe there is an untapped poten-tial market out there that will be very grateful that Allen & Heath has been bothered to produce some-thing new for them. Here is a novelty then—a really good, affordable ana-logue console brand new for 1998. ■

Contact

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Digidesign ProControl

Digidesign's ProControl offers the kind of authority over a Pro Tools session that transforms a project studio environment into a world-class digital recording facility. **Roger Nichols** clicks on the icon

OVER RECENT YEARS, Pro Tools has become a progressively more powerful instrument for record production. The problem at one time was that Pro Tools was thought of as a powerful piece of outboard gear, rather than the focus of the recording environment. Typically, studios had a console, some tape recording machines and a Pro Tools system. When asked how the Pro Tools system was used, the answer was that after recording to tape, material was transferred into Pro Tools, edited, tuned up, noise reduced, time stretched, and transferred back to the tape machine. When it came time to mix, the tapes were played back through the console using the console automation for level control. The major complaint was that when trying to use only Pro Tools for recording and mixing it was too hard to make quick changes to EQ, set up stereo headphone mixes, or perform final mix-downs with a mouse.

During the autumn of 1997 I helped Bela Fleck record an album using Pro Tools. The recording, editing, overdubbing and mixing were done entirely within the Pro Tools environment; so now we know that it can be done. The only thing lacking was a professional control surface to make the recording process feel more like a standard recording environment. Enter ProControl.

I recently got my hands on a ProControl and connected it to my Pro Tools 24 system. ProControl requires Pro Tools v4.2 or higher to enable the remote control functions. The connection from host computer to the control surface is via Ethernet. After connecting the cable and updating my software, I enabled ProControl in the new Preferences menu. Instantly the names of the tracks in the Pro Tools session showed up on the scribble strips above the touch-sensitive moving faders on the ProControl. A blue border surrounds the track name on the Pro Tools screen, telling you which tracks are being controlled by the ProControl. Total time to get the whole system connected and running was less than 10 minutes.

My initial impression was that ProControl looked and operated like one of the million-dollar digital consoles that I had mixed an album on just a few months ago. The basic ProControl system consists of a centre section with controls and functions that are common to all tracks, and a motorised 8-fader section that contains the track-specific controls, such as Fader Level, Mute, Record Enable, Pan, Solo, Mute, Insert Select. Automation mode and scribble displays above each fader. The scribble strip displays the channel names, send levels, fader levels and channel delay. Optional expansion fader units, allow you to expand in increments of 8 faders, up to a total of 32, are also available. Above

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the moving 100mm fader is a rotary control with LED indication of knob position (along with a scribble strip for parameter names and values). These knobs are used for send levels, pan position and I-O assignment. Eight stereo peak meters are included in the fader section. The 100mm fader controls audio level with 1,024 steps of resolution. This amounts to 1/10th dB steps through 90% of the fader travel. None of the low cost digital consoles I am familiar with have that high a degree of fader resolution. (On playback, Digidesign's DAE interpolates between those values to yield 24 bits of resolution, which is near-analogue performance.)

The centre section of the control surface provides the command centre for the Pro Tools session. The DSP edit area offers 8 rotary controls with LED display area for assigning and editing plug-ins without going to the computer display screen. Just turn a knob to select a plug-in, press one button to load it, and then all the editable parameters are displayed for easy tweaking. If you decide to perform your edits on the computer screen, there is a built-in track pad and shuttle wheel to help, and the results are simultaneously displayed on ProControl.

Six monitor-level meters and monitor switching give control over stereo and 6-channel surround mixes. Bank

PRO TOOLS 4.2

I mentioned that a new version of Pro Tools was required to support the ProControl, but that is not all you will find in the v4.2 update. Aux sends are now stereo—when you activate a stereo send, you get an additional little fader above the primary fader on the screen, along with a meter display and pan-position indicator. It is now easy to set up a stereo headphone mix for overdubs. As a matter of fact, if you have a mix you like on the main faders, you can just copy it to the aux sends.

The next new feature I noticed was the increase in available tracks. With one d/24 card you can have 32 voices and 64 tracks, and with the d/24 expansion kit you have 64 voices and 128 tracks of audio at your disposal. You are still limited to 72 channels of I-O, but I find that I always need more internal tracks for things like alternate solos or additional vocal attempts that need track space but do not compete for I-O.

switches let you switch among virtual banks of tracks so that you can control as many tracks as you want with the number of physical faders in your configuration. That is, if you are dealing with 128 tracks, you can page through them 8 tracks at a time with the basic ProControl configuration (US$11,995), or you can page through the 32 tracks at a time if you have the three additional 8-channel fader packs (US$6,495 each). If you are going to be doing big mixes, then the 32-channel setup will be well worth the additional cost.

Pro Tools began life primarily as a computer program with displays on the screen and hardware hung on the computer to get the audio in and out digitally. Since 1988 when the first Sound Designer software was released, the user primarily has been dealing with the computer to get the work done. When Pro Tools was introduced, third-party vendors designed hardware boxes that would control things, like fader movement, within Pro Tools, but they were not motorised faders and you still had to focus most of your attention on mixing by mousing around on the screen.

When you first start using ProControl, the feeling of 'a computer with stuff plugged in' is quickly replaced with 'digital hardware console with computer support'. What you have is a high-end control surface with a remote rack of processing gear and I-O interfaces. In this case the gear in the rack happens to be a Macintosh, 888/24 I-O boxes, ADAT I-Os, USD synchronisation interfaces and disk drives. The beauty of all of this is that you can add functionality as you need it. I needed extra tracks, so I added a Pro Tools 24 expansion kit. I needed more on-line storage, so I added another 9Gb hard disk.

ProControl has taken the Pro Tools environment from a hard-disk digital audio recorder with some mixing features, to a full-fledged hardware digital recording console with built-in, multi-track, hard-disk recording. Digital audio production will never be the same again. I still enjoy working at world class studios with high end gear, but in my project room, I can now have the same recording capability, and enough money left to buy a small island country in the Caribbean. I just have to think of a name for it. ∎

LA Audio DigEQ

Making a grand statement with its first digital outboard unit, LA Audio's DigEQ is a powerful equaliser and dynamics processor. **Dave Foister** weighs his catch

YOU KNOW a small battle has been won somewhere when a dyed-in-the-wool analogue company goes digital. LA Audio's reputation has been quickly and successfully established with a very traditional range of analogue signal tweakers, so it must have been a major decision to develop the DigEQ, a new venture that squeezes an awful lot of traditionally analogue processing into a digital box.

Both the name and the front-panel's first impressions rather underplay what this unit can do. The panel is built round a large blue screen, and one of many features designed to provide analogue-style accessibility is a splendid array of small push buttons, labelled with ISO third-octave centre frequencies. This and the name could give the impression that the DigEQ is nothing more than a smart graphic equaliser, whereas in fact it goes much further; not only does it provide virtually every flavour of EQ you can think of, but it adds dynamic processing as well, a fact only hinted at by the presence of a single DYNAMICS button next to the display.

The box supplied, ahead of actual release, surprised me by only having analogue ins and outs; I would have expected the easier option to have been the digital interface, omitting the convertors. The full version has the lot, with AES-EBU and SPDIF alongside word clock. Processing is 24-bit and the AES-EBU interface will handle the full word length, and the onboard convertors both ends are also 24-bit.

It is apparent, right out of the box, that the DigEQ is going to be easy to use. Making it as easy to get to grips with as an analogue unit is expressly stated as a design objective, and a cursory glance reveals that the mission has, by and large, been accomplished. The screen dominates, and there are no soft keys in sight; although the piano-key-style row of frequency buttons does have more than one function, depending on what the unit is doing. These are colour-coded and clearly labelled on the panel, and obviously select parameters for adjustment. It is also obvious that adjustment is to be performed with the big DATA knob, a continuous rotary encoder, and the only knob on the panel. An adjacent pair of LEDs suggests the knob can also be used for parameter selection, and the only thing that caught me out in the absence of the full-blown manual was the fact that the knob must be pressed to switch between its two functions. Once you know that, operation becomes intuitive and fast, a model of how to control a digital processor.

Which is just as well as there is a lot to control. The three primary functions of Graphic, Parametric-filters and Dynamics are all summoned with big dedicated buttons beside the screen, presenting the relevant display page and activating the corresponding controls. The initial impression that there is a 31-band graphic lurking within is, of course, correct; the screen shows the sliders, and the knob moves them up and down as selected by the row of buttons. Unfortunately, the row of buttons is longer than the slider display, so the two do not line up, and it is easy to hit a button below a particular slider and end up moving one several bands away. It is important to actually look at the frequency label on a button before hitting it, and, fortunately, despite the density they are reasonably easy to see.

The graphic can work in ganged stereo or as two independent channels like the other EQs, or even as two linked channels with an offset. Both channels are shown simultaneously, one having solid 'knobs' on the screen while those on the other are shown in outline. A neat trick is the ability to select a group of several bands, by holding down the buttons of the outer two, so that they can all be adjusted together. This works even when the existing gains of the bands within the group are different, adding an offset to the whole curve. The paperwork says that a second press of the master GRAPHIC button should display the actual frequency-response curve resulting from the settings, but this was not implemented on the review sample. When it is in place it should form a useful and illuminating adjunct to the main slider display, reminding us that despite its name, a graphic EQ is not quite a WYSIWYG device. I am told that it will, in fact, go further than that, and show the final overall curve resulting from the settings of all the EQ sections together, possibly with the option of selecting which elements' **>**

< contributions should be drawn and which omitted for the sake of clarity.

The next group of functions provides the parametric and shelving EQ together with high and low-pass filters, and it is important to note that all the various EQ options can be used together —nothing is ever compromised to provide DSP power for another function. Parametric EQ comes in the form of three identical bands with fully variable frequency, gain and Q. Access to these parameters is via certain of the buttons used for selecting graphic frequencies; there are three sets of three, coloured blue to make them stand out from the black ones around them, and labelled appropriately. Thus every band has direct access to every one of its parameters live, all the time, just needing one button push to give the DATA knob control.

The range of some of the DigEQ's parameters is unusual in itself. In the parametric, for example, the gain is adjustable from +15dB to -40dB, and the bandwidth goes as low as $\frac{1}{9}$ of an octave, giving the potential for some pretty surgical notches. Of course, the frequency adjustment is not continuously variable, but goes up and down in small steps; at $\frac{1}{8}$ of an octave, these are much narrower than the spacing of the bands in the graphic, and close enough together that sweeping the con-

trol sounds pretty much like a continuous pot. The narrowness of the filters at their extreme shows up the only disadvantage, where an attempt to notch out a particular frequency (in my case the slightly off-pitch 1kHz oscillator on the console) fails to catch it as it falls between two steps, and the bandwidth is so tight that it misses it. In a sense this is a credit to the design, and is certainly no different from any other digital EQ implementation I have seen.

On the same screen are the high-pass and low-pass filters, which besides variable cut-off frequencies (with outrageous ranges) also have the facility to alter their characteristics. This is something we have come across before in digital EQ, but is rarely seen in analogue form. Three shapes are on offer: Bessel and Butterworth 12dB per octave configurations; and Linkwitz-Riley 24dB per octave. The two filters can have different shapes, and the resulting precision, together with the fact that the cut-off frequencies virtually meet in the middle at around 650Hz, almost looks like overkill.

A second push on the main Parametric function button calls up the

shelving equalisers, one at each end. These have fully variable gain and turnover frequencies, and again can be used at the same time as the parametrics and the graphic. The available gain range is a good bit smaller than on the other EQ sections, so these are clearly for a touch of overall sweetening after the hard work has been done elsewhere.

This much EQ power all at the same time is rarely seen together in one box, but the DigEQ goes a stage further by including a full set of dynamic processing functions. Two screens of parameters are available, one controlling the compressor-limiter and the other the expander-gate. Note that in this case, unlike the EQ, it is an either-or decision; you can not have compression and limiting together, but you can use either of these with the expander-gate section.

The parameters are not quite so all-encompassing as they are in the equalisers, but a fair range is provided that covers most situations. Thus Threshold, Attack, Release and Ratio are all adjustable in fairly sensible increments, and the gain can be made up after processing. The ratio goes all the way from 1.4:1 to brickwall limiting; although the attack time remains variable in limiting mode allowing substantial overshoot on transients if that is what you want—it also goes

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COMPRESSOR/LIMITER `ON` Ch:A+B

Gain: -1.5 dB
Threshold: +5.0 dBu
Ratio: Limiter
Attack: 2 ms
Release: 50 ms
Knee: soft

A B
-6
-20
-40

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down to 10µs for proper peak limiting. There is a pair of nudge buttons beside the DATA knob for selecting parameters; on the EQ screens it jumps around with no apparent logic, but here in the dynamics displays it is the quickest way to climb up and down the menus. A pair of gain-reduction meters show what is going on, and, again, the two channels can be used independently, or as a linked stereo pair. Unlike the graphic, however, it is not possible to have the two channels linked, but with differing settings; putting it into stereo when the two are not identical produces a display prompting to copy either channel's set-up into the other. Settings are entirely manual; unusually there is no automatic or programme-dependent function for the time constants. Many will say this is no bad thing, forcing the user to think about what the compressor should be doing, but often an auto setting can be much more forgiving and flexible when presented with a signal whose character is not constant.

The DYNAMICS button toggles between this screen and the one dealing with the expander-gate. Here again the setup is relatively straightforward, with variable threshold and time constants, and

```
EXPANDER/GATE              ON         Ch:A+B
                                  A    B
Threshold:        -60.0 dBu            -6
Ratio:               Gate
Attack:              10 us             -20
Release:            100 ms
Knee:               hard              -40
```
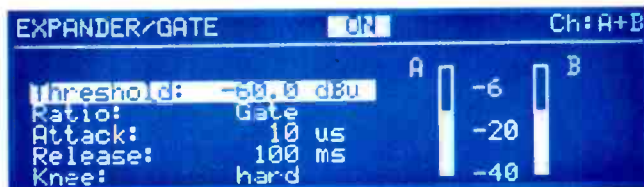
a ratio control with Gate as its extreme setting. Gain-reduction meters again show the unit's operation, and it is easy to set it up for clean straightforward problem solving.

Once memories on an equaliser would have seemed an odd idea, but with this lot on board it is essential. There are 98 program-memories, and they can all be named as well as numbered. A particularly useful Compare function allows checks to be made not only on what you have just changed, but how the current edit compares with any other preset.

An interesting option, not fitted to the review sample, is a Real Time Analyser board, that since it has its own DSP can be used even when all the DigEQ's conventional facilities are running. The front-panel controls are already in place, and compensation for several different types of measurement microphone is available. Another option is remote control, with possible wireless operation and master control of up to 16 units; even without a remote,

multiple DigEQs can be set up as a Master-Slave system.

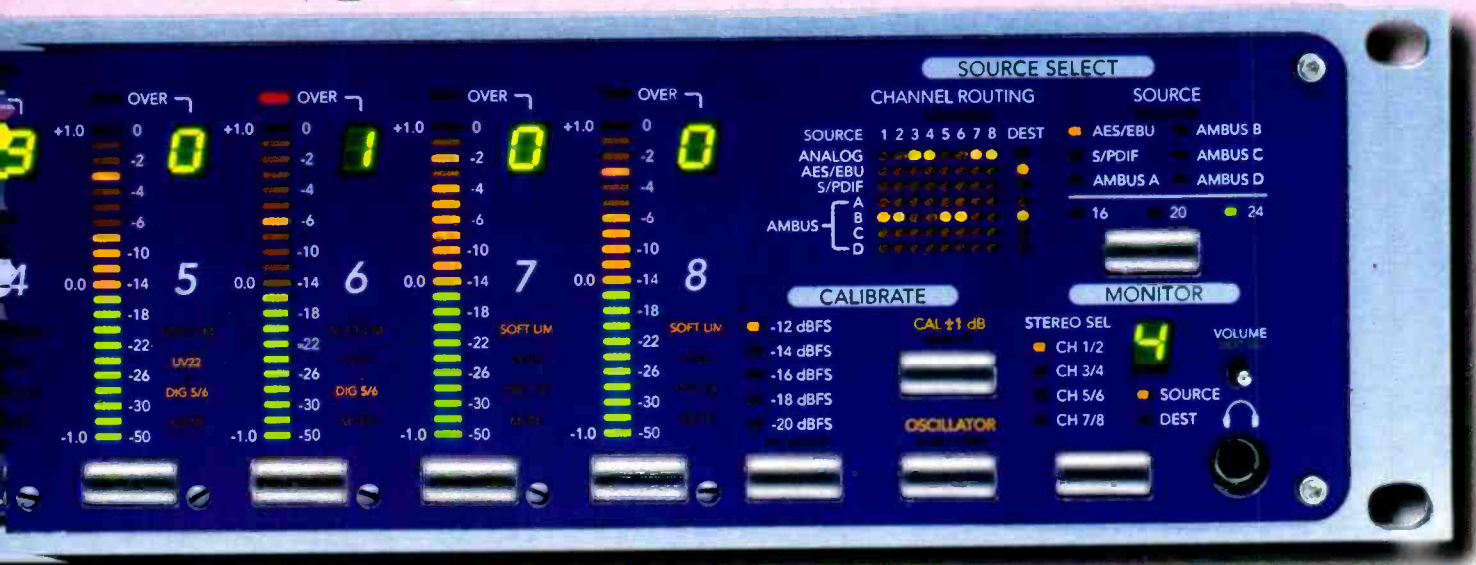One gripe I have is the supposedly helpful time-out on the display screen, making it go blank if you do not adjust anything for a while. Not only is the time-out period very short, but it ignores the fact that you might want to watch the dynamics meters, and I understand this is to be made switchable. Besides this, the convertor inputs have no variable gain controls and are set to 0dBm=-12dBFS, which means that a -18 setup causes overload difficulties.

But it is worth getting it right as the raw power of the DigEQ is coupled with a good controllable sound, with all the EQ sections having an excellent clarity and neutrality. There is not much in the way of character, which may bother some; instead there is a feeling that it is doing exactly what you tell it to, no more no less, and as it is so easy to tell it what to do and keep track of it, a formidable tool indeed. The dynamics sections are workman-like and competent, more than just the icing on the cake, but not the reason for buying one. The real strength is the simultaneous and easily-accessible availability of virtually any kind of EQ you could want, keeping all camps happy and providing real power to attack a situation from many angles, solving problems as well as gently tweaking. ■

Bellari RP583 and RP562

No longer the preserve of the well-heeled studio-owner, valve outboard is infiltrating every level of recording. **George Shilling** evaluates Bellari's comp-limiter and sonic exciter

THERE MUST COME a point when the market is sufficiently saturated with valve equipment that pro-audio manufacturers will rediscover 'the clean vintage sound of transistors'. Until that day, however, we are likely to be bombarded with equipment featuring valves at all ends of the market. These Bellari units most definitely fall into the lower end—few competitors come to mind at these prices—however, these are as much solid-state as they are valve units, featuring just one buffering Bellari-branded dual-triode 7025-ECC83 per channel. At this end of the market, you may be surprised to find out that Bellaris are 'carefully hand-assembled and individually tested in Salt Lake City, Utah' in the good ol' US of A.

The first examples of these two units received from the UK distributor were a bit of a disaster. I suspect that some time after the careful hand-assembly they were handled less than carefully in transit. Surprisingly, they both were inextricably connected to US mains plugs: (no IEC sockets here). I may be wrong, but I was under the impression that any electrical equipment sold in the UK now had (by law) to be fitted with a UK 3-pin plug. Anyway, it was off to the shops for some plugs, and out with the knife and screwdriver. After using the RP583 for a few minutes it started making some very strange noises: self-generated whooshing sounds from an industrial sci-fi soundtrack, even when I unplugged the inputs. Onto the RP562, and I nearly blew my speaker drivers with the deafening whistling and throbbing noises coming from the outputs. This one was DOA. A few weeks later I received replacement units directly from Rolls in Utah. This time things proved rather more worthwhile.

The Bellari range is characterised by brushed gold-effect front panels, not dissimilar from that found on your parents' fifties Belling electric fire. Their knobs have a vintage look, perhaps like a sixties electric guitar, but with a pleasantly damped feel, while their push buttons are reminiscent of some you may have pressed on a black-and-white TV a few decades ago.

The RP583 is housed in a fairly shallow 2U-high case that is rugged, but quite lightweight. This photo-electric stereo compressor is well-featured by any standard. The back-panel features clearly labelled and well-positioned inputs and outputs on mono jacks and also XLR sockets. While the female sockets do not feature any latch, the males are apparently equipped with plated pins. Also, each channel features a further two jack sockets for side-chain input and output. The mains lead comes straight out of the back panel through a grommet, and no fuses are accessible outside the case.

On the front panel each channel features separate controls. Furthest left is a push button labelled ACTIVE with a red LED. There follow knobs for OUTPUT LEVEL, THRESHOLD, RATIO, ATTACK and RELEASE. Apart from the OUTPUT knobs, all have a somewhat pointless vaguely centre-ish detent. Next there is a STEREO LINK push button, then a push button on each channel to select meter indi-

RP583 stereo compressor: well featured by any standard

cation as Output level or Gain Reduction. Two vu meters are arranged side-by-side, illuminated by a yellow glow. Finally to the far right is a rocker-type power switch.

The STEREO LINK button and separate controls make this unit fairly flexible: it will happily work as two independent mono compressors. In Linked mode, Channel 2 oddly becomes the master channel. This usefully leaves both channels OUTPUT knobs active, but none of the other Channel 1 controls operate.

When the unit is bypassed, Output gain is fixed, but strangely, there is a small increase in gain of a few dBs when inserted in a signal path. With the unit switched in, some inaccuracies of the legending become apparent. Zero is marked on the OUTPUT knob only a quarter of the way up from completely Off, and unity gain is about half-way between indicated Zero and Off. There is an indicated 22dB of gain at the top of the OUTPUT knob, and this is approximately true, although one rarely needs this much gain. This is not a precision calibrated piece of equipment: there are slight gain variations between the two channels. Threshold is indicated as variable between ±20dB, but as the compression knee is very hard a small

change to this control can make a huge difference. Ratio is variable from 2:1 to Infinity, via 1:1 at 10 o'clock and 5:1 at 2 o'clock. Attack is marked as being variable between 0.5 and 100ms, and Release as 0.1 to 2.0s. Release characteristics are pleasant enough, but I did not find the Attack characteristics particularly likable. At the onset of a signal there is a sudden step as the compression kicks in. This is partly a characteristic of optical-sensing compression, but is hard and unlike valve compression as most of us know it. A similar, but friendlier, sound can be achieved with, for example, a Joemeek, but with this unit it is often impossible to avoid, and there is not the range of adjustment here to enable any real smoothness with moderate or heavy

> The Bellari range is characterised by brushed gold-effect front panels, not dissimilar from that found on your parents' fifties Belling electric fire. Their knobs have a vintage look, perhaps like a sixties electric guitar

compression. The ATTACK control works almost as if it is simply a delay setting before the sudden onset of limiting. It does not really go to a slow enough setting for my taste for vocals; although it was just about usable, with care. On bass guitar the unit was better. The attack it gave to the transients of the notes and the smoothing of their tails was actually very good, and not dissimilar to a dbx 160 in character. On heavily strummed acoustic guitar the Bellari brings a great deal of energy to the sound, but you still have to be careful to avoid the attack blip at the start of a section. On drums the compression tends to be squashy, choking >

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Stereo Tube Sonic Exciter · Bellari · RP562

< and lacking air. On entire programme material, the unit is surprisingly good on rocky guitar music, but it has a tendency to sound spongy.

THE RP562 Stereo Tube Sonic Exciter has a 1U-high case of similar depth to that of the RP583, making it very small indeed. Like the Compressor, inputs and outputs are available on jacks and XLRs. There is an extra jack socket is labelled SUB OUTPUT. There are knobs for Sub Frequency and Sub Level, a Sub Clip LED, an Active push button and LED, and knobs marked BOTTOM and DEFINITION. There are two tiny round illuminated vu meters to the right. A rocker switch to the far right clicks the power on.

Despite the centre detent clicks of the BOTTOM and DEFINITION knobs they are legended 0 to 10 where flat is approximately 4 on BOTTOM and 0 on Definition. Having said that, some tonal coloration of the midrange is apparent, even in these positions. Increasing Bottom seems to add a broad low-frequency shelf EQ from about 200Hz down, and simultane-ously reduces middle and top, giving an impression of huge warmth. Decreasing from flat seems to roll off this low shelf and increase the frequency at which this happens as you approach zero. Definition is close in character to a slightly harsh-sounding 10kHz (very) broad EQ boost. Apart from the obvious EQ changes, Definition seems to add a subtle distortion not unlike that of a worn vinyl record; although I have certainly heard nastier sounding exciters than this.

The Sub Output is controlled by the SUB FREQUENCY and SUB LEVEL knobs, which have no effect on the main outputs. This is designed to be fed to a subwoofer amp and speaker, and operates regardless of the ACTIVE button. FREQUENCY controls the crossover point from which the output operates downwards. Frequencies range from 35Hz to 200Hz. This works fine, but as there is no corresponding roll-off of the main outputs it is only really half a crossover. Also, a third channel of amplification would be needed for the subwoofer. You could, of course, use the Sub Output in a mixing setup, to add a low frequency boost to certain signals, perhaps without other processing applied to the main signal.

Neither of these units sounds especially valvey, but construction is solid and internal layout neat. They feature minimal venting to the rear of their cases, but do not seem to get particularly hot. On both units the unbalanced jack connections operate at the same (vaguely) +4dBu reference level as the XLRs. This is strange for cheaper units such as these, which you would imagine would mostly find homes in home studios and lower-end project studios which operate at -10dBV. Their manuals are simple and straightforward, if a little amateurish. When used at the correct operating level these units are reasonably quiet and have usable headroom. The Exciter is certainly unusual, and at this price the Compressor is definitely worth a look. ∎

Sonic Foundry Soft Encode

Extending the ready compatibility of Dolby Surround with the emergent DVD medium, software encoders are on the rise. **Rob James** checks the Foundry's hottest casting

DOLBY DIGITAL, introduced in 1992, is the trademark for Dolby's AC-3 data compression technology. It has established itself as a major standard for theatrical releases, and is the audio format adopted for HDTV by the Grand Alliance in the US and, more recently, as the only required format apart from PCM for all PAL and NTSC regions of DVD. (Although Region 2 retains the option of MPEG2) As with the original analogue Dolby Surround, for theatrical release the encoding hardware is owned by Dolby and often only installed temporarily for the final stage of the mixing process. All material displaying the Dolby logo is required to be licensed and the company reserves the right to check quality. Stand-alone hardware encoders and PCI-card-based encoders have already appeared for non-theatrical use. To my knowledge this is the first software-only encoder to hit the general market.

Soft Encode appears deceptively simple to use. Open a selection of .WAV or raw PCM files, assign them to the six available channels and hit the ENCODE button. In fact, there are a vast number of options and parameters to consider. But first, a couple of snags.

One might innocently presume, having encoded your masterpiece, all you would have to do to check it, would be to plug the digital output of the sound card into a consumer level 'home cinema' decoder. Unfortunately, at the time of writing, this will not work with any known PC sound-card because the cards add an 'audio' data bit to the signal that most domestic decoders interpret as an instruction to pass the encoded data as audio. This is not good for the speakers. Professional decoders ignore the audio data-bit and get on with decoding. Even if this problem were solved, using a consumer decoder for checking masters is definitely not recommended by Dolby because there are a number of decoding modes in consumer decoders, such as compression and downmixing, that are factory preset and likely to mislead. It is important that the mastering engineer should be able to emulate various consumer decoders to check compatibility. Fortunately, Soft Encode will also decode an AC3 file to enable an audible check to be made. This brings me to the second snag.

With current Pentium processors the encode process takes around 6x real

time to achieve. Obviously this will improve as processors become more powerful. Meanwhile Soft Encode provides tools which help to ameliorate the problem if a hardware decoder is available. It is possible to select an area that looks as if it may be a problem and preview it. If parameter changes are then necessary the time overhead need not be great. There are also powerful batch processing functions that will enable jobs to be carried out without intervention, perhaps overnight. Dolby Digital is a great deal more than a simple data compression system for the delivery of program. For a start there are no fewer than seven possible channel designations plus the LFE (Low Frequency Effects) option at original sampling rates of 32kHz, 44.1kHz or 48kHz (uncompressed). Then there are a number of additional information fields carrying data regarding the bit rate; how the various channels should be downmixed for playout where there are fewer channels (stereo or mono); whether the material is copyrighted; if it is encoded in Dolby Surround; whether it is an original or copy; what the original dialogue sound pressure level was when mixed; and the type of room it was monitored in. Once this lot have been decided and set there are a number of preprocessing options. De-emphasis; DC removal; low-pass filtering; plus another low-pass filtering option specific to the LFE channel and a 90 phase shift option for the surround channels, which is useful if downmixing to two channels. There are also a number of parameters relating to dynamic range compression. Space does not permit me to go into all the ramifications of all these here.

Soft encode is not a mixing tool. It does provide for changing the balance between the tracks, but this is intended for minor trimming. Similarly, this is not an editing tool. Sync between tracks can be adjusted, but this is a strictly manual

operation and should be treated with caution. The facility is useful for correcting minor timing differences, but the assumption is all source files will already be synchronous. The main screen is a familiar looking horizontal track display. Files are opened in the usual Windows manner and appear as tracks that can be assigned to channels by clicking on a graphic representation of the speaker positions. A maximum of six tracks can be opened together. This can become an issue since files can contain multiple channels of audio and each channel in a file is opened as a separate track. To the left of the track display is a block of mixing and playback controls per track. Transport controls are much the same as other Sonic Foundry applications with the exceptions of a PREVIEW key and an ENCODE key. The track display can be zoomed vertically and horizontally.

With Soft Encode, Sonic Foundry has brought Dolby Digital encoding within the reach of users who, until now, would not have considered doing their own encoding. There are up and down sides to this; if you though CD mastering was fraught with peril the number of variables in Dolby Digital makes it look like a kindergarten. In the hands of a careful operator, who either has the required knowledge and experience or is capable of learning to get things right for the intended market, Soft Encode is an extremely powerful tool. The downside is that it may be all too easy to get it wrong. Soft Encodes provide a golden opportunity to learn about the whole Dolby Digital process and to gain control over encoding in house. ∎

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Quested VS2108

Studio Sound's 'bench test' loudspeaker reviews continue with the VS2108. **Keith Holland** reports

THE QUESTED VS2108 is a 2-way active loudspeaker based around a 200mm doped-pulp woofer, a 28mm soft-dome tweeter, and a built-in amplifier and crossover package. The cabinet has two front-mounted ports, and external dimensions of 338mm wide by 340mm deep by 400mm high. The loudspeaker is magnetically shielded. Connection to the amplifier package is via an XLR3-type connector that may be used with either balanced or unbalanced equipment. There are LF and HF contour switches, and a selector for +4dBu or -10dBu input sensitivity. The crossover is specified as having 24dB

per octave slopes with a crossover frequency of 1.25kHz, and includes 27Hz 'subsonic', and 250kHz ultrasonic filters. The manufacturers specify a maximum SPL of 108dB(A), but do not state the acoustic conditions where this level is reached, or whether this is for a single loudspeaker or for a pair.

The manual states that 'the VS2108 is designed to be mounted vertically and should not be turned on its side'. The cabinet is fitted with heatsinks at the rear of the sidewalls that have vertical fins; horizontal mounting

could result in overheating as well as degradation of the acoustic performance.

At 22kg, the VS2108 is the heaviest, and, indeed, largest loudspeaker tested in this series to date; speaker stands are recommended, but meterbridge mounting should be possible on the more rigid consoles. The loudspeaker is finished to a high standard, and feels as solid as its weight suggests.

Fig.1 charts the on-axis frequency response and harmonic distortion for the Quested VS2108. The response is shown to be within ±3dB from 45Hz to 3.5kHz—except for a narrow dip at 1.6kHz—and within ±5dB from 40Hz to 20kHz. The roll-off below 50Hz is reasonably gentle considering the ported cabinet and built-in 'subsonic' filter; this feature is borne out by the acoustic centre result (Fig.2) that charts slightly less group delay at low frequencies than other, similarly aligned systems. Harmonic distortion is maintained at less than 40dB (1%) below the fundamental except for a narrow peak to -37dB (1.4%) at 800Hz; the low-frequency distortion performance is particularly good. The horizontal directivity (Fig.5) is well controlled except for a widening of the polar pattern at 4kHz; there is little evidence of side-lobes from either drive-unit. The vertical directivity (Fig.6) is similarly well controlled except for the inevitable dip at the crossover frequency due to the physical spacing of the drive-units.

Fig.3 charts the step response of the loudspeaker. It can be seen that there is a delay of some 1.8ms between the start of the high-frequencies and that of the mid frequencies. Reference to the acoustic centre plot (Fig.2) confirms this delay; mid frequency transients (1kHz) effectively

start some 0.25m behind high frequency ones (10kHz). These results indicate that there may be a crossover alignment problem. The power cepstrum plot (Fig.4) is well behaved with some evidence of echoes at 180µs and 400µs; these may be due to baffle edge diffraction and are probably responsible for the mid-high frequency response aberrations shown on-axis in Fig.1. Fig.7 charts the total power output response of the loudspeaker. The most notable

features are the dip at the crossover frequency of 1.25kHz and the widening of the directivity pattern around 4kHz.

To sum up. The Quested VS2108 performs reasonably well, with very commendable low frequency performance, both in terms of response smoothness and harmonic distortion. At higher frequencies though, the loudspeaker does have some frequency response peculiarities which take it outside the desirable ±3dB tolerance. ■
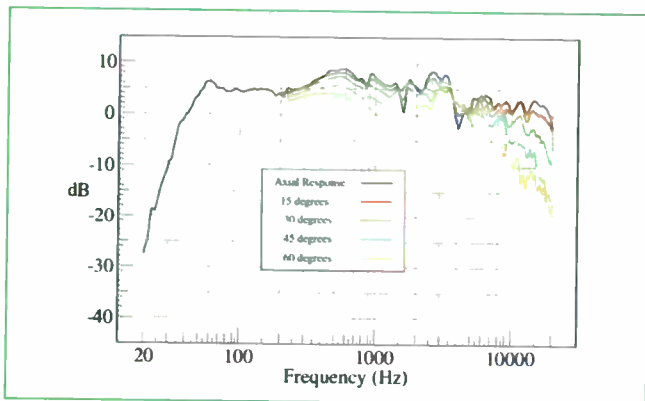

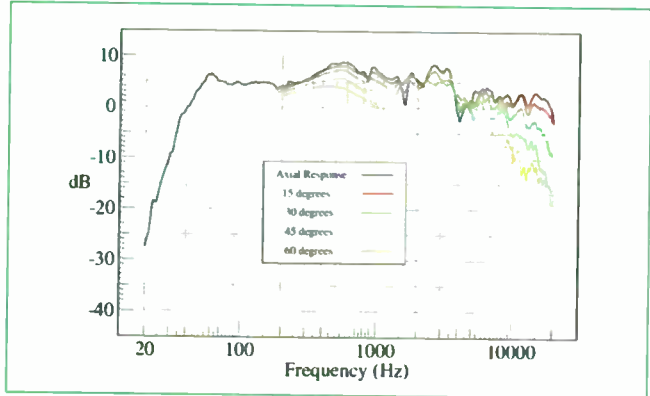**Fig.5: Horizontal directivity**

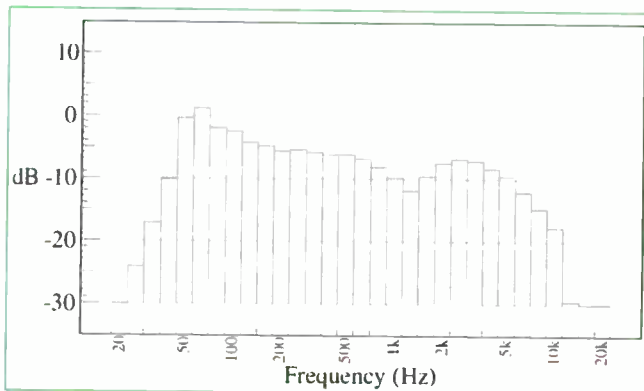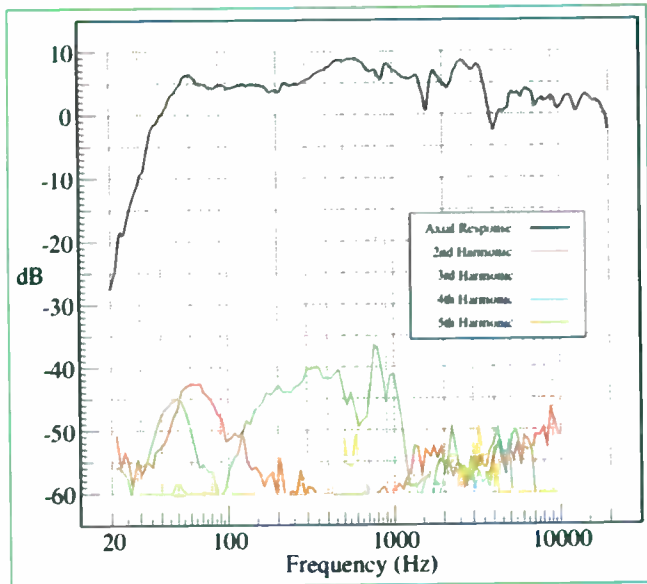**Fig.6: Vertical directivity**
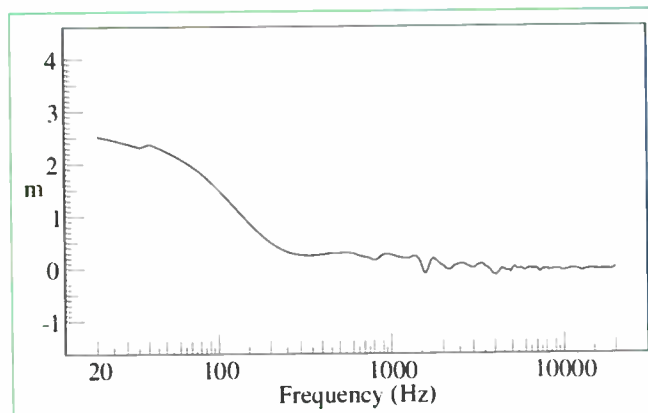

Fig.7: Power response

Fig. 1: On-axis response and distortion

Fig 2: Acoustic centre

Fig.3: Step response

Fig.4: Power cepstrum

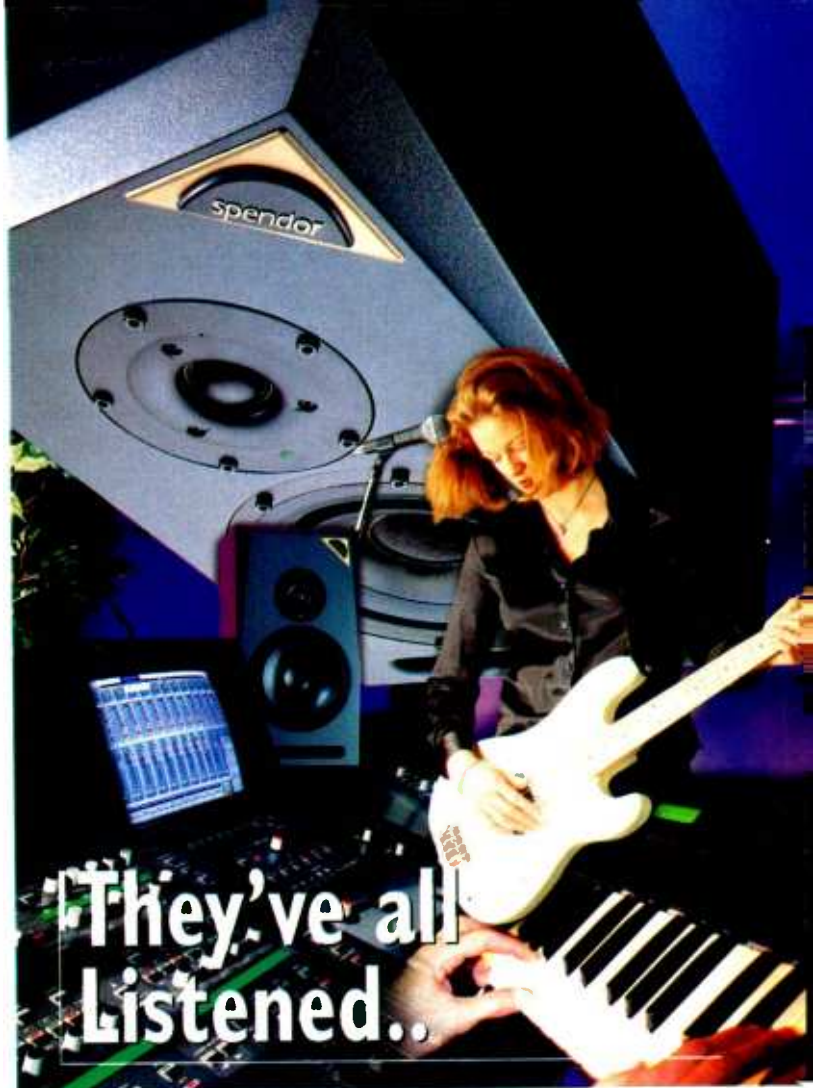**Studio Sound** July 1998

www.americanradiohistory.com

NEW TECHNOLOGIES

AVI mini monitor

AVI's NuNeutrons compact monitors measure only 265mm high x 140mm wide x 205mm deep and are intended for wall or stand mounting. Available in cherry, black ash and ebony the company claims it has virtually eliminated phase anomalies in the crossover.

The monitors comprise a 5-litre rear-ported enclosure with a 5-inch doped paper, curvilinear, diaphragmed, bass driver with a 25mm voice-coil. The 28mm tweeter has a viscous damped fabric diaphragm while the second-order Linkwitz-Riley crossover uses large ferrite inductors and polystyrene capacitors on a 2oz copper PCB. Prices start at £499 UK.
AVI, UK. Tel: +44 1453 752656

ARX ambience

New additions to ARX's Ambience series include the AmbiDrive 3-channel integrated power amp with built-in electronic crossover, and ISC speaker processing. Channels A and B deliver 160W into 4Ω while the mono subwoofer output delivers 200W into 4Ω. AmbiSub is a compact subwoofer designed for enhancement of the Ambience 1 and 5 systems.
ARX, Australia. Tel: +44 181 742 0350

Apogee arrays

Apogee Sound has unveiled the ALA9 linear array loudspeaker that extends the concepts used in the ALA5 and ALA3 systems. It comprises dual 15-inch cone drivers in a slot-loaded configuration, dual 10-inch cone drivers loaded by two damped midrange horns and dual 2-inch compression drivers loaded by paired high frequency waveguides. The pairing of waveguides produces a vertical dispersion angle of 10° while horizontal dispersion is 60° and extends to the lower frequencies by the doublet principle.

The company has also introduced a battery backup unit for use with its MA line of modular, class-D switching type power amps. Called the BB1, the unit is housed in a 2U-high chassis and interconnects to the MA series amps via a standard Neutrik cable assembly. Battery life can be as long as several hours depending on loads and programme type. The unit is said to be smaller and more efficient than other designs in >

Studio Sound July 1998

Liquid Audio Liquifier Pro v3.0

The Internet is changing our conception of global music distribution. **Rob James** investigates an approach to the profits of Net publishing

THERE ARE SIGNS emerging that serious money can be made from e-commerce, and not just by conning the gullible into investing in hi-tech startups. (An electronic bookshop is doing serious business and has made its owner very rich). Unsurprisingly, record companies large and small are taking an interest in how the Net could be used to generate revenue. As security for financial transactions is improving the missing components are sufficient bandwidth, a clear strategy and delivery model, means of discouraging piracy and the tools to prepare and deliver music to the eager punters.

One of the more promising and well thought-out delivery models, that takes into account the realities of current bandwidth limitations, is the Liquid Audio solution. The complete system consists of three components. The Liquid MusicServer handles the on-line distribution, including rights tracking and the financial transactions. The Liquid MusicPlayerCD software is freely available for downloading and enables customers to stream (play) material in real time; download material after payment; or to order CDs. The subject of this article is Liquifier Pro v3.0 which prepares the raw material, sounds, artwork and text information for publication by a LiquidMusic server.

Liquid Audio makes use of Dolby's AC-3 technology to encode masters for low bit-rate streaming and has added its own enhancements specifically for Net transmission.

Liquifier is available as a plug-in for Digidesign's Pro Tools on the Mac and as a stand-alone package on the PC platform.

The suggested sequence of operation is to first enter text (plain vanilla text files or type) and graphics (GIF only) in the various fields of the Media pane, then save as a Workspace file. Audio may be recorded and saved as a WAV file in the Record pane and edited in the Edit pane. The encoded audio is auditioned in the Preview pane. The final step in production is to use the Liquify pane to combine all the elements, text graphics and sound into the final package, a Liquid Master File. This file is published to a Liquid Music Server by uploading via the net. At which point it is available to customers. Before the publishing stage, a Liquid Audio Site Certificate must be downloaded from the Web site with the Liquid Music Server you intend to use.

In addition to recording audio via a soundcard (SoundBlaster compatible, I used a TripleDAT) existing WAV, AC-3, RA and PCM files may be opened; although audio can only be saved in WAV format. The software also has the capability to extract audio from CDs, but only Plextor drives are supported. Liquid Audio refers users to third-party software if they wish to use other manufacturers drives.

There is not space here to give a blow by blow account of the various recording and editing functions, suffice to say they are will all seem familiar to anyone used to PC hard-disk recording packages. However, the Preview and Liquify panes deserve closer inspection—the Preview pane allows the complete Song and-or Clips, defined in the Edit pane to be heard with data compression applied as it will sound when streamed at Internet data rates from 14.4 Kbps to ISDN-2. The product of this part of the process is up to five separate Preview Images which can be saved as part of the Workspace file and Preset instructions to the Liquifier as to what parameters to use when encoding at each selected data rate. Each of the five rate groups has a number of factory presets. If none of these fit the bill you can design your own and save them for future use. Selecting a new user preset opens a window with six blocks showing the groups of parameters that may be adjusted—the first allows selection of mono or stereo. This is followed by a choice of sampling rates plus the option of 'faster' or 'better' conversion. The next two blocks provide paragraphic EQ and dynamics. The penultimate block is Watermark—if desired the digital equivalent of a paper watermark can be inserted into the audio containing copyright information that is extremely difficult to remove. The last block gives access to the Dolby Digital parameters. To hear the effect of changes to any of these blocks it is necessary to do another Preview.

Once all the required data has been entered, the audio recorded or loaded and edited, and the encoding parameters set there are two options for actually 'Liquifying' your work. The Liquify pane will process one Workspace file at a time or the Batch Liquify pane does as the name implies. The Liquify pane allows control over which data rates are to be included in the Liquid Master Output file together with options, where appropriate, as to whether each data rate is to be enabled for free download or secure-paid download. The latter option will encrypt the file.

Once all this has been completed, hitting LIQUIFY starts the actual encoding process, after which the encoded file can be Verified and then Published, which involves Binding the files to the users Site Certificate and uploading onto the server.

Liquifier Pro provides a comprehensive toolkit for preparing material for Internet distribution. It is equally applicable to a small company with one person carrying out the whole process or a large organisation where a number of people may be involved in the various stages of production. Good organisation and planning are required, as with any manufacturing process. To get the best of the audio-processing parts of the package requires expertise and practise, and should be seen as equivalent to the mastering process for other delivery systems. ■

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Lexicon PCM 91

Long regarded as the masters of digital reverb, Lexicon is aiming to secure the accolade with its latest processor. **Dave Foister** reports

SPECIALISM IS A DIRTY word—for some at least. Lexicon, however, clearly doesn't share this view; Lexicon equals reverb, and like an actor who knows that type-casting means he'll always work, Lexicon is happy to stick to what it knows and carry on doing it at least as well as anybody else. As a result, its reverbs are highly sought after in any area of the market.

Two new models are out to reinforce this position. The PCM 81 is an all-singing, all-dancing multieffects box with reverb the strongest of its many suits, while the PCM 91 here is dedicated to reverb pure and sim-ple—not that Lexicon reverb is ever simple. The variety and complexity of what's on offer is amply illustrated by the 450 factory pre-sets and space for 100 user programs. By way of acknowledging the extent of the available range, there are particularly smart ways of finding your way around this enor-mous palette.

The PCM 91 has Lexicon written all over it, figuratively as well as literally. The 1U-high front panel has the usual colour scheme and the usual way of getting at what's inside, and a familiar blue-on-black display screen that has rather more information on it than earlier Lexicons. In the unit's basic state this shows the name of the currently-selected preset on the lower line, and the broad group it belongs in above. It also has plenty of room for dis-playing editing parameters and their values, which again have the extensive scope we have come to expect.

Full compatibility with the outside world is ensured by the addition of both digital in-out formats to the expected ana-logue connections. Analogue inputs are on dual-purpose Neutrik XLR-jack connectors with associated level switch, while outputs are available to both standards simultaneously on separate connectors.

Operation of the PCM 91 depends on a small control panel of 12 keys, navigating around the possibilities in the expected ways. Programs are organised as 9 banks of 50, and recalling one of them involves step-ping through banks and programs and hit-ting LOAD when the desired patch is found. While others try to minimise the taken to switch from one preset to another, even mor-phing between them, Lexicon sees no rea-son to change its familiar lag when programs are changed; even manual changing invokes a significant Mute period before the new sound kicks in, and MIDI changes are sim-ilarly slow. No doubt we are meant to regard

this as the price one has to pay for this cal-ibre of reverberation.

For this is indeed reverberation of the high-est calibre. Basic reverb types range from halls to plates to rooms, and there is a wide selec-tion of specialised studio effects as well, all essentially reverb based. There is probably more variety among any hundred of them than in most smaller boxes' limited presets, with the expected OTT flexibility in the editing of any one of them.

Two editing modes allow the choice of adventurous access to the real nuts and bolts—and some effects can have dozens of vari-ables—or simplified control of the essential character of a program without getting bogged down. Editing parameters are arranged in a matrix of rows and columns; UP-DOWN but-tons switch between rows while the SELECT knob chooses a parameter within it, and the ADJUST knob does the rest. The neat bit is that the full Pro mode allows access to all of this, while Go mode gives only the top row, which is carefully selected to give quick direct con-trol over the important functions. One key function is always live on the ADJUST knob when a preset is loaded, and this can even be a patch consisting of multiple parameters. This kind of organisation is possible in user pre-sets as well, as is the use of Keywords for sort-ing. Up to four Keywords, such as Acoustic, Dark, Dialog, and so on, from a list of 50 can be attached to each preset, and the words can then be used to nar-row searches, only offering those that match.

The fact that this is neces-sary at all is an indication of the sheer power of the PCM 91. This, of course, is the key; even if the programs couldn't be edited at all, this would be worth having for the quality and range of its sounds. In the time available for review I couldn't even lis-ten properly to all the presets, never mind explore the editing possibilities of all the algorithms, and I suspect with this much horsepower available few owners would ever become fully conversant with what it can do. Some intriguing things popped up on the screen; a Lecture Room preset has as its main Adjust function Attendance, con-trolling the number of virtual people in the virtual room—clearly a patch to several para-meters simultaneously. This kind of detail, clearly presented (although the manual is not great), coupled with some of the best reverbs you'll find, are the trademarks that have put Lexicon where it is, and the PCM91 should help keep it there. ∎

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< not wasting power in stepping up the line voltage.
Apogee Sound, US.Tel: +1 707 778 8887

DVD-R dupe
Hoei Sangyo and Microboards Technology's DSR-8000 is being heralded as the first DVD-R duplicator capable of copying DVD-R authored discs. The eight DVD-R drive exter-nal configuration can burn eight DVD-R discs simultaneously within 54 minutes.

The master-slave configuration, direct SCSI and dual functionality are said to make the machine an economical hybrid solu-tion for CD-R and DVD-R duplication. The DSR8000 has three Versatile Media Interface channels and when configured as a DVD duplicator each VMI slot card supports two external DVD recorders.
Microboards, US.Tel: +1 612 470 1848

New EDACs
EDAC's 521 series of connectors are avail-able with 28 hermaphroditic Edacon con-tacts along with six 75Ω or 500Ω contacts or six power contacts each capable of 40 amps.

The design permits the mixing of differ-ent types of connector requirements which could include eight audio channels and a six component video channels with 75Ω coax contacts. It would also allow for two pairs of edacon contacts to be used for sync, control signal or parallel filtered power up to 16 amps.
EDAC, US. Tel: +1 416 754 3322

24-bit SC3
The Joemeek SC3 compressor is now avail-able in 24-bit form as opposed to the orig-inal 20-bit. The new version of the VC3 Pro Channel has dual high level balanced out-put and modified compressor drive chain. The revised unit has new packaging and a black 'retro' front panel.
JoeMeek, UK. Tel: +44 1626 333948

UK amps
The UK Power Series of power amps from Matrix Audio Developments are targeted at fixed installations and mobile use, and are housed in shallow rackmounts.

Two-speed thermistor-controlled fans, short-circuit protection, DC sensing active clip limiter and output relays are standard. The range is complemented by the M-series amplifiers with outputs of up to 400W.
Matrix, UK. Tel: +44 1562 883116   >

TC MasterX TDM Plug-in

Derived from the successful Finalizer, multiband dynamics processing is now available to ProTools. **Dave Foister** goes for the soft

TC ELECTRONIC did not hang around joining in the TDM party. TC Tools, with its TDM versions of the company's acclaimed reverb, chorus and effects algorithms, has been around for some time, reflecting tc's digital roots and bringing familiar and desirable processing to the environment. More recently tc's Finalizer has scored major success in its role as a glorified Better knob, and the heart of that elaborate hardware processor is now on offer to the TDM world as tc MasterX.

The Finalizer itself incorporates a whole string of possible processes, from 5-band EQ to gain normalising to stereo image adjustment, but its special feature is its multiband dynamics processing. This central element is what TC MasterX gives us, with all the functionality and even some of the same factory presets. It comprises a compressor, a limiter and an expander, all stereo, and available simultaneously; the crux is that all three processes operate in three separate frequency bands, with independent gain control over each band. The result could be an almost bewildering range of adjustable parameters, but the user has the choice as to how deeply to delve into the possibilities, with substantial rewards available at every level.

As its name suggests, MasterX's role, like

> ## The more adventurous can get further into the system and adjust the crossover frequencies between bands

that of the Finalizer, is to process a complete stereo mix to add the final overall gloss. It is, perhaps, at its best when turning an anonymous mix into something louder, punchier and more memorable, but it has the power to be transparent and gentle when required, or to provide simple system protection without any other interference.

The key is the complete programmability of every element of the processes, and in particular the multiband operation. Compression and limiting of finished mixes are problematic jobs, more so than dealing with individual instrument tracks; big peaks in one part of the spectrum —thumping kick drum, for example—can modulate the entire mix, producing all too familiar pumping and sucking effects, and undermining the initial intention of bringing the overall perceived loudness up. Sometimes, taken to extremes, this is just what is required, locking the meters in place and saturating with sound. More often it achieves the opposite effect to that intended, making a mix sound weak and lacking in punch. Frequency conscious compression is not a new

idea, but it is not often taken to the lengths tc presents us with in this package.

In its simplest form each element in the chain can be treated as a conventional single-band compressor, limiter or expander, but when it starts working it always treats each band separately. Thus an overall threshold can be set in the compressor, and across-the-board time constants and ratio, but each band only compresses when its own contents cross that threshold and only to the extent that its own signal controls it, without reference to the other two bands. This means that the kick drum can thump away as much as it likes, without affecting the contents of the middle or HF bands, and the cymbals can crash without modulating the piano. The result is real increased loudness without the drawbacks.

The more adventurous can get further into the system and adjust the crossover frequencies between bands, and the inter-relation between their operation. Here the computer screen comes into its own to allow MasterX to use a few tricks denied the little Finalizer display. A single window allows crossover frequencies to be dragged around, and compressor gains to be adjusted within each band; as elsewhere, all the variables can be controlled using any of the familiar methods, from numeric entry to mouse movements, with a fine trim mode invoked with the Command key. Shift-clicking allows pairs or groups of adjustment to be linked for simultaneous adjustment.

Another view not available on the Finalizer is a set of curves showing the overall dynamics curve for each band, with 'LEDs' showing limiter and expander activity and a gain reduction meter for the compressor. Clicking on these displays can even solo the bands to hear precisely what they are doing.

Where the Finalizer makes you tweak the behaviour of each band individually to shape the response of the processors, MasterX provides a much simpler way using Target Curves. This is not EQ, but a bias as to how hard each band will work and what its final gain will be. The result tailors the sound in a choice of four ways: linear; Hyped, which cranks up the top end; Pink, which rolls it off; and, would you believe, Smiley, which increases the extremes in the manner of a loudness control. For all of them, the extent of their effect is adjustable as Target Focus, which controls how much the curve will influence the processing.

Metering is good and includes Over monitoring; since the plug-in is designed to maximise the levels from the source, a useful function is the digital ceiling, which prevents the signal rising above a threshold.

I have found the Finalizer invaluable in adding the finishing touches, subtle or radical, to overall mixes, and the bit that gives it the edge is right here in MasterX. ∎

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PC preamp

Rocktron's PC Preamp is an all-in-one interface for sources to be connected directly to a PC with a sound card. The 2-channel preamp has a distortion tone with HUSH noise reduction, clean tone with compression and a spring reverb. There's also a stereo effect loop and a stereo aux input. It has been designed to be placed beneath the computer monitor to look part of the system.
Rocktron, US. Tel: +1 248 853 3055

Test set

A new, lower cost version of the Leader L5100 component-HD waveform monitor is now available. Designed for TV and video measurements, the new instrument can measure the level and phase of an analogue

component video signal or the level of a composite video signal. It can be used for analogue component signals (525/60 and 625/50) and HDTV signals (1125/60).
TTI, UK. Tel: +44 1480 412451

Six-channel amp

The CP660 6-channel power amp from Crown provides independent channels in 2U of rack space. With 75W output per channel it is suitable for surround applications as well as paging, zone and background music applications. Pairs of channels can be bridged for double the power. Controls and connectors are mounted on the rear panel and a quiet variable speed fan provides cooling.

Crown IQ for Window 3.0 features enhanced custom controls that allow the user to create control panels designed specifically for each application. The upgrade allows scheduling of dataframes and scenes and for those who want hands-on control it offers a Scenes sequencer to choreograph a series of complex events. Crown has added an Administrator Password for global access. IQ for Windows 3.0 expands on Crown's IQ NET with the introduction of a Chat Utility which allows users to communicate with all others on their local IQNET.
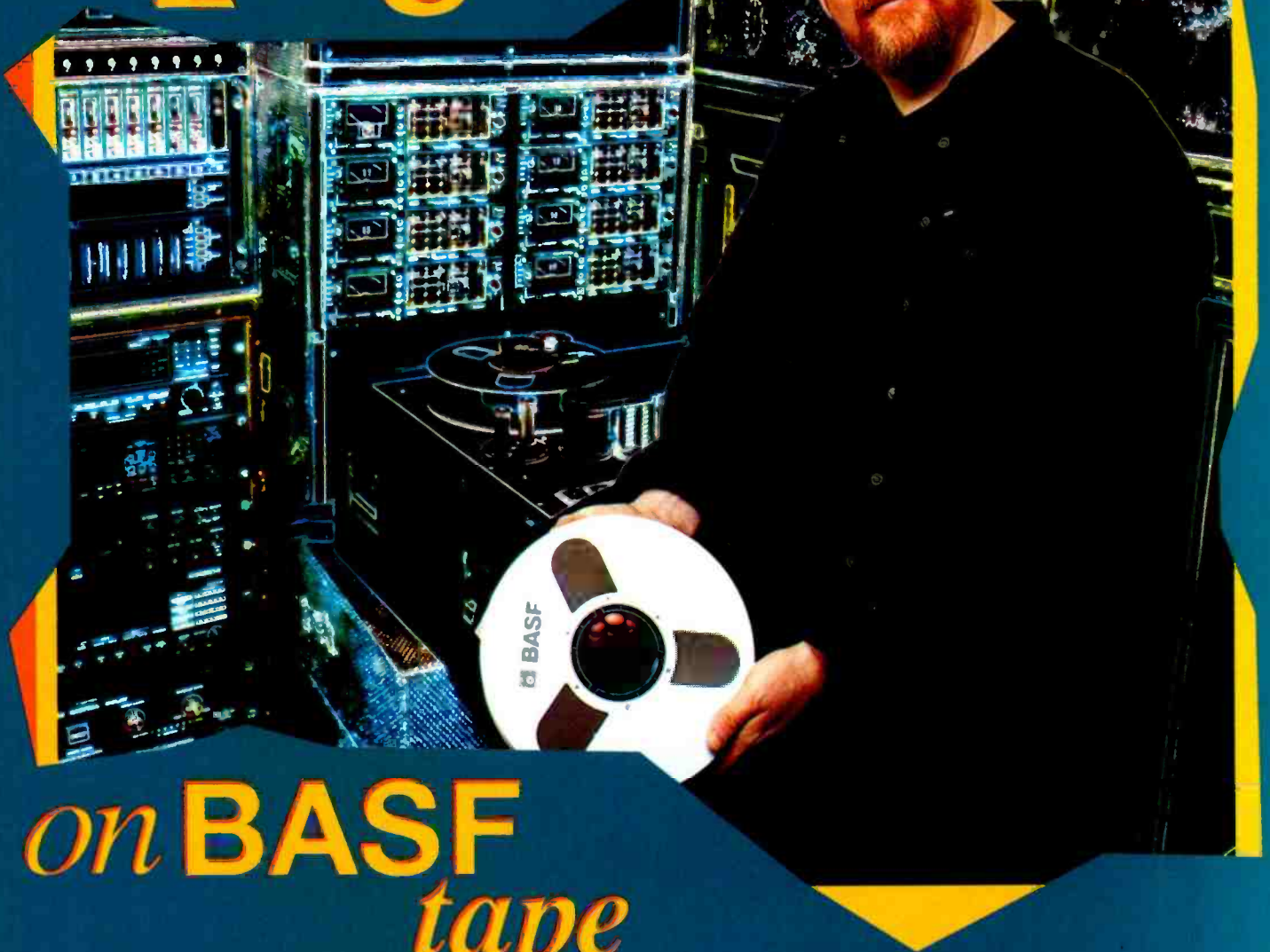**Crown, US. Tel: +1 219 294 8208**

Eventide DSP4500

Eventide's DSP4500 includes all the presets from the DSP4000 standard, guitar and broadcast versions plus the Alchemy 101 package of 225 third-party presets. It has an 87-second sampler included and at more than 1000 presets is claimed to offer the >

Mike Hedges

on BASF tape

"When we changed to SM 911, 7 or 8 years ago, we got that sound back. It has a really good musical edge.

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

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

STUDIO MASTER 900 *maxima*

SM 900 maxima is a high-output analogue tape designed specifically for multi-track recording and mastering, with extra wide dynamic range, low noise and low print through.

EMTEC Magnetics

For more information contact +44 01295-227838 or visit EMTEC Magnetics' web site at http://www.emtec-magnetics.com

BASF

Iomega ADSG drives-media

The goal's of ADSG lie beyond developing new digital storage for the film industry in the defining of standards. **Rob James** goes for a spin

THE ADVANCED DIGITAL Systems Group is a division of Sony Pictures charged with the mission of 'Developing technologies for film and digital playback systems for theatres'. One of its projects is developing digital storage systems for all aspects of the film-making process, of which the first tangible results are now available in the shape of ADSG professional drives and media which are being touted as the solution to the perennial problem of finding a digital storage medium that can achieve the ubiquity of 35mm magnetic film. ADSG has taken the Iomega 2Gb Jaz drive and done work to optimise it for AV use—I am told that this includes changes to the low-level firmware. Both media and drives are checked and certified by ADSG for professional use and each disk will be uniquely identified to facilitate project management. ADSG's confidence in the concept is demonstrated by the choice of the drives for its new DADR-500 digital dubber, and it hopes ultimately to achieve worldwide acceptance of the disk format. Ambitious stuff.

I carried out some comparative tests on the ADSG drive against an IBM 4.3Gb fixed SCSI drive. This IBM is no ball of lightning;

it is a 5,400rpm spindle-speed device and is pretty pedestrian compared to the current crop of 7,200 and 10,000rpm devices. First a warning, these figures should not be taken as gospel. There are far too many other variables involved, such as the capabilities of the PC and SCSI adaptor, to treat them as absolutes. Rather, their value lies solely in the comparison.

For my initial test I used Adaptec's SCSI bench software which enables you to get some idea of a drive's performance, but bears little resemblance to real-world audio use. In the least meaningful test, reads from the same disk sector, the IBM was at least twice as quick as the ADSG. In sequential reads the IBM was merely a shade quicker and in random reads about 25% faster. To get some idea of real-world performance I used Creamware's Triple-DAT. First, I recorded a number of cues at 48kHz, all around one to two minutes in duration, onto the ADSG drive. These were then copied in one operation using Windows Explorer to the IBM. To keep things as fair as possible I removed the original files from the ADSG and then copied them back from the IBM, also in one operation.

With identical settings in TripleDAT I man-

aged to get 18 simultaneous stereo tracks to play reliably for 1 minute from the IBM and 11 minutes from the ADSG. Again I would stress these are purely empirical results and several other factors govern the total number of tracks to be expected from TripleDAT or other software. The sole useful conclusion from this is the ADSG is not as fast as an 'economy' SCSI drive let alone a current generation AV drive. There is an argument that this does not matter provided the performance is adequate in the specific application. I do not agree.

Operating with a drive close to its performance limits means fragmentation becomes more of an issue and the safety margin is reduced. As a disk becomes fragmented with use the performance declines. Defragmenting can be designed into a DAW to happen without the user initiating it, but the process takes a finite amount of time and can therefore slow down the responses of the machine. A faster drive can clearly cope better with performance lost through fragmentation.

For pro-audio purposes digital storage is at a crossroads. Until now the issues have been, 'Can we get enough record time and tracks per disk at a reasonable price?' The inexorable progress of computer development has all but solved two of these three problems. The issues will then become price per Gb and, 'Is everybody else using the same form factor?' Plus longevity, reliability and support.

There are a number of other technologies competing in this area. Most notably LIMDOW Light Intensity Direct OverWrite optical drives. In their latest form these will have approximately the same kind of performance as the ADSG. The drives are however, more expensive than ADSG but the media is considerably cheaper. Fixed disks are still falling in price. The IBM I used for comparison is actually cheaper per Gb than ADSG disks. On the other hand, fixed disks are impractical as a transit medium. In my experience people will pay extra for the assurance of quality and reliable support, but not that much extra, say 25% maximum.

It is my belief ADSG has the right ideas about standardisation, certification and project management. I hope it succeeds, but I am not sure they have backed the right contender for the actual medium.

Much has been written about digital audio interchange in recent months. I am optimistic the situation may at long last improve. But, even if interchange at the file-format level is sorted

out that still leaves the problem of physically moving projects from machine to machine. Networking will (eventually) answer most of this but hard copy storage will always be required. Meanwhile... As a consultant and ex-facility manager I am well aware digital storage was and is a major headache. Near universal acceptance of one form factor of storage would do much to relieve it. ■

< greatest number of presets on any effects processor.

A total of 147 distinct effects modules can be built in to programs and custom programs can be saved to removable PC cards which can also be used for importing third-party programs.
Eventide, US. Tel: +1 201 641 1200

Cable reels

Hannay offers a range of cable reels for recording studios, film, TV and radio and live applications. AV, AVX and AVC series reels are lightweight and have carrying

handles. All models have a non-reflective matt black finish and are stackable. Removable side panels offer customisation of XLR and BNC connector patterns. AVX models have trays for storing stage box units and some have adjustable friction brakes to prevent cable over-run.
Metool Products, UK. Tel:+44 115 9225931

Multichannel monitoring

Adgil Design's Director is a programmable microprocessor-controlled system for monitoring in multichannel environments. It can handle one main monitoring system with eight output channels and two stereo auxiliary systems.

Functions include mute, dim, mono and solo, individual cut on speakers, preset monitoring levels, and variable levels. The chassis is built around a motherboard that provides interconnection between various plug in cards with DB25 Tascam pin-out terminations. Plug in modules include and input module, a bus amp-insert send module, insert return-mono module, output module, comms module and matrix input card. The Director consists of the rackmount chassis and a remote control.
Sascom, US. Tel: +1 905 469 8080   >

Focusrite Platinum Tone Factory

The first of two new Focusrite processors offers an extravagant tour of total possibilities. **Zenon Schoepe** buys a ticket

THAT FOCUSRITE should choose to aim the first two boxes in its new Platinum entry-level series of outboard processors so blatantly at specific user profiles bodes well for the new range. Placed bottom of the stack, and substantially below the still not quite affordable enough Green series, the Platinums weigh in with the Tone Factory instrument recording channel looked at here and the Voice Master (p50) vocal recording channel.

The Greens were a distillation of ideas that already existed in units like the Reds, but the Platinum approach is altogether more specific and clearly targeted at users seeking high quality signal chains that go to hard disk. Admittedly the Greens did also address this territory, but these cheaper boxes target in a more specialised manner, and do so by combining diverse blocks of processing and less academic descriptions of what the blocks do.

An instrument-level input appeared for the first time on a Focusrite with the arrival of the Greens, but the Tone Factory has gone for this far more aggressively and could be regarded as an instrument channel first and foremost with mic-and-line capability thrown in as an extra.

It combines a discrete transistor input, with filter, opto-compressor, tone controller, parametric EQ and noise-gate sections culminating with a master OUTPUT LEVEL pot.

This may seem a cheap box, but you get an awful lot of pots and switches for the money, with each processing block individually bypassable. Rear-panel connections give an extraordinary range of possibilities with phantom-powered mic input; balanced line-input; and balanced and unbalanced line-outputs; complemented by compressor sidechain access; gate key and insert, plus, of all things, a guitar amp level output. This a rare inclusion, even on dedicated guitar pres.

Instruments are plugged into the front panel. A wide-ranging GAIN control works in conjunction with signal present and overload LEDS, and the input stage is far better equipped to cope with the spurious dynamics that an electric guitar can generate than the overly sensitive instrument input that appeared on the Green Focus EQ's back panel.

Sweepable high and low-pass filters can be switched into the gate sensing circuit, while the opto-compressor has THRESHOLD; RELEASE; and OUTPUT pots. Switches access two attack speeds and a 'hard' ratio compression setting. A Tone Controller block attempts to reproduce the sort of tone-shaping found on a gui-

tar amp with BASS, MID and TREBLE pots, the last of which is switchable to act as more of a high mid. It is preceded by an OVERDRIVE control with defeatable speaker simulation. It is not bad at all, and capable of some fine variation, but it is unlikely to fool a player used to having his flares flapped by a selection of cardboard cones on a regular basis. It will stretch to a fair degree of overdrive, manages excellent crunch tones, and very pleasing clean sounds that can be forced to the ragged edge of dirty through playing dynamics. I am sorry to admit this, but I did not think Focusrite could do this sort of stuff.

A 2-band parametric offers ±18dB over 40Hz–1kHz and 500Hz–20kHz with shelf-bell switching and two Q values on both. This section does most for the Tone factory's versatility beyond the instrument input as it is here that you will tune mics passing through. You are then into the noise gate with variable Threshold, switchable Release or HOLD pot, two attack-time constants and two attenuation ranges. It is all rather natty and clearly thought out to create a very flexible single channel.

Do not dismiss the box's mic handling abilities because while they are not as elaborate as the Voice Master Platinum unit, the Tone Factory is no slouch and it shares its mic pre front end. A touch of excellent compression—better I believe on vocals than it is on guitar—and some refined EQ on the broader bell-setting and you have a nice quality direct feed.

Keyboards also benefit from the treatment, particularly the overdrive section and its tone controls. Predictably, it is a corking bass guitar channel with the 2-band parametric able to step in and iron out any of the resonances that can occur on this instrument.

I could see Tone Factories bought in bulk and relegating so-so mixers to monitoring duties in conjunction with hard disk systems. If I have any reservations it is that you cannot alter the order of the processing. It is not a major point, but being able to reverse the tone, EQ, and compression blocks, would have extended the flexibility even further.

I am not sure whether all-out guitarists will be smitten by this unit, but I would expect that occasional guitarists, and the project fraternity is filled to bursting with these, are likely to pay attention to the Focusrite branding and will appreciate some of the pure recording features that this box offers rather than its abilities as a pure guitar channel. There are guitar pres that perform better, but none are as optimised for the recording environment to the degree that the Tone Factory is.

A perfectly workable device and a bold move by the company. It sounds good on instruments, but general mic processing duties are also well within its abilities. There is not much else that attempts to address this task with this sort of conviction. And what a price. ■

Contact
Focusrite,
Lincoln Road,
High Wycombe,
Bucks HP12 3FX, UK
Tel: +44 1494 462246
Web: www.focusrite.com

Sanken shotgun
Sanken's CS3 short shotgun mic follows on from the CSS5 stereo shotgun and features a low-frequency roll-off switch in which case it can operate as traditional shotgun or with increased high-directivity in the low frequency ranges. Measuring 27cm in length it maintains sharp directivity by a combination of a second gradient mic and a line mic. Performance is attributed to an array of three new directional condenser elements. The poly-phenylene sulfide diaphragms claim exceptional humidity-temperature stability.
Sanken, Japan. Tel: +81 3 5397 7092

White units
Using 32/40-bit floating point DSP, White Instruments Paramedic digital parametric
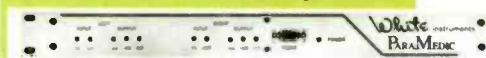
EQ offers 70 filters. These can be configured as parametric and very narrow notch filters, one-third and one-sixth octave graphic filters, and high pass, low pass and shelving filters. Frequency centres are adjusted in 1Hz increments and amplitude in 0.1dB steps. Paramedic Plus models add delay to the package. Servo balanced I-Os are standard but transformers are an option.

White has also introduced the single channel 4700XL and dual channel 4700-2XL digitally controlled one-third octave graphics. Control is afforded by RS232 connection to a PC running suitable software. Also new are the 2-channel DSP5022XL and 4-output DSP5024XL digital signal processors again with PC control.

DSP and graphic EQs have been released for the Crown IQ system.
White Instrument. Tel: +1 512 389 3800

Nucleus monitors
British manufacturer TDL's Nucleus 2 close-field is a reflex loaded design with a tapered reflex port at the back. It uses a 130mm doped paper cone and a 19mm-soft dome ferrofluid-cooled tweeter with crossover at 3kHz. Sensitivity is claimed as 89dB at 1m

for 1W and frequency response 50Hz–20kHz.

The Nucleus 3 floor standing loudspeaker is a reflex loading design based on the Nucleus 2 in a larger enclosure to extend bass performance. The separate lower chamber can be loaded with sand to achieve solid ground support. The frequency range is extended to 40Hz. Both models retail for under £200 in the UK.
TDL, UK. Tel: +44 1628 850111 >

Focusrite Platinum Voice Master

Another voice-recording channel, but this one carries the 'F' word and pretends it has valves in it. **Zenon Schoepe** prepares an audition

THE NEW PLATINUM units enjoy excellent construction with none of the design overkill that accompanied the arrival of the Green series. The sole concession on the Voice Master, like the Tone Factory, is a subtle, but tasteful subliminal legending of a large Focusrite logo on the brushed aluminium front panel.

Despite competitive pricing the Voice Master sports a phenomenal number of switches and pots for the money, but then it attempts

a lot. Dedicated to the provision of a single channel direct-to-tape recording path, predominantly for vocals, its skills extend well into the area of line input processing. Distinct sections provide an expander; a vocal saturator for valve effects; an opto-compressor; EQ; and an opto-de-esser.

At the end of the chain sits a master output-level pot with corresponding output-level meter. This passes out through two balanced XLRs—for pre and post de-esser—and an unbalanced jack output. Inputs are provided for mic on XLR and balanced line on jack plus a TRS insert.

You kick off with the same discrete mic preamp as that found in the Tone Factory with sweepable high-pass filter that goes up to 320Hz, phantom power, phase reverse and line selection. Gain ranges from 0dB to +60dB with signal present and overload LEDs. The expander section offers expander and gate modes working together with variable threshold and depth pots. Each section can be bypassed separately.

What Focusrite calls a vocal saturator section is said to emulate valve and tape saturation effects. This it attempts with a drive pot—graduated from clean to unclean—that determines how much saturation occurs, and has a corresponding LED that glows longer the more processing is involved. A Tuning pot meanwhile adjusts the centre frequency of the saturation with a full-bandwidth switch making the effect broadband. What we are adding here is valve-style harmonic distortion.

The opto-compressor section is close to that found on the Tone Factory, but adds a treble pot for boosting compression-related high-end in the processed signal. Pots are provided for

the SL15 and SL30. The reflex ports are located on the side panels of the cabinet to prevent screen resonances and dust spots and the cabinets can be floor mounted or flown.

A complete Stage Accompany Super Space Saving system can be flown behind a projection screen in a space of only 28cm.
Stage Accompany, The Netherlands. Tel: +31 229 28293.

Philips suppression
When used together with dedicated >

threshold, release and output with switchable fast attack and hard ratio values. EQ is an altogether different matter with the naming of parameters reflecting their effect rather than their function. Thus Breath equates to a 10kHz shelf and Presence at 4.5kHz peak. Warmth appears to be a broad bell that can be tuned between 120Hz–600Hz, and an ABSENCE switch claims to remove 'harshness' and would seem to be a gentle voice-band broadish mid cut. Despite its enthusiastic claims the ABSENCE switch is pretty damn subtle.

The last block of processing takes care of de-essing and combines a nicely tuneable centre frequency pot covering 2.2–9.2kHz, threshold pot and a 'something happening' LED.

This is a powerful little channel's worth of processing that belies its cost.

There are some peculiarities which become apparent when trying to balance the channel with most of its constituents in place. This comes down to a matter of comparing the contribution of sections. The order of processing is not quite as the front-panel flow would suggest because the vocal saturator is actually post compressor. This becomes apparent when you whack up the saturator circuit and set the compressor in a way that lets the odd blast through as the former section's overload LED starts up. It is one you have to watch, but max valve simulation circuit followed by floored opto-compressor does equal king retro mic sound. And you can tune it, so you only get the effect in part. It is very good with shades of the excellent Studer Multivalve.

With the price it is a shame that some Voice Masters will only see dynamic mics, but the unit's performance on these is quite remarkable, and certainly nudges the results well into the realms of acceptability. You can give a dynamic mic more backbone, definition and class with this channel, but it is still a waste of its talents. The better the mic the better the results, and while it is too far fetched to suggest that it can make a dynamic sound like a condenser, or to add virtual valves to condensers via its processing sections, there can be no denying that the preamp front end is extraordinarily good. It is wide, clear and clean, and out of this price league. What the channel does from then on is allow you to contour what goes in gently—there is no really dramatic hoodlum-shaping available on the Voice Channel. You are offered restraint. The compressor is probably the most brutal block, but even this is intensely musical on the limit, the vocal saturator does not get offensive at full throttle, and the EQ encourages you to work a band at a time emphasising treble, pulling up definition and adding or subtracting thickness. And you can apply this to the miking of instruments, such as acoustic guitar, just as comfortably as you can to vocals.

Its a very, very good voice channel. Best of all it is a bit different from the rest, and knocks the tastefully mottled spots off its similarly price competition. ■

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Net: www.focusrite.com

Adaptec Toast CD-DA

With burning CDs becoming increasingly popular the need for a pop-up solution grows. **Dave Foister** lands butter-side down

EVERY SO OFTEN the facility to do something that was previously the reserve of the specialised professional becomes so widely available that the profession has to reassert itself. Affordable and increasingly-sophisticated home recording had a profound and well-documented impact on the recording industry, and since then people have begun to think they can master their demos and produce their own CDs, aided and abetted by the hardware companies' willingness to slash prices until they are giving stuff away. It's now common for a home PC to have a CD writer in it, and already these are by default re-writable ones—the write-once variety is becoming hard to find, to the despair of pros

for whom certain writers stand out as doing the dedicated audio job properly. What we all hope of course is that it never gets to the stage that knowledge and understanding of the process becomes unnecessary, and that there will still be a need for us, our skills and our equipment.

Such as a big Macintosh and a rack full of Digidesign hardware, running a whole drive full of expensive software to create the whole product, through tracking to mixing to post-production to mastering to CD burning, all in the one virtual studio. For many people this is now the natural way to work, and the better the final mastering stage integrates with the rest of it the faster the job will be. Despite Digi having its own CD preparation software, considerable success has been achieved by Adaptec with its Toast package, which is available in a couple of different forms for different areas of work. One is CD-ROM compilation, with a simple option for sticking bits of audio together for writing, but for the audio specialist Toast CD-DA is the one to look at.

Toast assumes that the basic preparation of individual tracks for a CD has already been done, producing audio files in one of several possible formats. Digi's position at the forefront of Macintosh work is reflected in the strong support its various file formats have elsewhere, and Toast is no

exception. Sound Designer II files in particular are catered for, along with the program's playlists and markers, but split stereo files are also acceptable, of the type used by Pro Tools among others. Broad compatibility is assured by the ability to import AIFF and WAV files, and Toast's independence is further underlined by the fact that it does not rely on the Digi hardware—it can do its job with nothing more than a Mac and a CD writer.

The main window for Toast has a panel at the top containing track and time displays as well as transport controls and buttons for adding and removing tracks. The main area is the playlist, showing all the relevant information about the assembled tracks and allowing editing of the important parameters.

Standard file menus can be used for importing audio tracks into this window, but Toast also makes extensive use of drag and drop, where any audio file can be picked up off the desktop and dropped into the playlist. Great flexibility is available here; the files for the various tracks do not have to be all of the same format but can be mixed. As long as they are all at 44.1kHz, any combination of the supported formats can be used. Again, Sound Designer II is particularly well provided for, with options to import whole soundfiles, individual Regions, or finished Playlists.

The tracks immediately appear in the order they were imported, spaced by a user-adjustable default gap. Changing the running order is a simple matter of dragging them, individually or in groups, around the screen, and the gaps can be adjusted. Particularly powerful is the ability to control crossfades between consecutive tracks, which comes with a palette almost as extensive as that in Sound Designer itself. Tracks can be made to overlap with crossfades of various shapes and lengths, or a track can have fades up and down added to it—assuming there is more audio available outside the marked ends of the track. Crossfades are rendered rather than real-time, and can be auditioned using the transport controls like any other aspect of the CD.

The sadly under-used Index points can be set and edited within a track, and if a SDII playlist is the source then regions within it can be used to mark the indices. Up to 50 points are available within each track. There is a field for entering ISRC codes for each track, and a set of headings to identify the project and various items of information about it, is provided to help prepare a final PQ printout.

If a lot of this sounds pretty obvious, that's because Toast is set out as a pretty obvious and straightforward package. Although it does much more than simply bolt files together, it is all so easy to find that it looks more basic than it is. Its flexibility and its sharp focus on the requirements of the job make it easy to understand why it's already such a success. ■

< Acoustic Feedback Suppression software, the Philips Profecta audio processing platform can eliminate audio feedback

in a variety of venues. Under certain acoustic conditions up to 20dB of gain above the level of howling can be attained, claims the company.

The Profecta is a modular digital signal processing system that can be inserted into any sound-reinforcement installation between the preamp and the power amp stages. It runs a variety of application-specific software that can enhance sound quality, remove unwanted acoustic effects or to improve the performance of the equipment. It has its own processor, internal non-volatile memory and a menu-driven interface with front panel keys and backlit display. The software is downloaded from a PC to enable Profecta to carry out the desired processing functions. Hardware options include A–D and D–A convertors, a digital interface and RS485 or RS232 interface.
Philips Comms and Security, The Netherlands. Tel: +31 76 572 1559

Harrison Surround

Harrison has demonstrated the SMS Surround Monitoring System for multichannel film, video and DVD mixing. This computer-automated system may be retrofitted to any mixing console with at least 8 bus outputs. Its operation is controlled via an Interactive Touchscreen Interface, and permits mixing in LCRS, 5.1 or 7.1 surround formats. Film-style or music recording-style monitoring interfaces are included and an unlimited number of monitoring setups may be saved and recalled.
Harrison, US. Tel: +1 615 370 9001

AKG updates C1000S

AKG has updated the 11-year-old C1000S. The new version still has a Polar Pattern convertor but adds a Presence Boost Adaptor. In cardioid mode this provides a 5dB peak between 5 and 9kHz. Cosmetically the new model has a satin-silver finish and an improved battery compartment.
AKG, Austria. Tel: +43 1 866540

www.americanradiohistory.com

Radio activity

Like an out-take from *The Conversation*, four location recordists gossip about radio mics and the problems encountered when using them on location

WHEN TWO OR MORE sound recordists are confined together you can bet it will not be long before the topic of conversation moves onto radio mics. Admittedly, it is usually by way of, 'Are there location caterers here?', 'What time do we break for lunch?', 'Do they put out those really nice pink wafer biscuits for us with the coffee urn?', but believe me, it's a cert. It is not easy for us journeymen, but we do take comfort in our mutual criticism of these wireless wealth-easers. Ever mindful of the assertion that critics are like harem eunuchs—they see it done every night, but can't do it themselves—four freelance sound recordists huddled around a virtual trestle-table set in a bleak virtual field drinking an unlimited supply of location coffee, the taste of which is always universally bland and from a recipe known only to German-émigré rocket-scientists as a propulsion fuel. We spoke at length on our respective experiences with these tools of the Devil. There were virtual pink wafer biscuits too.

Neil Hillman: I use Sony 810/820 UHF transmitter-receivers with ECM 77 capsules, what do the rest of you use?
John Rodda: I use Audio Ltd's UHF RMS 2020 kits of which I own four complete systems. I have an Audio Ltd 4-way diversity rack that I have built into my drama trolley. I have a UHF hand-held transmitter that complements the system perfectly. I can attach any of my Schoeps capsules with active leads and often use it as a 'placed' mic in drama situations where it would be impossible to conceal a Sennheiser 415 on a radio link. Only very rarely do I use the hand-held as a presenter microphone. I also use the small 2-way diversity rack for documentaries. I'm particularly pleased with the slot in 'hot-shoe' design that automatically makes all necessary connections when the receiver is placed in the 4-way or 2-way rack systems. I have six or seven Trams, both black and white, because I hate to see the

mic cable through a thin white shirt or t-shirt—as worn and seen on Michael Caine in *Mona Lisa* for instance—and a couple of Sanken COS-11s. I find that between them you can cope with most combinations of clothing and the need to hide microphones on a contributor. For cordless booms on drama I'll use Sennheiser 415T or 815T microphones.
Barrie White: I use Audio Ltd RMS 2000 because, when I bought them they appeared to be more transparent than the others I was familiar with, by that I mean that there appeared to be faithful reproduction of the audio signal. Also they are British, and, therefore, I felt that servicing would be quicker if undertaken by the manufacturer.
John Biddlecombe: Audio RMS 2000s too, but on VHF, and with either Tram TR 50s or Sanken COS-11s. I've been really pleased with the after-sales service I've received.
NH: What other rads have you used? I owned both Micron and Audio Ltd before my switch to the UHF Sonys.
JR: I used to have 4 VHF Micron CNS transmitter kits until about 1989. These suffered from unpredictable 'bangs' that sounded like a distant door slam. I understand that this was caused by the design of the compander circuitry. In about 1990 I bought 4 Audio Ltd VHF RMS2000 kits. These consisted of the pocket transmitters, but had large receivers with a slide-on battery pack. I found the audio quality excellent, but felt the time had come to update in 1997 when the first UHF RMS2020s became available.
BW: I had experience of using non-diversity Microns when on staff at HTV, but I remember being none too happy with the audio quality at the time, and, of course, there was always problems with the RF. Using them with Sennheiser MK2 mics made an improvement to the audio over the supplied Sony ECM50s. I also tested the, then, new Sennheiser UHF models but they appeared not to have any range. No doubt there has been some improvement since then.

JB: My Audio's replaced a pair of Microns some years ago now.
NH: Does the general close-distances of presenters we encounter in single-camera/documentary work justify the added expense of diversity receivers?
JR: Yes, definitely. Diversity radio microphones overcome the major shortcoming experienced with non-diversity types—phase cancellation. The drop-outs we all used to suffer when a direct and reflected signal arrive at the receiver aerial at the same time are all but eliminated as the receiver switches automatically and silently to the receiver that is picking up the strongest signal. It doesn't matter what job you're doing, you don't want to have to ask for another 'take' because the radios have let you down.
BW: Diversity units are less prone to RF interference by design, but my radio kits are non-diversity because they were a lot cheaper than diversity when they were bought eight years ago. That was a decision I have regretted on more than one occasion because of the reliability of diversity over non-diversity and the difficulty in predicting their range or where they will or will not work.
NH: Does your heart sink when you are asked to work on radios?
JR: My heart sinks when an inexperienced producer-director does not realise that it's my decision to choose radio mics, not theirs. They should explain the shot and let us make the choice. Radios do provide a high degree of flexibility and when placed correctly, with the right mix of open microphone, can sound completely natural. I have worked in a number of very hostile environments over the years and like to hear the detail in the dialogue despite the thundering traffic, taxiing aircraft or whatever. None of this would be possible without a personal microphone on a radio link unless we were to severely restrict the shooting style of a production. On most documentary shoots these days I like to be able to radio my main contributors and know that I'll always have good sound from them as they walk and talk. In a quiet, static situation nothing can beat the sound of an open >

John Biddlecombe

Neil Hillmann

John Rodda

Barrie White

< microphone on a stand or boom.

BW: I'm not keen on using radios because they are prone to RF interference and generally the microphone quality is poor. Ineffective wind gags, clothes that rustle bother me, and when in place the mics can be masked by an arm, or a magazine or clipboard that the wearer is holding. Unless the microphone is mounted somewhere on the head then it is only too easy for the audio to be off mic when the head is moved. There is also the problem of the person who is wearing the mic touching it. It is so difficult to predict the range and where they will or will not work; here's an example, an exterior shoot at the Hay-on-Wye book festival. Maximum working range 25m, unobstructed line of site, the presenter was required to walk along wooden boarding covered by a canvas awning which was supported by vertical scaffold poles. As the presenter passed each vertical pole there was a glitch. I changed to a different frequency and cured the problem.

JB: Yes, that's exactly it, they can be just so unpredictable—one day you might 'hoof-in' a signal from 50 yards away, the next you are pushed to achieve 5.

NH: Do you ever use radio's as a Time-Code link or as a feed of mixer output to a video camera as we evolved to with the Sony 810/820 on BBC's *Antiques Roadshow*

JR: I use a VHF Audio Ltd RMS2000 for programme sound to camera on Beta and Digi-Beta shoots when we need to work without wires. I use either my Fostex PD-4 or HHB PortaDat in these situations to record simultaneously on DAT and transmit time code back from camera for sync via a time-code transmitter from Black Box Video .

BW: I use a radio link to provide a feed to video cameras, to assist editing, when recording on a separate audio recorder such as my HHB PDR1000TC DAT, but I would not like to use radios as the only link to a camera as you do on the *Antiques Roadshow*, I know that's a question of not tripping members of the public, but I prefer the security of leads.

NH: There just are hostile places to use radio mics...

JR: Heathrow Airport—750 various radio frequencies in regular use and transmitting a lot more power than any of my kit. The key is not to transmit programme sound on a transmitter that's too close in frequency to the receivers you're using on the contributors, if it's all intended to fit in the usual mixer or DAT carrying bag. Because of the proximity in frequency and short distance apart, you'll knock down the range of the contributors' radios to less than one third of their normal range. That's why I transmit programme sound on VHF while the contributors are on UHF.

JB: I can remember Neil and I working together in a Building Society's RF 'safe-room' and I tracked an artist wearing a VHF Micron with a half-wave dipole mounted on the end of a boom pole just out of shot.

NH: So what would be on your radio wish-list? I like the idea of a global CPU that could be easily reprogrammed on a PC by the local dealer, offering the opportunity to stay legal in whichever country you are working in.

JR: Something that sounds like an Audio RMS2020 and costs around £500.

BW: A hundred percent signal reliability, no RF interference and faithful audio reproduction. Good quality mics with non-microphonic cables and effective windgags.

JB: Good range, long battery life and a rustle-free mounting.

NH, JR, BW, JB: Would anyone mind if I had that last pink wafer-biscuit? ■

John Rodda was recently awarded a BAFTA for his sound recording of the BBC documentary *Airport*. **Barrie White** was the recordist for the 1998 BAFTA-winning film short *The Deadness of Dad*. **John Biddlecombe** is currently recording on location with the BBC *Vets in Practice* team. **Neil Hillman**, who has never won an award, ever, is the recordist for Bill Bryson's *Notes From A Small Island* travelogue, to be seen on ITV sometime in Autumn 1998.

Interviewing
Quincy Jones
purely on the
basis of his music
production credits
is almost an insult
to the man's talent
and achievements.
Richard Buskin
talks to an
American great

Right on Q

HE HAS WORKED for more than 50 years in the music business. He has a career that has encompassed the roles of artist; composer; arranger; conductor; record producer; film producer; TV producer; music publisher; magazine publisher; record company executive, and multimedia entrepreneur. He is the recipient of 26 Grammy Awards; the Recording Academy's Trustees Award; the Grammy Living Legend Award; an Emmy Award, the Academy of Motion Picture Arts and Sciences' Jean Hersholt Humanitarian Award; France's Legion d'Honneur; Italy's Rudolph Valentino Award and the Royal Swedish Academy of Music's Polar Music Prize. He was nominated for seven Oscars and a record-breaking 77 Grammy Awards; and awarded honourary doctorates from no less than nine universities and colleges across America. His name is Quincy Jones, and, in an era when many of his contemporaries are inaccurately described as such, he is the real thing—a living legend.

Purely in terms of his work as a producer and arranger Q—as he is affectionately known—is in a league of his own. Others have earned distinction courtesy of their landmark collaborations with, perhaps, one or two great artists, or with numerous big names during some peak years, yet Quincy Jones has managed to achieve the best of both worlds. Over the course of six decades he has consistently worked with many of the music world's most notable talents, while also immersing himself in a wide range of different musical genres.

From Frank Sinatra to Michael Jackson, Bono to Jacques Brel, Duke Ellington to Gloria Estefan, and Stevie Wonder to Ella Fitzgerald, he has journeyed from post-swing to high-tech, while creating his own musical hybrids along the way, fusing pop, rock, jazz, soul, hip hop and classical, as well as the styles and sounds of Africa and Brazil. During the mid-fifties he was the first popular conductor-arranger to record with a Fender bass. In the sixties he was the first black composer to be embraced by the Hollywood establishment. In the early-seventies his title music for the hit series *Ironside* was the first synth-based pop theme song, and in the eighties he produced, among other things, *Thriller*, the best-selling album of all time. Indeed, it is a testament to not only Jones' talent and ingenuity but also his determination and adaptability that he ever made it so far.

Born in Chicago on the 14th March, 1933, and raised in Seattle on the sounds of Clark Terry, Duke Ellington, Gil Evans, Charlie Parker, Miles Davis, Ray Brown and Dizzy Gillespie, Quincy Jones began playing the trumpet while in junior high school and at the age of 12 he was singing in a gospel quartet. His musical studies then continued at the Berklee College of Music in Boston, before he got the chance to tour with Lionel Hampton's band as a trumpeter, arranger and part-time pianist.

'As a kid I used to hang around the theatre a lot, and when I was about 14 years old I gave him an arrangement that I'd written for 'The Four Winds',' Jones recalls. 'There was a girl singer in the band named Janet Thurlow and she kept reminding Hamp about me, and so when I was 15 he wanted me to join the band. I went to get on the bus because I didn't want to take any chances that he'd change his mind, but his wife came along and said, "Get that child off this bus. Send him back to school." I was really bummed, you know. Then, two years later, while I was at school in Boston on a scholarship, they called me from New York, and I joined the band and stayed with them for three years.'

Jones' reputation as an arranger grew quickly, and by the mid-fifties he was arranging and recording artists such as Sarah Vaughan, Ray Charles, Count Basie, Duke Ellington, Big Maybelle, Dinah Washington, Cannonball Adderly and LeVern Baker.

'I've been arranging all my life—I used to teach Ray Charles the arrangements in Braille,' he recalls. 'In the beginning I used to do all of the be bop arrangements, and we didn't have any reference. There was only one radio show out of San Francisco and *Downbeat* magazine, and the rest we had to get through word of mouth in New York, through travelling musicians. Blues bands from California would come up and Dizzy Gillespie, Clark Terry, Count Basie and those guys would pass through, and we would be right on their butt, listening to everything that came out of there. There was no other way to communicate, there was no television.'

In 1957 Jones decided to continue his musical education by studying with legendary Parisian composer Nadia Boulanger, and to subsidise his studies he took a job with Barclay Disques, Mercury's French distributor. While in Europe he recorded Charles Aznavour, Jacques Brel and Henri Salvador, as well as Americans such as Sarah Vaughan, Billy Eckstine and Andy Williams. Many years later he would revive his European connection by way of co-producing the Montreux Jazz and World Music Festival.

'These days there aren't many arrangers left in the business,' he says. 'It was as an arranger that I got into producing records. You know, back in the early-fifties I would tell someone like Bobby Shad that I'd just met the best alto player I'd ever heard, and say, "You've got to sign him. He sounds like Charlie Parker. You should hear him." And he'd say, "No, I don't want to hear him. You write six or eight tunes for him, call up some horns, get a stu- >

< dio, get an engineer and I'll see you Tuesday". That's how I started; writing the stuff, conducting it, getting the band going and hooking it up with the soloist.

'At that time the A&R man would come in and say, "Take one", and after a while you'd realise that you were doing everything that a producer does. I didn't know what that word was when I started—to me he was the A&R man, and we worked for A&R men all the time. So eventually, after conducting for all of these singers across the country and recording them in the studio, I'd been prepared to become a producer.

'When I collaborated with someone like Count Basie I was dealing with a legendary style, in terms of Freddie Green playing 4/4 on acoustic guitar and just what Count Basie's rhythm sound was about. He was the King of Swing, you know, no matter what they say. These people have their own style and everybody knows what the ground rules are, but the arranger has to interpret that, be it Thad Jones or Ernie Wilkins or Johnny Mandel or Neil Hefti. You have to interpret what Basie's sound is and distil it through your own personality. That's what the collaboration is about; understanding what Basie is traditionally, all the way back to Lester Young and Walter Page.

'In my case I just concentrated on the kind of music I liked to hear Basie play. I base everything on that; what will turn me on. I know it will turn Basie on. Part of a producer and arranger's role is to love the essence of a human being, which is what their music is about. The same goes for Sinatra. I knew every note he could make and could hit, and every now and then I would try to push him past what I knew his limit to be. At first he'd feel like he was jumping without a net, but the love between us was so strong that there would be this total trust in each other. He'd know that I'd never embarrass him, and that if it worked it would only be better.

'The same thing happened with Michael [Jackson]. We added a lot of things to Michael, including an extra fourth on each end of his range, because when he was with The Jackson Five he was singing high bubblegum stuff all of the time, and so I had to bring him down. For instance, when we first did 'Don't Stop Until You Get Enough' I said, "Now, Michael, I would like to do the melody down in this register", and he didn't understand this until he actually heard it. Everything was in octaves and I needed him to do solo ad-libs down there to show the difference in the range. So, at first there was a little resistance, because he was 19, he'd been recording since he was eight years old and he'd got into certain habits. Every one does, especially if they're successful, not thinking that there's even more room to grow. Eventually they get it.'

'Some get a flying start in the talent sweepstakes: 'I met Michael when he was 12, Little Stevie Wonder when he was 12, Tevin Campbell when he was 12, Dinah Washington told me about Aretha when she was 12, they told me about Whitney when she was 16, and I first saw Patti Austin when she was four years old! She was awesome, and she still is. When they've got it that early, boy, you know that they're not playing around.'

While Quincy's 1961 appointment as vice-president at Mercury made him the first high-level black executive of a major record company, the middle of that decade gave rise to his three-year stint as conductor and arranger with Frank Sinatra. In turn this would team him with Count Basie for the *Sinatra At The Sands* album which contained the classic arrangement of 'Fly Me to the Moon'. Having thus acquainted himself with the Sinatra voice towards the end of its halcyon period, Q would subsequently get to produce and conduct 'LA is My Lady' in 1984. With the vocal sweetness, range and dexterity not what it used to be, the then 68-year-old singer nevertheless turned in some bravura performances on what turned out to be his last solo album, for, according to Quincy, his artistic instincts were still fully intact.

'He'd know when something was right,' he recalls. 'Inside he'd know. He'd look at me and say, "I guess you want to do another take", and I'd say, "No, not necessarily, but I think you do". You see, a guy like Sinatra you can't coach like an ordinary person, because he's such a legendary stylist. Your suggestions are minimal in that respect, just peripheral to the area in which he is working, and so at each stage you feel him out. You feel out what he's open to hearing feedback on—you have to be sensitive to that, and if he's not open to feedback then don't volunteer it.

'After all, when there was something that he was off on he was still hardly off, and he didn't go off too often. His instincts were awesome. He overdubbed bits of 'LA is My Lady' and he didn't like to do that, so he came back to LA and he wanted to take another shot at it, otherwise he wouldn't be happy. I'll never forget, as he finished the last take he knew he had it nailed, and he winked at me and he just said, "Well, I guess I'll unpack my bags and stick around here a little while longer". It was so cute. That just fell out of his mouth.'

Meanwhile, it was what fell out of Michael Jackson's eyes that Jones cites as an example when describing his work ethic of leaving in rough edges and 'searching for the truth all of the time'. Aside from *Thriller*, Jones also produced two of Jackson's other multi-platinum solo albums, *Off the Wall* and *Bad*, and it was a display of emotion during the former project which convinced the producer to retain the >

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< artist's feelings on tape.

'I had the song "She's Out of My Life" for about three or four years after it had been written,' Jones recalls. 'A lot of people thought it was for Sinatra, but I held onto it and then when Michael came along I thought it would be perfect for him, because he'd never sung that kind of mature song before. We recorded about 10 or 11 takes and he'd cry every time we'd end it, so I thought, "The hell with it Leave it in, let him cry". That's what he felt, you know, and I couldn't even appreciate how he was identifying with a mature relationship like that, because he was 19 or 20 years old. So I just left the tears and everything in there. I mean, if it's the truth, man, the truth is gospel and you let the truth come on out.'

One truth for black composers and arrangers during much of this century was that film scores were basically out of their reach.

'I wanted to do that since I was 15, but I couldn't get the shot so I waited until I was 30,' says Jones. Indeed, by the early-1960s he had established enough leverage for himself in the music business so that, towards the end of his association with Mercury Records, he could at last turn his attention towards the hallowed world of the movies.

His initial outing was on a Swedish

Mr Jones' regrets

● 'Whitney. When she was 16, Ashford & Simpson told me about her and I couldn't do anything because I was in the middle of the Michael Jackson mania.'

● 'Luther Vandross. He was on my record and you could hear his talent a mile away, but again I was too busy to do more with him.'

● 'Bobby McFerrin. He was also sent to me when I was in the middle of the work with Michael.'

● 'Marvin Gaye. He's my biggest regret. We were friends for a long, long, long time. We made personal appearances on the road and always talked about doing a record together, but I had no idea that he'd be leaving here before me. That never crossed my mind. I thought it would be the reverse. But I loved Marvin, man, and Sam Cooke too. He used to come to my office in New York when I was vice president at Mercury, and he'd sit there in his white suit and white shoes, and we'd talk and have a little lunch.'

film entitled *The Boy in the Tree*, and this was followed by Sidney Lumet's 1964 picture, *The Pawnbroker*. Since then there have been a further 33 major film scores credited to the name of Quincy Jones, yet the man himself asserts that he doesn't take a uniform approach to each and every project.

'Every film tells each different personality exactly what it wants to be,' he says. 'Thirty different composers will score it thirty different ways. Of course, you look at it and you deal with the technical restraints in terms of synchronisation, click tracks and so forth, and then, once you decide on what the tech-

nical peripheral boundaries are, you go onto the emotional side and go with the images, and every time it's different. Musically it puts you in a zone that you'd never go to with records. If you heard the [Oscar-nominated] score to *In Cold Blood* [1967] on a record you'd probably run out of the room. I had a housekeeper quit me when I was writing that. She said, "No more". She was fed up with murder.'

In terms of his own studio recordings, during the past decade Quincy has enjoyed widespread success with albums that have brilliantly melded the eclectic attributes of sharply contrasting talents. His landmark 1989 album, *Back on the Block*—named Album of the Year at the 1990 Grammys—brought such legends as Dizzy Gillespie, Ella Fitzgerald, Sarah Vaughan and Miles Davis together with Ice T, Big Daddy Kane and Melle Mel to create the first fusion of the be bop and hip hop musical traditions. Then, more recently, Q's Jook Joint showcased Quincy's ability to blend the performances of singers and musicians from Coolio and Phil Collins to Chaka Khan and Queen Latifah in a reference to the backwoods club houses of rural America in the thirties, forties and fifties. Now he's involved with a big band album that will utilise the talents of arranger Sammy Nestico together with music made up from standards and new material, as well as a compilation set entitled *From Q, With Love*, featuring many of the smoochy hit numbers that our Mr Jones has been involved with down the years.

'I'm very excited about that,' he says of *From Q With Love* that, in addition to his own name, will feature those of Sinatra, Basie, Jackson, Vaughan, George Benson, James Ingram, Patti Austin, Barry White, Luther Vandross and Aretha Franklin, among others. The inspiration for it was personal. I used to take this kind of collection on the road with me, because my girlfriend and I didn't want any fast tunes to interrupt anything.

'It includes an arrangement that I wrote for Frank of 'The Shadow of Your Smile', which he decided to put in his show when we were on the plane one night on our way to Vegas. I said, "Okay, I'll write it. Do we have the lyrics?" He said, "By the time you have the music I'll have the lyrics", and he took out a legal pad and wrote the lyrics down 18 times to plant them in his subconscious. I'd never seen anybody do that before. Frank was something else.'

Given Jones' experience in the studio, he intuitively knows how a session is progressing very early on, and then directs it accordingly. 'The trick is to know when it's working and when it's not working,' he says. 'If it's >

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< working then why isn't it working? I mean, if I don't know what's wrong, but I just don't like it I can't tell the musicians that.

'A major thing that is necessary for a good session is chemistry. People who enjoy working with each other have a sense of the sum being bigger than the parts, so it's a case of knowing how to play that kind of team game. Then there's the right material; it all starts with the right material. It's all about the song. You know, without a song there's nothing to talk about, and I think that half of a producer's job is to find the right songs from the beginning.'

Although occasionally hands-on in terms of studio equipment, Q confesses to usually having other considerations on his mind during a session. 'When we're really serious I don't want to think about that,' he says. 'I mean, you have so many options, and I have been spoiled by working with Phil Ramone and Bruce Swedien for the past 34 years.' They're the best in the world, and I'm never going to compete with Bruce's background regarding the technology because that's not my thing. Still, it helps if you understand it. If you don't understand it you can fall in and you're experimenting all of the time.

'There have been sessions where Bruce and I will just go in to play with toys in order to test their muscles, going through all of the things to find out what

they can do. I've seen everything that's been going on in the last 50 years, you know, and we started with 78s. I used the very first synthesiser on the *Ironside* theme and the very first Fender bass with [William 'Monk'] Montgomery in Lionel Hampton's band, and without the Fender bass there would be no rock 'n' roll. The electric guitar and bass are what made rock 'n' roll.'

As previously mentioned, what has made certain other musical fusions is the innovative talent of Quincy Jones. Still, when asked whether he views such fusions as interesting experiments or necessary advances when a particular genre of music is running out of steam, his response points towards an artist's improvisational feel for the creative art-form rather than any pre-planned intentions. 'Man, you know, it's just like eating a meal,' he replies. 'I like to mix up apple sauce with sliced bananas and peanuts and cookies and sliced peaches and roasted pecans to see how many tastes can be different yet compatible in one batch. I love that: the compatible richness. What I enjoy most is changing and doing different things, because a change is like a rest to me. I always try to transcend past standards and that's the hard part; I did this before, so how

**2: A childhood in Chicago South side.
3: On the road with Lionel Hampton.
4: Q with Duke Ellington in 1973**

can I do it different and better? That quest is really what it's all about, as well as breaking down musical categories. I don't like categories at all.

'Everyday I learn something new, and the older you get the more you realise you don't know. If you put a lot of energy into the beginning of your life and career, really grabbing onto the one thing that you love, then that's everything. I did that since I was 12. You know, I love writing music, orchestrating and arranging, and I've put a long time and all my energy into trying to be the best at that. I've worked with almost every pop singer and instrumentalist in American music, and what I've learned has provided me with the basis and discipline to approach anything else, be it film, television, magazines, whatever. After all, the principle's the same: trying to find excellence and having a real careful eye for detail. There may just be 12 notes, but a lot of great people have done a lot of great things with those same 12 notes...'

Most notably the acclaimed Mr Jones. After all, just consider where his professional life has led him. For one thing there have been his sorties into the twin worlds of film and television production, as marked by 1985's *The Colour Purple* and the launch of the hit NBC TV series *The Fresh Prince of Bel Air* in 1991. Two years later Jones and David Salzman merged their companies to form QDE, a film, TV and live entertainment production co-venture with Time Warner that has seen the production of such widely seen shows as the annual Academy Awards spectacular. In partnership with Time Life Ventures the company also publishes *Vibe* magazine. Then there's QD7, another Jones-Salzman coventure that, in alliance with multimedia publisher, 7th Level Inc, produces interactive multimedia titles; Jones' own Qwest Records, with an artist roster that includes New Order; Tevin Campbell; Milt Jackson; Tamia; Keith Washington; The Winans; Dori Caymmi and Justin Warfield; Qwest Broadcasting, which is currently one of the largest minority-owned TV broadcasting companies in the United States; and Quincy Jones Music Publishing, which boasts a catalogue with more than 1,300 titles that span five decades of music and, of course, several genres.

'It's wonderful and it's been an amazing life,' says the man of many moments. 'I mean, you only live one time. You have 29,000 days if you live to 80 and you've got to do it now, man. You can't say, "I think I'll wait 'til later and come back". I don't know what I'd do just laying around fishing for eight months a year. That's too simple for me. I had two brain operations, I almost died twice, so I don't look at life like that. I want to know I came through here... and really be sure!' ∎

Godzilla

The thunder of tiny feet

Big on action, effects and budget, Godzilla needs sound to match. **Richard Buskin** talks to the sound designers, effects editor and dialogue editor about their run with a reptile that runs riot in old Manhattan

IT FIRST APPEARED in 1954, the product of Japan's Toho Studio. Filmed in glorious Tohoscope, the giant fire-breathing lizard known as Godzilla threatened civilisation numerous times down the years, and also, when it felt like it, provided some scaly support—not to mention plenty of unintentional laughs. You see, the Godzilla movies, while very popular in their native land, were also extremely cheap and incredibly hokey, prompting reviewers to dismiss them with comments such as 'a hoho from Toho', or 'more sci-fi shenanigans, providing a field day for the special effects crew and a day off for the scriptwriters'.

Well, courtesy of Sony Studios, the outsized amphibian is back, having crossed the Pacific to wreak havoc on New York City, and it has to be said that, while the story-line is not about to challenge *Gone With the Wind* (or even *Blue Hawaii*), the special effects are stunning. This is pretty much what you would expect given the huge budget and the fact that Roland Emmerich, the director of *Independence Day*, is at the

helm. Starring Matthew Broderick and Jean Reno, Godzilla is a highly entertaining monster movie that drew on a lot of talent behind the scenes in order to produce the stunning sounds and visuals. After all, how should the reptile's voice be updated from the elephantine braying of yesteryear? That was the task facing Scott Gershin when, as one of five sound designers assigned to *Godzilla*, he and his colleagues at audio post facility Soundelux Hollywood first became involved with the project early in 1997.

Since its inception during the early eighties, Soundelux Hollywood has worked with all of the major film studios in the Los Angeles area, in the process earning itself a number of American and British Academy Awards as well as Golden Reel Awards for Best Sound Effects Editing and Best ADR Editing. It is now home to the world's largest sound library, comprising over 5.5m sound entries and a running time of more than 5,000 hours, and the company also provides sound transfer and dailies in conventional analogue 35mm

film formats and transfer services for nonlinear digital editing systems such as the Avid, Waveframe and Fairlight.

Per Hallberg and Wylie Stateman were the two in-house sound supervisors on *Godzilla*, and a big extravaganza was scheduled for the ShowWest trade convention in Las Vegas in March of 1997, announcing that the film was going into production. A promotional trailer had to be produced, and Scott Gershin's job was to come up with what he describes as 'the classic roar'.

'We had a team of people working on it,' he recalls. 'To start with, [designer] Martin Lopez did a lot of research into what that sound was made up of, and he created a vast library of information. We've got everything from baby elephants, hippos and badgers to trumpets being blown in the stairwell and metal being scraped. We came up with a couple of thousand different sounds that we thought were similar to that of the original *Godzilla*, and the reason why we did this was also because we needed to expand his vocabulary for the actual movie. At that point [designers] Mike Regan, Alan Rankin and myself started to experiment and create something.'

It had taken two months to create that 'classic roar' for the trailer. Now they were into the movie proper. Foley artist Gary Hecker was recruited to vocalise the grunts and growls of the 'big guy'

—as Gershin affectionately refers to the lumbering lizard—and the results were subsequently used as a template for all of the various possibilities; happy Godzilla, angry Godzilla, quizzical Godzilla, and so on.

'We replaced or added according to what we needed,' Gershin explains. 'We did some more vocalisations ourselves and also used the whole animal kingdom together with the thousands of sounds that we'd accumulated. That enabled us to create the right combination and the right variance. You know, you don't want to use the same sound over and over again, but rather something similar with a slightly different twist. Then we used the classic roar as the thread that ties all of the sounds together.

'Basically, Godzilla's got to sound like Godzilla can sound and also like T-rex, so it was a case of, "Okay, this is where we're starting and this is where we want to end", while always staying within the lines to ensure you know that it is Godzilla. In doing that, Alan and I saw the picture pretty quickly and we were then able to construct the vocalisations.' The film's computer graphics were being created well into the project's schedule, and so there were elements that the audio people at Soundelux never even saw until the last couple of weeks, but that they nevertheless had to create a sound characteristic for. Then there were the Godzilla babies which they never saw until a couple of weeks before they began finalling, and so, all in all, the effects crew had to use its imagination quite extensively for much of the time.

'Sometimes we ended up seeing some rough graphics without any kind of texture map,' says Gershin. 'So we might get an idea about the movement of the head, but then all of a sudden the texture map would appear, and they had put the eyes in, and the eye would do a little blink, and the head would do a little nod, and we were thinking, "Maybe we should go this way". Frank Walker also helped out with the sounds of the babies; although we did not end up using quite as much of Frank's material. We wanted the babies to have a lot more personality, because visually we wanted to confuse the audience a little bit. You see, they're 9-foot babies, so at one point we wanted them to look and sound innocent, and to that end we listened to the way that birds and other animals reacted. Then, within about 20 minutes we needed to have them grow up to be in their teens. So, they had to go through the bratty 2-year-old stage, where they're curious and want to get into everything, and then they needed to be a definite threat to the people in the film. The audience has to start off going, "Oh, they're so cute," and "Oooh, they're so big," and then by the end be saying, "Oh-oh, look out, they're a huge

threat.". At that stage in the movie we learn that they are asexual and can procreate by themselves, and that, as 200 babies have just been born, within a year there could be 20,000 dominating the Earth. Accordingly their voices become less cute.'

Waveframe and ProTools 24 were the two main systems employed at Soundelux for the Godzilla project, together with a wide array of outboard gear; a combination of Lexicon PCM80 and PCM81, Eventide 4000 and 4500, Lexicon 480L, M5000 and numerous plug-ins.

'We didn't want things to sound synthetic, or too transparent either,' Gershin explains. 'When we were recreating the vocalisation we took the trumpets and gave them a little pronunciation, and to get it away from sounding like a trumpet I ended up using multiple Harmonizers. I've got a light-beam controller, and so as I move my hand through it I can control and change the various attributes that manipulate the outboard gear. Therefore, as I moved my hand up and down to picture I could start tailoring the sound to be able to pitch, which was a combination of the Lexicon and the Eventide. Then I used a multiband distortion device to get it out of that beautiful fundamental that trumpets have and give it a little more grit. So, on the one hand I was dividing the pitches to create a more dissonant, complex sound, and then the multiband distortion added the gutteral, throaty noises. I found that by using normal distortion it just sounded like distorted audio, whereas with multiband distortion, it had a whole new personality. I think we tried every technique and every process that we'd heard of and probably invented a few. We really delved into areas that were new to us or we went back to old tricks, and there was a certain amount of time just given over to experimentation; although we didn't have all that much time. I mean, any film with computer graphics is always evolving.'

Sound effects editor, Chris Assells, could not agree more.

'The most difficult thing on this picture was that a lot of it contained digital CGI effects,' he says. 'I would get some footage and there would be Godzilla running down this canyon, and so I'd cut my stuff and we'd send it to the dub stage. Then, a couple of days before the dub they would send us this new version and all of a sudden there'd

be 20 helicopters following him. I'd now have to cut those in, and sometimes there was a very short time to cut a lot of stuff.

'These days they can generate images so quick and they look so beautiful. Virtually every helicopter in this film is a CGI helicopter, and it's only in recent years that things can get thrown at me so fast. Plus it seems that schedules have got shorter, and on this film in particular we were up against the wall with regard to the release.'

Indeed, rather than deal with the monster, Assells was responsible for the sounds of objects such as helicopters, jets and cars, all of which had more of a grounding in reality. Other effects editors included Dino Di Muro, and both he and Assells are from the old school that saw them cutting on movieolas before 'sneaking into this digital thing'.

Assells edited *Godzilla* on a Cyberframe, asserting that he finds it to be 'very film-friendly. I use it pretty much like a splicer and a synchroniser and a movieola, and transferring over to it from a movieola was relatively easy because it rolls like film, I splice it like film and I can do crossfades just like I did with film.'

Using virtually no outboard gear, Assells was another Soundelux man to make good use of the in-house library.

'We have an incredible jet library,' he

says. 'We just did Airforce One, and so I was able to use a lot of the stuff from that movie in this film. Then, on top of that I also received a lot of good sweetener effects from Peter Sullivan. He created these low-end parts to bring the jets right in your face, and I combined these with helicopter sounds and I think it worked really well. I went pretty literal with this show, and so what you see is pretty much what you hear, except for sweeteners. For instance, sometimes when a jet would boom by we'd sweeten it with a cannon blast or with a thunder hit, but nothing manufactured or synthesised. They were pretty organic sounds. I didn't want people to be hearing CGI helicopters and CGI jets.'

Elsewhere on the audio effects front Tony Lamberti was assigned the task of designing the sound of Godzilla's >
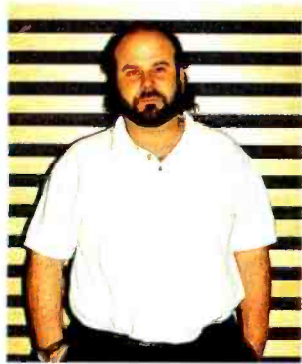
Scott Gershin, sound designer   Tony Lamberti, sound designer

< giant footsteps as well as the explosions, crashing debris and smashing noises that take place as it trashes buildings and vehicles around Manhattan. 'I'd already done creature footsteps and different types of destruction when working on 'The Relic,' he says, 'so I guess they figured I'd be best suited to this department on Godzilla.

'Obviously it's a huge, huge lizard, so we wanted its footsteps to have a lot of weight and really shake the room. I therefore did things like taking multiple explosions and slowing them down, and making new layers of multiple explosions to use for really super-low impacts that would play out of the main speakers—not using the subwoofer at that time—and really push a lot of air. I'd played with that technique a little bit in the past, but not to this degree;

for other creature footstep sounds that I've done I've used things like gunshots, but never really big explosions that were layered on top of each other to get the kind of impact that we were after this time.

'The explosions themselves were made from scratch and created from various elements. We did a whole recording session where, together with Skywalker and a couple of other sound effects houses, we went out in the field and captured a bunch of real explosions. Then we came back and used those as raw elements, stacking them up, sampling them, pitching them down and creating new explosions.'

When using the recordings from the field, it certainly wasn't a case of 'any explosion will do'.

'I played it various ways and so I wanted to have something with a long tail to it,' says Lamberti, referring to the blast, not the monster. 'For example, there's a scene where there are people standing on the street in New York City and you can see the cars bouncing while Godzilla is bounding along in the distance. So, we were playing with perspective and as the footsteps come

closer we use different types of explosions with longer tails, and so on. It's not like we just took one and hit it over and over and over again.

'Other elements that we created for the footsteps included a subwoofer impact sweetener, which can almost be described as an impulse. It's like four cycles of a single wave form that, when played through the subwoofer, just kicks it and then gets out of the way without getting too muddy. So, that's how we got the real impact, and I put some M5000 reverb on it to give it more of a tail. Also, I did a quick mixdown of my initial footstep impact and ran that through a tc multi delay to get the reports off the buildings.'

As for the concussion sounds of buildings such as Madison Square Garden deconstructing, these derived from explosion effects in the Soundelux library that were tailored by Lamberti to fit the picture. Meanwhile, for ripping noises cosupervisor Wylie Stateman made some special recordings with metal on the Foley stage and Peter Sullivan captured the scraping sounds of dry ice. Using an Emulator E4 and various outboard items Lamberti then sampled and manipulated these in order to achieve the desired results. He didn't, however, have time to create the sound of the babies' footsteps, which were expertly taken care of by Foley supervisor Craig Jaeger.   >

< Lauren Stephens edited dialogue on the picture, and what with all of the opticals that kept appearing and changing she was in constant update mode. Then there was the rain, plenty of it, teeming down in Manhattan throughout the film, and along with all of this water came some predictable sound problems.

'They shot the picture in stereo but it wasn't true stereo,' says Stephens. 'It was body mic and boom, so there were pretty much eight tracks of material to sort through. Well, one side was always just hash—rain and misery—and I had to get rid of that. Then on the other side the backgrounds would be very different, the rain would be very different, the quality of the sound would be very different, and they were also replacing an extraordinary amount of dialogue.

'We go into the situation knowing that they're going to probably try to replace at least half of it. A lot of the computer-generated stuff is MOS, but then they have a lot of huge wind machines and generators and vocal directions from the director, and a lot of it is completely unusable. I mean, they had scenes in that movie where cars are blowing over and debris is flying everywhere, and the characters are trying to carry on some kind of conversation—in fact, pretty much any exterior scene where Matthew Broderick and the other actors are trying to speak to each other presented a problem. However, with other scenes

we'd kind of go at it like there's no such thing as ADR and we'd do the best we could with the available material, so in the end I'd say they kept between 30% to 50% of the original dialogue... Going into an action film as a dialogue editor you know you're not going to shine. It's not your time. That belongs to the effects guys.

'Kevin O'Connell was the dialogue mixer and he's really good, and we also used NoNoise to varying degrees. For instance, there's a taxi scene where we're in the cab and rain is shooting down on it as we're screaming down the street. Well, there was just the most wide range of loud hiss that you've ever heard in your life, and even the NoNoise guys sent the stuff back saying, "What in the hell was that?", so, there's only so much you can do, and it was particularly challenging to try to give the director and the mixers some options.'

While the effects team on *Godzilla* worked at Soundelux, the in-house dialogue, Foley and ADR people went to Sony Studios in nearby Culver City where they used the Waveframe and laid back to 6-track mag tape. Mag editors updated any changes to the pre-

Chris Assells, sound effects editor   Lauren Stephens, dialogue editor

dubs, while Lauren Stephens handled the changes to dialogue and ADR. In all she had just six weeks in which to complete her work—an extremely short schedule given the size and scope of the movie, albeit that weekends were never used—and thereafter the mix also took place at Sony on a Harrison MPC console in the Cary Grant Theatre.

'I'm really happy with the mix, really happy with the way my stuff turned out,' says Chris Assells. 'Often you spend all of this time cutting stuff, and you send it to the stage, to the mixers and the supervisor, and then you go and see the movie and you think, "Where is all that stuff that I cut?". On this project we sent the material to the stage and to the mixers and when we saw the finished film we went, "Hey—there's all that stuff we cut". It was very nice.' ■

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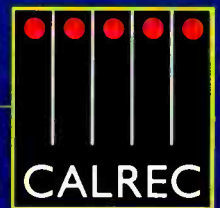
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The magic bootabout

Emotive enough to empty the streets and restaurants of London on match days, the 1998 World Cup is setting standards that other events will struggle to equal. **Kevin Hilton** reports

BLUES AND YELLOWS. Giant rubber flowers. Out-of-work actors dressed up as bees bouncing up and down on trampolines, ejecting little coloured, helium-filled balloons from their enlarged posteriors so that they drifted languidly into the clear blue sky above Paris, watched by thousands in the glistening Stade de France and many thousands more in bars and homes around the world, all of them with a single thought: what the hell does this have to do with football?

Another major sporting event and another over-the-top, totally inexplicable opening ceremony, watched by football fans who applauded the sub-Serge Danot surreality of it all with more good nature than it deserved, secretly hoping they would get on with it and things could kick off for real. Still, at least it was more entertainingly professional than what the Brits came up with for Euro 96.

For just over a month, the 1998 World Cup has dominated television and radio schedules around the world, with the possible exception of the US. Stateside critics blame some of the chronic lack of interest on the TV coverage, saying that the shots are too wide, that there are not enough close-ups and consequently all viewers can see are lots of little figures running around a huge pitch with no real indication of who is who and what is going on.

No doubt the French TV directors responsible would take umbrage at such criticism, particularly as the international coverage of France 98 is under the direction of six of the most experienced practitioners of this particular art in the country, having worked variously at TF1, Canal+, Tele Monte-Carlo and France Television. While the overall production plan stated that there was no intention to break new ground, the matches have been covered in a straightforward way, offering plenty of different views of specific incidents and only being confusing if the teams were not wearing distinctive enough strips. It all reaches its conclusion this month, with the final taking place at the still shiny and new Stade de France, just outside Paris. To get to this point, 32 teams have been whittled down to just two, supposedly the two best in the world today. To get here, viewers have witnessed hours of players running up and down the various pitches, referees being affected by the red mists, protracted segments of analysis and opinion, the always popular 'Who doesn't know their national anthem' competition and, regrettably, scenes of violence from people who deliberately forget that it is only a game.

This time around, it was truly a World Cup competition, with teams from Europe, Africa, Asia and the Americas. This global interest, combined a general world-wide appetite for sport warranted a massive broadcast operation, providing international feeds for over 180 television broadcast organisations and in excess of 130 radio stations, which require facilities to tailor the coverage for their domestic audiences, and more than 4,500 accredited rights-holding broadcasters. Logically such an undertaking could not be the responsibility of a single broadcast organisation. Consequently the broadcasting of France 98 was entrusted to TVRS (Television and Radio Services) 98, a consortium established by the French Broadcasters Group of the EBU, comprising France 2, France 3, Radio France, Radio France International, TF1, Canal+ and TDF. Formed in May 1995, this organisation put together the production plan for the tournament coverage and co-ordinated the assembly of the facilities needed to accommodate the broadcasters, journalists and their associated entourages. The operational outline was drawn up by a committee that included former French international, and now president of France 98, Michel Platini, and executives from TF1, France TV, Canal+, the EBU Sports Working Party and TVRS itself.

From this initial plan, which covered camera positions, varieties of shot and, after further consultation, graphics and an additional camera, the setup that went on-air at the Stade de France on 10th June was ratified. With the primary aim of producing a high-quality international pool feed, a fully digital production chain was decided on, a first for the TV coverage of a World Cup.

The ten match venues have identical setups, each based around a digital scanner truck. Action is captured by 17 cameras, each positioned to give comprehensive coverage of the match with the ability to focus in on the decisive moments of each clash. The core is Camera 1, positioned high in the nearside stand, giving a wide shot of the field. The other cameras offer specific handling shots and views at field level, from behind the goals, the sidelines and minicam shots around the goal.

Due to rules laid down by governing body FIFA, only two of the 17 cameras are hand-helds, and these are purely to show the players coming on to the pitch, the reactions of the substitutes and coaching teams on the bench and for 'flash' interviews, catching coaches and players immediately after that glorious victory or ignominious defeat. FIFA's insistence on as little unnecessary movement on the touchline as possible—which extends to managers jumping about and shouting to rally their troops —applies to the audio. Of the 20, primarily shotgun, microphones positioned **>**

< around the pitch, eight are hand-held by operators with radio comms, who can be instructed to zero in on specific action to complement the pictures of an encounter with the usual sporting exchanges between players. As these are fixed positions, all the technicians can do is extend their arms in the appropriate direction.

The production plan outlines 'clear and objective' coverage, stating that the aim is 'not to break new ground but to maintain a high production standard throughout'. This document also says that 'stereo sound production will not interfere with the audio coverage, and efforts will be made to ensure crowd noise will not drown out the thud of a kicked ball or verbal exchanges between players. Jean-Yves Pouloux, technical manager at TVRS 98, says that this is entirely in keeping with how football matches are traditionally presented in France, and that it was decided not to try anything special, merely to give a basic, well-produced pool sound to be used by all subscribing broadcasters.

But the coming of new digital services means that the use of future techniques—specifically wide-screen pictures and multichannel sound—could not be forgotten. France Television (France 2 in association with France 3) is producing a 16:9 wide-screen feed with Dolby Surround Sound, which is being broadcast by French satellite channel TPS as Superfoot 98 and by Italian public service broadcaster RAI. France 2 is using a different set of cameras from those of TVRS to source this material, which is mixed in two of its scanners, equipped with Dolby's SEU4 encoders and SDU4 decoders for the audio.

The BBC is broadcasting 27 of the 44 matches shot in 16:9 on its Choice service, but with a digitally compressed MPEG stereo soundtrack. NHK has sent over two mobiles to provide footage for its domestic high-definition output, taking the international stereo feed and adding surround information. German broadcasters ADR and ZDF have also sent scanners to cover proceedings with their own cameras, taking the basic left and right signals and adding surround information from their own microphones.

For the international broadcast organisations attending each match, TVRS has laid on extensive facilities to enable each station to tailor its coverage to its specific domestic specifications. These are divided into facilities that are not associated to the match, that is studios, tape playback equipment and transmission time used in the run-up to a game; and facilities associated to the match, being commentary and announcer positions, half-time interviews and studio, play-back and transmission time when a game is on.

Two vital parts of any sports coverage are the on-screen graphics and

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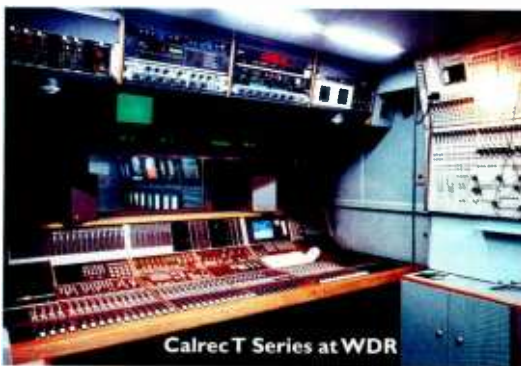
slow-motion replays, to emphasis the agony or the ecstasy. TVRS instigated a basic graphics insertion software, enabling easy retrieval of material from an on-line database.

Statistics are made available by the Stat+ package, developed by Compusport, which gives a range of real-time data to be overlayed onto the picture during matches and replays. Eight slo-mo units are used during matches, all digital Live Slow Motion devices, with four being Super Live Slo-Mos, giving 75 frames per second.

Commentators are the unseen stars of TV football coverage: from the howling enthusiasm of Brazil to the mind-numbing obviousness of the UK (one British pundit has already this tournament uttered the immortal phrase: 'The team that scores the most goals will win'.), every broadcaster wants to put its own spin on proceedings. Providing the equipment is AEQ, which is leasing to TVRS the new system that got its first try-out at the recent Winter Olympics in Nagano. For the opening and closing matches at the Stade de France, 150 positions are available; Round One fixtures only had 100, with the number increasing to 125 for matches in Round Two.

Each position features a commentary unit, two headsets and microphones and two TV monitors, with the ability to select either the international feed or the on-air output of TF1, France 2, France 3 or Canal+. When matches were played concurrently during the early phases of the competition, remote coverage of the other games was also provided. The commentary positions were installed and managed by Radio France, which offers a choice of audio circuits of either 3.4kHz, 7kHz or 15kHz bandwidth.

From the scanner, the international signal is sent to the Technical Operations Centre in the broadcast area of the stadium, thence to the International Broadcasting Centre in Paris. This is the core of France 98's broadcast coverage, where all the feeds are distributed and where the visiting broadcasters have their main studios and administration offices. Housed in two halls of Paris Expo, an exhibition complex in the south-western Porte de Versailles district of the capital, the IBC's technological infrastructure has been provided by France Telecom, TVRS' official part-

ner in this venture, and two official suppliers, Sony Broadcast & Professional and Tektronix.

France Telecom is providing the telecommunications network between each stadium and the IBC (one permanent fibre-optic circuit for the international feed at 34Mb/s ETSI quality; a satellite link for the duration of games to back up the international signal; and a fibre-optic circuit, again for the duration of the match, to carry the clean international feed) and is operating the joint booking service used by rights-holding broadcasters. Other services provided by the French telecoms group are unilateral broadcast requirements between venues and the IBC (including commentary and co-ordination circuits, which are carried over fibre-optics); transmissions from any OB van brought in by broadcasters to the Technical Operations Centre; video transmissions from the IBC, either on fibre or satellite uplinks (the latter from a dish farm at the back of the IBC or FT teleports; the provision of 400 outgoing audio circuits (ranging from 3kHz to 15kHz); provision of satellite news gathering links, transmission circuits, ISDN connections and production and post-production facilities at team training grounds and hotels; and full supervision and security on video and audio circuits.

Sony has provided more than 65 video machines from its Betacam SX and Digital Betacam video ranges, which are being used for the acquisition, editing and transmission of programme material. SX is being used specifically because of its ability to integrate with Betacam SP, a format that is widely used in sports broadcasting. The host broadcaster VTR format is Digital Betacam 625/50. Tektronix is supplying a Grass Valley SMS7000 routeing switcher, two GV 1200 production switchers, test and measurement equipment, Deko character generators from the Pinnacle Systems range and sync pulse generators for all video production gear.

All of which should be unknown to the average viewer, leaping about the front room, avoiding knocking over the glass of beer or stepping in the half-eaten pizza. It really takes something special (or particularly bad) to draw attention to TV coverage. It was on the night that Morocco knocked Scotland out of the competition, but still went out themselves due to Norway beating Brazil. With the Scots' fate already sealed, the coverage dramatically cut to the Norwegian penalty; with this in the back of the net, everything cut back to the blissfully unaware Moroccans celebrating their victory. As the awful truth became known, one could not help but feel sorry for them, while acknowledging that, sports fan or not, it was damn good television. ■

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Meta Sound

Having reported on the introduction of D&R's flagship Octagon film console, *Studio Sound* now documents the first installation. **Zenon Schoepe** writes from Amsterdam

HERALDED as the only real console choice for those who are out of the league of the digital and digitally controlled analogue film monsters, D&R's Octagon multiformat analogue desk is now shipping. Indeed, the Dutch manufacturer achieved quite a coup with the powering up of its first installation during the course of the recent European AES in Amsterdam at leading Dutch film studio Meta Sound. And this when the manufacturer celebrated its 25th anniversary.

The desk is in the facility's newly refurbished Studio C mixing theatre and Meta has also added a Kinoton FP38 high-speed film projector, and a 24-bit, 24-track workstation from Dutch company Augan. Monitoring is by Stage Accompany, another Dutch company.

Meta Sound MD Ad Roest is pragmatic about his choice of console. 'Once we had come to the conclusion that the Harrison Series Twelve was too expensive for us we had to look elsewhere,' he says. 'The problem with an expensive desk like the Harrison is that when you can't afford it in the sort of size you'd

really want, you start looking at ways of making the desk smaller and cheaper. In the end you leave out all the sorts of things that you liked about the desk in the first place just to get closer to a price you can afford. You end up with a variation of the desk you wanted.'

Faced with such practicalities Roest claims that the Octagon was the only real alternative against the super league of film desks. 'It's a slight step backwards from what we would have wanted ideally, but in three or four years time when there is a really revolutionary digital film console for, perhaps, less money we will be in a different position because I will be free to change.'

Roest reminds that the economic pressures he is under are the complete antithesis of those that, for example, impact on a national broadcaster. His market by definition is small and its growth potential is limited.

Meta Sound may well be the leading film mixing stage in the Netherlands but that is the full extent of its influence. 'I can write the cost of the Octagon off in three years whereas

with the Harrison I would need probably more than ten years,' he continues. 'The upshot is that I will be able to buy a new console in three years time, but maybe I will keep it for ten.'

THE OCTAGON replaces a 13-year-old Studer 900 Series console from which Roest says they got excellent value for money —the desk has never had a fader, switch or pot problem in all that time. The single blemish on its record was a blown IC in a stereo equaliser. When the desk was bought Meta Sound was operating out of Hilversum, it moved to its present location on the Amsterdam ring road ten years ago to a newly constructed facility. Fortunately the wiring and arrangement of the two sites was similar enough to make the transplant to the new theatre relatively painless. Even so it was a week before the console was up and running whereas the recent Octagon install took four days in total—time down is time without income according to Roest.

Studio C has undergone some minor

including digitally controlled analogue switching, both faders are motorised on each channel, on-board software dynamics, and joysticks—a well-featured desk in total. The purchase of the Kinoton FP38 high speed projector will have a significant impact on productivity, according to Roest. The projector is slaved, like everything else in the studio, to Augan master time code.

He recalls that when the facility bought its first DAW, the Lexicon Opus, the fastest video machine was a Betacam, and while the Opus could jump an hour just by typing in the time the Betacam still had to spool.

'For five years the speed of the system was slowed down to the speed of the video machine,' he remembers. 'Now our video is on optical disk, and while they still slave to the audio they're in sync within a second. Sound has always been the master at Meta Sound, and that's a good way of working.'

THE FACILITY has four recording studios, two edit suites plus the Octagon-equipped mixing theatre. All have Augans and video on optical disc.

'The reason we bought the Augan was that the most frustrating thing about the Opus, apart from the fact that at the time hard disks were small and expensive, was that you couldn't move a project from one room to another. Augan was the first manufacturer to work with optical disc, and you can take them wherever you want. We work for Disney, and record in one studio, edit in another, and then we send the disks to Delta Sound in the UK where they mix from it. We have six systems here now, and sound designers can just walk in with a couple of disks and get on with it.

'Another important point is that we've never had a problem with the sound of the Augans and now with 24-bit and 96kHz on the machines it'll be even better,' he continues. 'I have no idea whether 24-96 will be the way we will be working in the future, but at least if somebody asks for it we can do it.' He adds that the Augan performs all the functions of a dubber and that he's considering using it for just that.

'When I started to think about re-equipping the studio I thought we should be able to run ten DA-88s or more, but the trouble is that now we have a high speed projector the DA-88 is the slowest transport. I'm thinking about another stripped-down Augan purely as a replay device. It's really fantastic to have just the one standard and to be able to run the disks from our first Augan production from five years ago on our newest machine. This company has been working for Disney for 15 years, we are asked sometimes to work with tapes from 10 or 15 years ago, and, of course, we have

them, and we can replay them. Film is very well standardised as a medium.'

Roest clearly feels passionately about file compatibility issues and despairs at the number of incompatible options. 'Manufacturers concentrate on all their different solutions, but nobody thinks about joining in with somebody else's format. That's the most stupid thing, and it's been happening ever since we first got computers.'

Meta contemplated D&R's Cinemix but deemed it too small for the Netherlands' biggest film theatre. 'Our Octagon has four times the number of inputs of our old Studer, and eight times the number of outputs. Rerouteing is an important consideration in film mixing, and it's easy on this desk.

'It won't do our business any harm,' claims Roest. 'The lack of faders on the Studer was very frustrating for everybody, and clients might have liked me as the mixer, but if I didn't have enough faders then is was difficult for me to do the job for them. The Octagon is so sophisticated in the routeing, panning and grouping.

'There must be a good market for this console because there are so many small countries with small film industries. We're always looking to what's going on in Hollywood and the glamour of the big studios, but the reality for most film-mixing theatres in most countries is far more down to earth, and all about small markets. We're always working with people who have spent the last of their money on a project that they want to finish because they believe in it, and we can't charge people like that 1,000 Dutch Guilders per hour.'

That being said, the Octagon's arrival will be accompanied by a room rate increase from 400 to just under 500 Guilders per hour. However, disregarding the issues of investment and return it is as a film mixer that Roest is most impressed with the Octagon.

'You can get this desk to do what you want it to do,' says Roest. 'What I like about the Harrison is the automation, and if you've worked with an 02R you'll know how complicated this can get, but as a mixer you are being paid to mix not program. The Octagon's EQ isn't automated, but that's how they achieve the price, and you have to live with it; although you can recall EQ settings. But the relatively low cost means we can still buy automated digital outboard for when we need that and buy some good reverbs.

'We've created what I think is a really nice studio here, and had I bought a Harrison there is no way that I would have been able to afford a new projector and that was important to me. The Octagon package is a fantastic one for the money,' states Roest. ∎

renovation with new flooring and equipment pods. More DA-88s have been added, and new Dolby units, including digitals, have been integrated. 'It all works very nicely with the monitoring system on the Octagon. They've thought of a lot of things, they've been very clever. It interconnects really well with the Dolbys,' comments Roest.

The Studer 900 was only 8-group to stereo and Roest admits that they had too few faders even on the very first film that was mixed on that desk. The layout and construction of the console precluded the addition of anymore whereas he points out that the Octagon is not only expandable, but that sections of the desk can be moved around freely if required.

The addition to the Studer of a sidecar Yamaha 02R had recently helped matters, but the alternation between the earthy analogue presentation of the 900 Series compared to the assignability of the 02R was not a comfortable solution. Panning possibilities were also limited.

The multiformat Meta Octagon is 48-channel for 96-inputs, and is fitted with D&R's fullest automation package

The Bug Bomb

Amid the barnstorming campaign about the implications of the Millennium Bug, **Kevin Hilton** talks to leading companies in broadcast and pro audio about solutions

ICK, TICK, TICK. The Millennium Bug, the Millennium Bomb, the Year 2000 Question. However you slice it, it means trouble for some, and fat consultancy fees for others. In the last six months, the issue has come storming into the public arena, amid general confusion and fears that the whole matter has been left much too late to do anything about. The UK Prime Minister, Tony Blair, has summed up the situation by saying, 'The Millennium Bug is one of the most serious problems facing not only British business, but the global economy today. Its impact cannot be underestimated.'despite such pronouncements, those monitoring the problem still say that its true significance has yet to be appreciated. Mark Rosselli at the Year 2000 Support Centre underlines this belief by saying, 'The situation is not being stated strongly enough.'

The main areas of concern are banking and financial institutions, health care, administration and management systems and other large-scale concerns that may depend on computer systems running older forms of software or operating systems.

The word is 'compliance'. Compliance with the future or, in simple terms, just knowing that it is the year 2000 and not 1900. In a world dependent on time, getting things done on, and in, time, not knowing the right date is going to cause problems.

Scare stories abound. But there are guidelines that give an idea as to whether everything is going to be all right, or whether it is time to dig out the manual typewriter and the abacus. There may be national variations, but those set out by the British Standards Institute (BSI) give a good outline of what is and is not compliant. The BSI definition states: 'Year 2000 conformity shall mean that neither performance nor functionality is affected by dates prior to, during and after the year 2000.' Specifically it sets out these rules: Rule 1: No value for current date will cause any interruption in operation. Rule 2: Date-based functionality must behave consistently for dates prior to, during and after year 2000. Rule 3: In all interfaces and data storage, the century in any date must be specified either explicitly or by unambiguous algorithms or inferencing rules. Rule 4: Year 2000 must be recognised as a leap year.

The three main areas of concern on computer systems are: the BIOS (Basic Input Output Systems), the OS (Operating System, which works over the BIOS), and whatever software is being used to run the application. While attention has immediately focused on PCs, experts say that the real problems will come with mainframe systems, embedded computer chips (for example in a video tape machine, which has a chip for any date-related information), networked situations (where a fully compliant device could be affected through interaction with a noncompliant system), and anything running old software, or operating systems, or on old computers.

In the campaign to bring the Bug to public attention, the areas of broadcast and recording have not been prominent, but they are still susceptible. Mark Rosselli observes, 'The implications are less catastrophic than in other areas, but not for those directly involved. Broadcasting is more dependent on live computer operation, while in studios there may be more embedded systems.'

Perhaps the most obvious use of computers in recording, and, to no lesser extent, broadcasting, at least in terms of daily working, is the digital audio workstation. Ted Hayton, marketing manager at Studio Audio, comments that every Pentium machine the company is using with its software is Year 2000 compliant. 'The only problems could occur where the PC is of dubious origin,' he says. 'That's when the BIOS store for the date would be of only two digits, but people are unlikely to be using that equipment in the Year 2000. The scare is spurious as far as PCs, and our profession are concerned because people will want to use the latest systems.'

'Systems not based on PC platforms are generally held to be compliant. We're confident that it is not relevant to the way our system works,' says Jason Power, product support engineer at AMS Neve. At DAR, managing director Mike Parker comments: 'We've had to look into the matter to prove to our users that the units will continue to work. We've also had to provide a certificate of conformity for our big corporate clients, but in general a lot of the work being done on these systems is not date dependent.' Although Digidesign's

Pro Tools now offers a Windows version, the company is certain of Mac compliance. Sonic Solutions is similarly confident about its Mac-based products, but points out that LightSpeed and MediaNet, which run on Windows NT, may need adjustment in the regional Settings Control Panel.

Nick Cook, UK operations director of Fairlight, simply says, 'The clock keeps ticking', while both Sonic Foundry and Creamware vouchsafe their systems' compliance.

It has become almost a fact that the Apple Macintosh is immune to the Millennium Bug. This is largely confirmed in the Year 2000 section on Apple's Web site: 'The Mac OS and most Mac applications will handle the year 2000 (and the next 27,940 years), no problem.' but the company does point out, 'The only applications that might hiccup at one minute past midnight on New Year's Day 2000 are the few that do not take advantage of the Macintosh Toolbox calls available in every Macintosh ROM since the beginning of time.' As with everything, check to make sure.

Over on the Microsoft page, Uncle Bill Gates sympathetically peers out at you from behind his glasses and empathises, 'Microsoft understands that Year 2000 concerns extend beyond technical considerations.' Absolutely, it could mean catastrophe, Bill, but not everybody can afford the lawyers he can. It is best to check which of his products are or are not compliant. In terms of operating systems used directly in broadcast and recording, both Windows 3.1 and Windows for Workgroups 3.11 are deemed to be compliant, but 'with minor issues', while, at the time of writing, Windows NT 3.51 had yet to be tested. Although Microsoft hails Windows NT 4.0 as compliant, it admits that this system does not recognise 2000 as a leap year, which, according to the BSI definition, means that it is not compliant. The company is offering tools and service advice through its Web site.

ICK, TICK, TICK. It may become necessary to destroy the world in order to save it. The thinking seems to be that people would have thrown away all the old computers, mainframes and dumped the ageing software and operating systems long before now. But we live in cost-effective times; if nobody notices you are still using a ten-year-old system and everything is okay, then don't let on. A radio executive in the States helpfully emailed on this topic: 'The latest date our traffic system will accept is >

How Many Codecs Do You Need To Make All Your ISDN Calls?

< 12/31/99. Try 1/1/00 and it chokes. (It only accepts two digits) The computer is fine, just that this particular traffic system is dying. We are going to switch our traffic-billing over to CBSI this summer, though, so that old software will hit the bit bucket after that. I can't complain, it has served us well.'

The real job now is to identify candidates for the bit bucket. As Brian Harper, head of central services at Pearson Television, observes, 'There are variations of compliance: either equipment is compliant, or it will be, or it isn't.' Dr John Emmett, formerly with Pearson and now running Broadcast Project Research, is of the reassuring opinion that broadcast is not a high-risk area but does say, 'There is very little production equipment that is time or date sensitive. Things like Odetics cart machines probably are, and so are scheduling systems, but these can be manually assisted or cached ahead.

'Systems that are at risk are those that are pulled rather than pushed, and domestic equipment. Where there are problems are library-type allocations and registrations, for example big administration systems at the State Department.'

Like other manufacturers, Odetics Broadcast has instigated a programme of testing. Director of sales and marketing Bob Stopford underlines the compliance of the company's latest product, Roswell, but says of others in the range, 'Some of our machines are almost ten years old, so we have identified the components that are not compliant, drawn up a policy document for our customers and produced a fairly straightforward upgrade path.'

Routeing and automation specialist Pro-Bel has appointed a Y2k manager within its software division, established a test suite, and has been working with some of its clients. 'We have our own target of the end of the third quarter to have checked all our products,' explains Neil Maycock, general manager of Pro-Bel software. Some of our customers have set the end of the year to be compliant and we will beat that by a quarter.'

Maycock comments that some users are still on Windows 3.1, the file manager program of which does not display above Y2k. 'But this doesn't affect our software,' he emphasises, 'but there are a few our products that run on old programs. In these cases we have come up with a fix.' Automation has also become established in radio; the ubiquitous system is Selector, developed by RCS. Managing director Sebastian Holmes admits that most of the current software does not comply, but underlines that a free upgrade will be available by 31st December. 'Selector was written in 1978,' he says, 'and nobody thought about the Year 2000. We have a rolling programme in place, with free upgrades to ensure compliance.'

For broadcasters like the Capital Radio Group, there is less risk here because the majority of its automation systems are relatively new. Head of IT Peter Willison says that central heating is more of a worry but that there are potential problems in terms of operating systems like 3.11 and PC BIOSs. 'Most PCs can be simply re-started with a change of time,' he says, 'but it is a question of what it will do each time you do that. We're not necessarily testing everything—the critical nature of the item will dictate whether we do or not.'

A corporation-wide survey is underway at the BBC, covering broadcast technology, IT, property systems and relationships with suppliers. Project director Stephen Reid comments, 'In broadcast technology we have relatively few problems with the hardware, and; although it appears there are some software systems that could cause concern, it is still a relatively small proportion.'

Internal problems could be bad enough, but for service providers, things going wrong for their clients could be potentially disastrous, if not litigious. Brian Harper at Pearson comments, 'It has been a matter of concern for our clients and they have asked us what our policy is and what we have been doing to deal with any potential problems.

'As Pearson is a worldwide distribution company we have been looking at all our operations in different locations. Another issue is our archive; because we have such a huge back catalogue of material, we daren't have a system that tells you that you don't have any product any more.'

Tick, tick, tick. Can you hear that? Europeans say that they can and are of the opinion that American and Asia regard the Millennium Bug as purely a European problem. This does not seem to be entirely the case; the Federal Communications Commission has devoted pages on its Web site to the issue and its chairman, William Kennard, has said, '[The Year 2000 problem] is more complicated, more serious and more costly than originally estimated.' Similarly the National Association of Broadcasters has highlighted the matter, working with the ITU and other organisations and companies in an attempt to provide as much information as possible.

Some broadcasters and studio operators believe that the Millennium Bug is being overstated. Others think that the facts have not been made plain enough.

Certainly, there are many areas of these two sectors where equipment or operations are date unambiguous and even if something does depend on time, then it can be manually helped through the changes or cached ahead. Like any scare, some will get reasonably rich from this, others may lose out. Perhaps it's best to make sure you're somewhere in between. Tick, tick, tick... ■

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Studio sound classified

b r o c h u r e
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a p p o i n t m e n t s

School daze

You may consider that you have left school behind, but schooling may not have left you behind. The march of the pro-audio educators continues writes **Dan Daley**

SEPTEMBER IS ALMOST HERE—it is back to school time. And while there is more to learn this year, there will be more places to learn it than ever before. There are an estimated 600-plus music technology and-or business programmes in place in the US, some situated in studios, others in academic institutions such as the 4-year BA programme at Middle Tennessee State University, and still others at dedicated facilities such as Full Sail in Orlando, and Berklee in Boston. But since many programmes are small and do not advertise widely, the actual numbers are probably 10% to 15% higher.

'But wait, there's more...' as the late-night commercials like to say. The European-based School of Audio Engineering (SAE) has said it will open its first US locations later this year —in New York and Nashville—and more are planned. And Gary Platt, who spent a decade working at Full Sail, has moved on, announcing that he will open a new facility in the San Francisco area, funded to the tune of an estimated $12m by a European investor.

The issue vis-a-vis formal education in professional audio is no longer whether facility owners and managers accept the notion. Up until a couple of years ago, most seemed at least uncomfortable with the idea, and many spurned it outright, saying that academic settings did not teach real life skills necessary in recording studios, and that 're-educating' graduates of audio programs was more trouble than it was worth. That viewpoint has retreated significantly. The increasing complexity and number of technologies and systems available have made exposure to them prior to entering the pro-audio workforce a necessity to a growing number of facilities. Howard Schwartz Recording, a major audio post facility in New York City, for instance, will not hire someone who has not been through some sort of programme. In addition, some schools report that as many as 40% of their graduates go directly from graduation into the ranks of personal studios. Thus, for some, their time in a school is literally the only exposure they ever get to what a conventional studio environment is like.

But now that more studios have come to accept the idea of formally trained entry-level staffers, the schools that offer these programmes—which have already become big business—are looking at their market sector as being poised for potential expansion. And that will change the way the schools approach the growing pool of would-be Phil Ramones, Butch Vigs and Puff Daddies.

One can already see some response to this pending expansion of the audio school market. Touting celebrities as guest lecturers has been a well-worn tactic at most of the larger schools and programmes, but in some cases the schools are putting them on the faculties, such as Blues Project producer-founder, solo recording artist and B3 master Al Kooper, who is now a full-time professor at Berklee. And Full Sail's upper management executives are taking the arrival of new schools quite seriously, considering toning down the carnival-like recruitment approach that the school has taken—sending out bus-loads of teachers and technology to cities and trade shows around the country to sell the MTV generation. They should take it seriously—Full Sail has an average of 1,500 students in its rolling admissions programme at any given time, each paying around $23,000 for the year-long course, that adds up to a minimum annual turnover of $34m. Other schools are becoming more proactive and media-savvy, retaining PR advisors and taking out glossy ads in not only trade magazines, but academic and youth-orientated publications.

Equally interesting is that these schools are doing well at a time when other aspects of technical education in the US are not. Businesses of all varieties are short on technically skilled personnel, which is good for wages and unemployment statistics (Tony Blair should be so lucky). But the number of US computer-science graduates has dropped

DVD: the revolution prompts threat of rebellion

The inability of the industry to correctly market DVD may result in the market taking DVD into its own hands writes **Barry Fox**

DECISION MAKERS in the US know pitifully little about Europe. Companies still hand out NTSC publicity tapes, oblivious to the fact that Europe is a PAL continent. Internet service providers wonder why usage is low in Europe because they still do not know that Europeans pay for local phone calls. The DVD-software industry risks killing DVD as a movie format in Europe by relying on Regional Coding to stop Europeans watching North American releases. If DVD-Movie fails, the recording industry loses out on dubbing and remixing. And there is little hope for DVD-Audio.

The Hollywood studios protest that they must have Regional Coding to support their staggered release pattern, with movies first sent out to cinemas across the US on 35mm film, then the prints collected, cleaned and exported while the American video release cycle begins. So foreign video release cycles have to lag behind America. Music DVD producers want to protect regional copyright deals and contract carve-ups. Coding is their easy fix, but it spells death to DVD in Europe.

Despite the Common Market, Europe does not function as one country—as does the United States of America. Instead, Europe is a loose collection of states, divided by disparate tongues, laws and social attitudes. So each DVD title for Europe must be released in three or four versions, to cope with the local language and censors. This compromises economy of scale in production. Most of the people living in Europe are perfectly happy with their VHS VCRs, and the picture and sound they get from rented tapes. They have no interest in a player that costs at least twice as much, does not record and offers better pictures and sound only if they invest in a better TV and surround-sound system. They have even less interest in the player when they realise the disc titles are already available on VHS at lower cost. The market for DVD, at least in the five years before we get disc recorders at prices that compare with VHS, lies with video and movie enthusiasts who indulge in home cinema.

If even a small percentage of the North American population buys a specialist product, the total number of sales will keep several factories busy. Translate that small percentage to the UK, France or Germany and

the real number is very low. Take it to Norway, Sweden or Finland and the total is tiny. Most enthusiasts in Europe already have LaserDisc players, hooked to multistandard TV sets. So they can either play PAL releases, or NTSC discs brought in from the US. Although local laws may ban the open sale of imported discs, there are any number of mail order or Internet suppliers only too willing to oblige.

These enthusiasts know that DVD can offer better quality than LaserDisc and multichannel digital surround in the European format. There is no provision for Dolby Digital AC-3 on PAL LaserDisc. So at least some LD buffs will be happy to upgrade to DVD, very likely buying the Pioneer player that handles either LD or DVD. Although DTS DVDs are arriving too late for the format to rival Dolby Digital in the mass market, enthusiasts seem to like the idea of DTS. It is another reason why they will upgrade from LD to DVD.

A European PAL DVD player will play American NTSC discs, putting out a local standard PAL signal. But only if the disc is coded for All Regions. If the discs are coded by region, European (Region 2) players will refuse to play American (Region 1) DVDs. And so far all American DVDs are coded to stop them playing in Europe. If uninformed owners of a European DVD player, used to importing Laser Discs, order American DVDs and cannot play them, they will turn against the new format. Informed owners, knowing the problem, will try to modify the player to defeat Regional Coding. On early DVD players this was a simple job, just flicking a dip switch or clipping the leg of a control chip. On more

from a high of 50,000 in 1986 to 36,000 in 1994, and the trend seems to be continuing. In fact, according to a National Science Foundation report, in 1995 30% of all R&D workers with doctorates in science and engineering were non-US born, as are a fifth of all undergraduates in computer-related fields and half of all doctoral candidates. Full Sail reported their foreign-born enrolment at about 12% to 14% last year and climbing. No wonder SAE and others are following the advice of bank robber Willie Sutton and going where the money is. (The true effects of this trend of overseas students will become noticeable in about three years, when your second engineer orders fried yak intestines for lunch.)

But the real upshot of a crowded audio-school market is a very positive one in the long term. And that is based on a theory that the Soviets never quite managed to grasp: increased competition breeds enhanced product quality. Curricula will likely become more focused, standards stricter and teaching abilities more intensely scrutinised. At the same time, more manufacturers will probably enter into alliances with a larger number of the major schools, hoping to build brand loyalties at the embryonic level, but simultaneously broadening the availability of technologies for students. And in the end, it is the studios that will benefit, from a better-educated pool of talent. The pro-audio industry is about to experience the real meaning of the old adage, 'A little learning goes a long way'.

recent players Regional Coding is controlled by factory-set software in the player, or by a remote control handset available only to accredited service engineers.

As I warned last month, the informed buff may simply buy a North American player and use it in Europe with a multistandard TV set. Sure enough, adverts in the European press are now appearing that offer American Region 1 players, and discs to play on them, at a price comparable to local prices for Region 2 players.

Time is of the essence, so let me spell it out again in the simplest possible terms. Once a European home has installed a Region 1 player, Regional Coding stops that family ever playing another European release. The European customer must continue to import Region 1 discs, even if the current thin choice of old and over-priced Region 2 titles turns into a bulging catalogue of new releases.

The best hope for DVD in Europe is that Regional Coding will be quietly abandoned. This can happen in three ways. Software producers can simply stop coding their discs, so that European releases play either on native Region 2 players or imported Region 1 units. Hackers may find a cure-all fix for the silly system, probably a DVD-ROM that reprograms the software controls inside a DVD player. Or the hardware companies, who never wanted Regional Coding in the first place, can produce players which have no coding control. Impossible? Not so—factory-fresh code-free players are already on sale in the Far East and now being advertised for special import into Europe.

Television Time Warp

Commercials are gnawing still further into US TV programmes, yet this comes as a revelation to its viewers writes **Kevin Hilton**

WE ARE ALL PROFESSIONALS here—so why is it that even the most hardened of us get all misty when talking about creativity in relation to commercialism? It's all 'Things are more linked to money than they used to be', or 'There is less creativity because things are so commercial'. Nonsense. People are being naive if they think that art, publishing and, most especially, broadcasting and the music business have not always revolved around the underlying desire to make bundles of cash without resorting to bank robbery or actually doing a proper job.

This line of thought came back to me during a recent trip to the US. Breathing heavily, walking up an incline in the Pacific Heights district of San Francisco, a headline came blaring out at me from under its glass protector: 'The television time warp—Networks' hour shows really are only 45 minutes long.' As badly kept secrets go, this is up there with Richard Nixon was a crook and George Michael is gay.

Europeans have long known for sure what Americans may have only suspected about their own TV programming: just because it is in an hour or 30-minute slot does not mean that the show itself is of those durations. When the hospital drama ER is shown on the UK's Channel 4, it barely makes it to 55 minutes—and that includes three commercial breaks and assorted promotional slots. The strict rules governing how much advertising can be shown during an hour reveals the extent to which the US networks are controlled by the advertising dollar.

'It is, after all, "commercial" TV,' continued the *San Francisco Examiner* headline, as I deposited my 25 cents and slid the paper from the kind of dispenser that is a familiar sight to anyone who has watched *NYPD Blue* in recent months. A survey commissioned by advertising trade groups reports that commercials now account for 11 minutes and 12 seconds of a programme hour (15 minutes with promos and public service announcements), as opposed to 9 minutes and 38 seconds in 1991.

A breakdown of the commercial channels does not present any major surprises. The network with the longer real programmes is CBS, filleting away only 14 minutes and 29 seconds for cloying nonsense about greetings cards or somebody shouting themselves hoarse for the virtues of Sports Utility Vehicles. NBC clocks up 15 minutes 19 seconds of ads, ABC 15 minutes 44 seconds and —paint me pink and call me Susan!—Rupey Murdoch's Fox clangs the bell with a whoop-ing 15 minutes and 54 seconds of kind of stuff that has nothing to do with Mulder & Scully finally working out that the Smoking Man is behind it all—again!

These figures start to give truth to the old joke that television programmes are now so poor that they are only there to fill in the gaps between the adverts. It was always obvious that commercial TV was about making money because of the adverts; and now it is getting closer for public service broadcasters as some of their promo slots are not so subtle plugs for parallel commercialism. This is underlined by a passage from a BBC Handbook of the 1930s that someone showed me recently, making the point that the Corporation had to think in a more commercial way.

All this made me wonder whether technology, and, specifically, automation has brought broadcasting to this point. Every time a TV station installs an automation system or upgrades its existing installation, the primary functions are for the insertion of commercials and promo slots. The programming is way down the list and is still largely played out from stand-alone VTRs, implying that it does not warrant that much cleverness because it is, after all, only programming.

All this made me wonder whether technology, and, specifically, automation has brought broadcasting to this point.

But if the ad departments of leading broadcasters had wanted to squeeze more spots into an hour, they would have found a way to do it, even with LMS-style video cart machines or banks of VTRs. The lure of currency is a strong one, something that could have easily worked around not having hard-disk recorders and video servers. The one piece of technology that can be blamed in part for this situation is the domestic remote control. This gives the viewer the power to stay rooted to the couch and zap around the channels anytime they wish. Steven Bochco, cocreator of *Hill Street Blues* and *LA Law*, has said he works on the premise that he has only 90 seconds—two minutes if he's lucky—to grab a viewer before they flip away to something else.

With more advertising and near-on the same break times, the only people who do not lose out are the advertisers. It is possible to flick around the main channels and get commercials each time. Because there are hardly any obvious junctions, you can slip back into a programme and suddenly realise that it was not the one you had been watching. That means that US network TV has done what Stephen Hawking can only dream of. It has rewritten the laws of time and space.

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Stereo mic techniques

Stereo is based on two simultaneous audio channels feeding two spaced loudspeakers.
John Watkinson explains the sonic delights of revealed stereophony

THE APPARENT position of a sound source between the two speakers can be controlled solely by the relative level of the sound emitted by each one. Signals in this format are called intensity stereo. In intensity stereo it is possible to 'steer' a monophonic signal from a single microphone into a particular position in a stereo image using a form of differential gain control. Fig.1 shows that this device, known as a panoramic potentiometer, or pan pot, will produce equal outputs when the control is set to the centre. If the pan pot is moved left or right, one output will increase and the other will reduce, moving, or 'panning' the stereo image to one side.

If the system is perfectly linear, more than one sound source can be panned into a stereo image, with each source heard in a different location. This is done using a stereo mixer in which monophonic inputs pass via pan pots to a stereo mix bus. Panpotted audio can never be as realistic as the results of using a stereo microphone because the pan pot causes all of the sound to appear at one place in the stereo image. In the real world, the direct sound should come from the true location, but reflections and reverberation should come from elsewhere. This is why artificial reverberation has to be used on pan-potted mixes.

The job of an intensity stereo microphone is to produce two audio signals which have no phase or time differences, but whose relative levels are a function of the direction from which sound arrives. The most spatially accurate technique involves the use of directional microphones which are coincidentally mounted, but with their polar diagrams pointing in different directions. This configuration is known variously as a crossed pair or a coincident pair.

Fig.2a shows a stereo microphone constructed by crossing a pair of figure-of-eight microphones at 90 . The output from the two microphones will be equal for a sound source straight ahead, but as the source moves left, the output from the left-facing microphone will increase and the output from the right-facing microphone will reduce. When a sound source has moved 45 off axis, it will be in the response null of one of the microphones and so only one loudspeaker will emit sound. Thus the fully left or fully right reproduction condition is reached at ±45 . The angle between nulls in L and R is called the acceptance angle and has some parallels with the field of view of a camera.

Sounds between 45 and 135 will be emitted out of phase and will not form an identifiable image. Important sound sources should not be placed in this region. Sounds between 135 and 225 are in phase and are mapped onto the frontal stereo image. The all-round pick-up of the crossed eight makes it particularly useful for classical music recording where it will capture the ambience of a grand hall.

Other polar diagrams can be used, for example the crossed cardioid, shown in Fig.2b. There is no obvious correct angle at which cardioids should be crossed, and the actual angle will depend on the application. Commercially available stereo microphones are generally built on the side-fire principle with one capsule vertically above the other. The two capsules can be independently rotated to any desired angle. Usually the polar diagrams of the two capsules can be changed.

In the SoundField microphone, four capsules are fitted in a tetrahedron. By adding and subtracting proportions of these four signals in various ways it is possible to synthesise a stereo microphone having any accep-
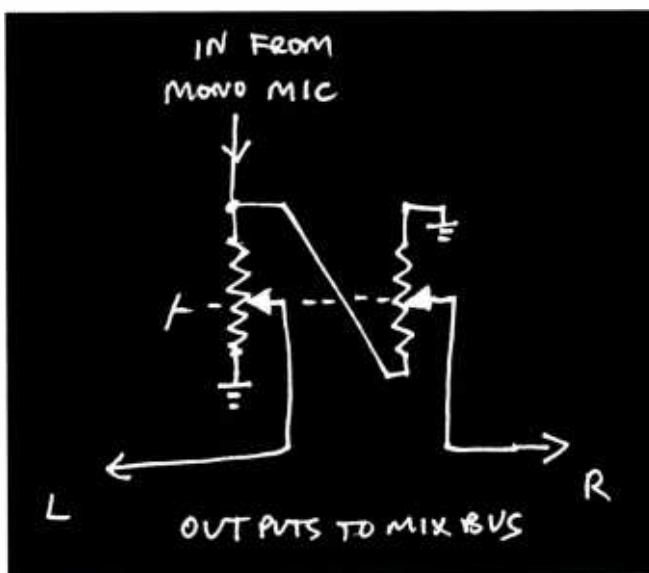
tance angle and to point it in any direction relative to the body of the microphone. This can be done using the control box supplied with the microphone. Although complex the SoundField microphone has the advantage that it can be electrically steered and so no physical access is needed after it is slung. If all four outputs are recorded, the steering process can be per-

formed in postproduction by connecting the control box to the recorder output.

Clearly the use of stereo will make it obvious if a microphone has been turned. In applications such as TV sound the microphone may be turned as a side effect of swinging a fishpole. This is undesirable and more precise handling techniques are necessary to keep the micro- **>**
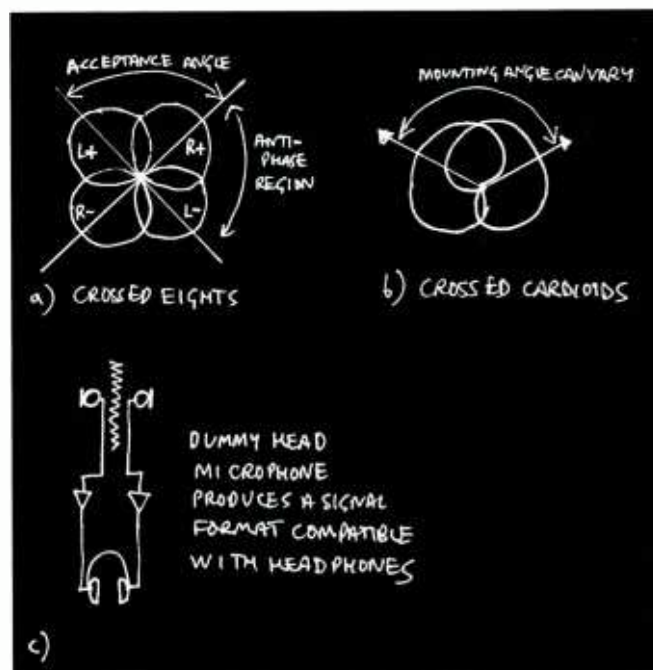

Fig.1: Pan pot locates a mono source at one point in a stereo image

Fig.2: Coincident and dummy-head formats are incompatible

< phone heading constant.

Intensity stereo is designed for loudspeaker listening and only works because both ears hear sound from both speakers. It is this double summing process that converts the amplitude differences at the speakers into time-of-arrival differences at the ears. Obviously the same result simply cannot be obtained with headphones because these prevent both ears receiving both channels. The result of monitoring intensity stereo signals on headphones is that there is no stereophonic image and the sound appears, quite unrealistically, to be inside the listener's head. Consequently conventional headphones are quite useless for monitoring the stereo image in signals designed for loudspeaker reproduction.

Highly realistic results can be obtained on headphones using the so-called dummy head microphone that Fig.2c shows is a more or less accurate replica
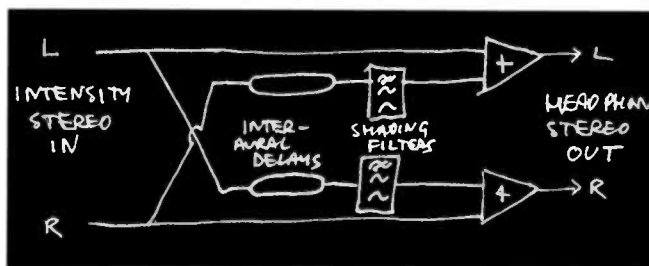

Fig.3: 'Shuffler' circuit is a standards convertor between intensity and dummy-head stereo

of the human head with a microphone at each side. Clearly the two audio channels simply move the listeners ears to the location of the dummy. Unfortunately dummy head signals are incompatible with loudspeaker reproduction and are not widely used. Headphones can be made compatible with signals intended for loudspeakers using a shuffler. This device, shown in Fig.3, simulates the cross-coupling of loudspeaker listening by feeding each channel to the other ear via a delay and a filter which attenuates high frequencies to simulate the effect of head shading. The result is a sound image that appears in front of the listener so that decisions regarding the spatial position of sources can be made. Although the advantages of the shuffler have been known for decades, the information appears to have eluded most equipment manufacturers who continue to provide headphone jacks which output incompatible signals.

Other microphone techniques will be found in stereo, such as the use of spaced omnidirectional microphones. Spaced microphones result in time of arrival differences and frequency dependent phase shifts in the output signals. When reproduced by loudspeakers and both signals are heard by both ears the location principle of intensity stereo simply cannot operate.

Consider a continuous off-axis source well to the left. This will result in two

microphone signals having a small amplitude difference, but a large time difference. If, initially, it is assumed that the source frequency is such that the path length difference is an integer number of wavelengths there will be no phase difference. This will be reproduced by two speakers as a source slightly left of centre because of the small amplitude difference. If the frequency is reduced slightly, the wavelength becomes longer and the right microphone signal will begin to lead in phase, moving the phantom image right. If the frequency is increased slightly, the wavelength becomes shorter and the right microphone signal lags, moving the image left. Consequently, the apparent location of a source is highly frequency dependent and the result is that all sources having a harmonic content appear to be very wide. Where the distance between microphones corresponds to an odd multiple of a half wavelength, the two microphone signals are out of phase. As both ears hear both loudspeakers the result is a dip in frequency response and no image at all. This comb-filtering effect is a serious drawback with spaced microphone techniques and makes monophonic compatibility questionable. A central source will give identical signals which create a central phantom image. However, the slightest movement of the source off-axis results in time differences which pull the image well to the side. The resulting hole-in-the-middle effect is often counteracted by a third central microphone which is equally fed into the two channels.

This technique is not strictly stereophonic reproduction because it fails to portray the spatial attributes of the original sound. However instead a spacious effect is the result and many people find it pleasing. As there is no scientific basis for the technique it is hardly surprising that there is no agreement on the disposition of the microphones and a great many spacings from a few centimetres to a few metres are used. One rule of thumb that has emerged is that the spacing should be no more than one third of the source width.

When listening on traditional loudspeakers which suffer image smear due to cabinet diffraction, the difference between intensity stereo and spaced-omni recordings is not very great. Perhaps when spaced microphone techniques were first developed loudspeakers were even poorer in performance and there was no perceived difference at all. This is no longer the case. ∎

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Desktop audio: quo vadis?

The epoch of discrete boxes performing specific audio operations is drawing to a close. **Frank Hund**, founder and president of CreamWare, ponders on the realities of the mighty host processor and 'native' processing

THE LAST FEW YEARS have shown us how quickly computer technology develops and how far it has extended into all areas of audio recording and production. For just a few thousand dollars, anyone can obtain computer-based multichannel systems with advanced software tools for editing, sequencing, and signal processing. In fact, a major achievement of this new technology has been to provide even the hobbyist musician with the same, or perhaps better, digital tools than those that were available only to high-end recording facilities a few years ago. The same goes for journalists in the broadcast industry—monitoring and editing audio on the desktop computer has become economical and common practice. But could we see an equal increase of productivity for the professional audio engineer? Well, not to the extent that desktop-audio technology has helped the amateur musician.

No doubt technology will continue to bring consumer and professional technologies closer together. And no doubt we will continue to see structural changes in our work environment: the information age and the digital revolution have just begun. In addition to the need for new business strategies, it has become crucial for the professional audio facility to use the latest technologies in order to maintain its productivity and quality advantage over the amateur studio. But where is it all going? What will the trends be over the next several years? How will professional computer-based audio technology develop and how fast? Will DSPs play any role in the future or will it all be 'native' processing? It is time for a brief analysis of the current status and further development of 'native' desktop audio systems under professional measures.

Looking at the breathtaking speed of processor development it is easy to predict that, thanks to ever-exploding MIPS and FLOPS specifications, desktop audio and its potential will advance just as fast. Many in the industry expect that, even in the near future, the almighty host processor will take care of everything—all running on that $1,000 machine from the computer discounter down the block. This sounds logical, but is it a realistic scenario for tomorrow? Will we all be using cheap, standard computers for all our audio needs

in just two or three years?

There are a number of people who, even in 1998, continue to use the Atari ST computer they bought about a decade ago for MIDI sequencing. If you ask them why, they mention points such as better reliability and better MIDI timing. Although processor speeds have increased by a factor of roughly 50 to 100 since then, it appears there are other criteria that offset the impact of gains in processor speed. Another often quoted example: if you compare a 486 machine running a word processor from 1993 with the latest Pentium running today's Word v8.00, you will not feel that you have gained a lot. The software takes about the same amount of time to load, and it takes about the same time to print, so where has the speed gain gone?

> The good news: many interesting modules have been, and continue to be developed in the industry. The bad news: there is no way to get all those elements working tightly together on today's general purpose computers

Now look at the development of audio applications. The idea of the 'studio in a box' has been with us a while now, and is making sure progress on the software side. But the current software-only packages are still not able to replace professional audio gear in terms of productivity. This is because in a professional environment, concepts such as dependability, availability and access are as important as the audio quality. The ideal world of pure processor speed does not recognise where the truly limiting factors in system development lie: in the architecture of a computer that was made for general purpose and office applications in the first place (this is valid for both Mac and PC) and in the structures and architecture of the application software itself, coping with the given computer environment.

The major weakness of today's computers with regard to audio processing is the operating system. As audio processing is a highly resource-intensive,

real-time signal-processing task, software developers have had to struggle, and will continue to struggle, with the limitations of the operating system's environment—limitations imposed by the ever-present focus on multitasking and multiple-user capabilities.

Here lies a fundamental problem: there is a general 'conflict of interest' between the requirements for real-time signal processing and the requirements for general-purpose multitasking as designed into the system's architecture. While signal-processing tasks require precisely timed and guaranteed time slots to execute with as little latency as possible, the common multitasking model tries to find the best compromise by implementing a complex priority management system. In doing so, the operating system's overhead (the time required to switch from one task to another: the time required to respond to an interrupt request, for example) has increased and become less predictable over the last few years. This led to the phenomenon that a computer that is twice as fast as another still exhibits the same latency timing problem as the slower machine. It is important to understand that without structural changes to the operating system, basic problems of implementing real-time signal processing on a general purpose machine cannot be resolved—regardless of the host processor power.

A number of other improvements can only be made within the audio application software itself. What do we expect from desktop audio in the future? If we're talking about the 'studio in a box', we expect to see all the elements of today's production environment including sound synthesis, free and flexible signal routeing, the integration of all the great algorithms implemented by well-known audio-processing companies, and the ability to integrate all our existing hardware studio components. We wish to work in a stable, configurable, modular environment in which all of today's software related problems involving delay, latency, and phase problems have been eliminated. We also expect to see improvements and changes in user-interaction—the integration of new hardware controllers to enable work to proceed as efficiently as with a dedicated hardware setup.

Responding to requirements such as those mentioned above is the actual task of the industry. In the effort to unite all the elements of today's studio, software developers must follow an abstract, object-oriented and modular course that leads inevitably to increased overheads, which is further complicated by the ever-increasing overheads of the operating system. This is the reason you can usually rely on new versions of programs to become bigger and slower, forcing you to buy a new computer just to be able to run them. If we rely >

< solely on the host processor, we have a long way yet to go before we approach a 'native' desktop solution that can replace today's professional hardware equipment.

Of course, the marketing machines of the different software manufacturers won't tell you this. It is up to you to find out by seeing how their products get you where you want to go. These days money is being made in the amateur and semipro segments—and software producers concentrate their development and marketing energies in producing software-only products in order to acquire market share. And, of course, they use the picture of the almighty host processor as this is has a very efficient marketing impact. By following this strategy they fail to recognise and address the really important technical issues that could help us get closer to the new age professional studio. However, as long as they are making money and acquiring market share in the large consumer segment, the technical innovations will wait.

We must take into account that the market rules the strategies of both Microsoft and the audio software manufacturers. Also, software development cycles have always been much longer than hardware cycles. No matter how fast the host processor gets, until certain basic problems on the software side have been solved or improved, we will not see a truly professional and complete audio studio on a standard computer.

The good news: many interesting modules have been, and continue to be developed in the industry. The bad news: there is no way to get all those elements working tightly together on today's general-purpose computers. The current software interface standards, such as DirectX, are simply not suited to building a system that can be called 'professional'.

No doubt we will see a massive shift from hardware solutions to software solutions in the future. And this development is truly exciting as it will offer new possibilities and help make our lives easier and our work more creative. But, no matter how much we anticipate and welcome the software age, it is not likely that host-processor-based, software-only solutions will take over professional production within the next 5 to 8 years.

Still, for the next several years the use of dedicated signal-processing hardware (DSP architecture) will be the only way to avoid latency and phase problems. It is the only way to guarantee performance and the availability of applied software modules, simply the only way to fulfil professional criteria. The best approach for the next decade is to use the computer for user-interaction, visualisation, sequencing and editing—and leave the actual signal processing to dedicated hardware.

This may sound conservative, but it is rather just a technical detail that delivers results when developing and improving desktop-audio technology. After all, technology is about progress, and such progress is possible. The speed of DSP processors and architectures is increasing similarly to general-purpose host processors, and prices for DSP chips are falling just as fast as they are with the host processors. And DSP systems have never had any of the latency problems because DSP-based architectures typically process audio in a cycle of single sample words, while all native processing must employ buffer structures, leading to latency and high RAM memory requirements. The DSP type of processor is certainly here to stay, and the future of audio innovations is with companies and developers that realise that, regardless of the hype surrounding 'native' audio processing, there is still specific professional demand that should be better answered.

What does all this mean for the professional audio engineer? Make sure you use technology to secure your competitive edge. And be sure to keep informed, so you can distinguish marketing driven hype from actual technological advances. Technology will change the way we work, but not as fast as Intel announces new products. ■

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Room for agreement

Debate over the concept of surround-capable control rooms must soon give way to agreement on their specifications, argues **David Bell**, director of White Mark

HAVE JUST RETURNED from the AES exhibition in Amsterdam where most of the efforts by exhibitors were to persuade the pro audio industry to support surround sound, and, unfortunately, all its variations of implementation, was clear.

This is an important moment for the industry, with a commercially viable opportunity to develop itself into a market that is technologically supported in the consumer area, and that offers artistic potential for creativity. Consequently, I had hoped to come away from the show with a clear sense of the way forward, a direction in which to go in support of those active in the recording and mixing of material for this medium.

Engineers are often correctly criticised for considering the scientific principles involved in a problem and pronouncing them as 'absolute' truths that should be used to shape any subsequent development. I have never been a member of this school of thought. If a professional sound engineer hears something, it is, I believe, the role of the scientist to try to understand and explain it. It is not the role of the scientist to set parameters within which listening should be carried out and thereby limit the developments possible to those parameters that emerge from study.

The issues seem to be relatively easy to identify, but difficult to tie down without practical experience. The whole development process is at the stage where little surround mixing has been done, certainly as a creative tool in the origination of music composed and created specifically for surround sound. There are, however, many requests from the equipment suppliers and system vendors for material to help them cash in on the new market. This has generated much pontification from on high as to the types of loudspeaker that are needed, the nature of the control room required; and the coding systems best suited to the whole process. The worrying aspect of this is the motive behind each impassioned claim to the high ground of surround sound truth.

The support of those willing to work in this area is required to include careful appraisal of results achieved with the various methods used by different practitioners, and gradual iteration towards recording and monitoring approaches that work, and that produce results that satisfy artists, producers and engineers. We must realise that the new medium can only be successful if the consumer is attracted to it, and accepts the need for the extra expenditure on equipment for its reproduction.

However, those willing to make the investment in facilities to do this work also need the support of those who wish to use such facilities. Studio owners who have the courage and are prepared to build rooms for surround mixing need advice from those who may wish to use them in order to meet their requirements.

London's Strongroom had the courage to build Studio 2 as a full 5.1 soffit-mounted surround room, and, following our involvement, we have worked together on understanding surround monitoring. Feedback, from studio users and demonstration attendees have been considered, along with detailed measurements taken in the room. White Mark has also worked on other facilities on surround issues including the Hit Factory's Studio 5; SSL's demo room; and four of the eight post rooms we have under construction in Soho, London.

Direct input is needed now from producers and engineers so owners who wish to develop their studios, and commit considerable money to providing facilities, can have confidence that people will want to work in them. Workshops held at the show succeeded in garnering agreement that all loudspeakers in the system should be equal in performance, but no clues as to positioning in the room, or other issues, were revealed.

In laying out real rooms, consideration must be given to the overall performance of the space from decay time, diffuse reverberation creation, early reflection suppression and stereo symmetry aspects. Doors consoles, outboard and sofas need integration, windows to allow artist-engineer contact need to be positioned. Add to this the requirement that five loudspeakers are to be soffit-mounted together with a near-field system and the complexity of the problem emerges.

We at White Mark have our views on the solutions to these problems, but we would like to hear from practitioners. Answers to such questions as loudspeaker placement and type; multistandard compatibility; sub-bass usage and positioning; and monitor control must be advanced by the recording community so that international standards committees and certain commercial interests do not legislate out the creative possibilities offered by the new medium of surround sound.

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