EXCLUSIVES

HHB CDR800
Soundscape V2
Drawmer MX30
TL Audio 3013
Elation KM-201
Otari PD-80
Garwood ADA8
Sony PCM-R700
Focusrite Green 4
Amek 9098 dynamics

TOTAL

BENSON & HEDGES

PEUGEOT

BENSON HEDGES

B&H

FORMULA ONE
Winning pit-stop production

The
ALBERT
BROTHERS
Interview

SOUTH AFRICA BREAKS FREE
POWER RANGERS POST
DYNAMICS: THE GUIDE
96KHZ CLASSICS

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**Russian Elation; 24-96kHz;
Power Rangers get tough; an
affordable dynamic revelation**

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INFINITY

Solid State Logic
Oxford · England

Interchange again

MY COMMENTS LAST MONTH on the requirement for manufacturers and users in the postproduction community to start some sort of serious dialogue concerning interchange seem to have pricked a few nerves. Responses from both camps have displayed a polarisation of opinion.

There are users, predominantly outside the traditional super technology league, who are keen to integrate themselves by whatever means, including interchange, in to any production flow. Then there are users, closer to the end of the food chain, who can appreciate a need for moderate import from lesser beings while the single brand facility house is often amused by the concept of interchange because it is involved from the beginning to the end of productions, and has its own means of moving the stuff around. By definition there are more in the first category than there are in the second, and more in the second than in the third.

Manufacturers can also be classified into three general types: those that want to play and are happy to, those that like the idea providing the material is only coming into their systems, and those that support the principle but are quietly confident that it won't add up to much because they provide complete systems solutions with limited need for outside contact.

Clearly the stratification of response comes down to the type of equipment used, and where it sits in the DAW hierarchy. Could this mean that the real drive for interchange will emerge from the middle facility market ground? Could this mean that it will be the middle market DAW manufacturers that summon the courage to drop their trousers and reveal their personal protocols in the interest of interchange? It looks increasingly as if it could.

If it does then an interesting situation may develop where a unified and communicating market will be book-ended by an isolated lower end and a top-end elite. Is this really in the interests of everyone?

Zenon Schoepe, executive editor

Model theory

IT WAS WHILE sipping beer in Colombia's Galle Face Hotel that I overheard a British man extolling the virtues of vocational training over more traditional academic certificates to a Sri Lankan woman. 'In America you can buy a degree for $10,' he quipped. 'Have you heard about NVQs—National Vocational Qualifications?'

I'm not certain that the object of the selling exercise was the NVQ. In fact, I'm pretty certain it wasn't, but the kind of real-world controversy that currently surrounds pro-audio education played no part in his revisionist vision.

It further transpired that our man and his female accomplice passionately believe that there is no need to devise new 'models' of learning because there is always an existing 'model' somewhere that can be adopted. Yet it will be evident to anyone who has seen the Sri Lankan railway system that this approach doesn't always work. Miles of broken telegraph poles alongside the railway track through the hills and the fact that the passing of a train only temporarily returns the track to its intended use before it reverts to being a footpath challenge this hypothesis. Yet the establishment of a railway system following the British 'model' fit perfectly with it.

I wonder what sci-fi author and technical visionary Arthur C Clarke would have had to say about the conversation had he been the eavesdropper—not so unlikely since he completed his Space Odyssey series while resident at the Galle Face. I suspect he would have agreed with me that the world has little room for the kind of arrogance that accompanied the spread of the British Empire. Modern communications and media are changing the world in a way beyond any previous means of communication. Consequently, in pro-audio as well as a wealth of other areas, new models of working, thinking and learning are essential if we're to embrace the future—just as they have been at points in our past evolution. The Englishman attempting to repackage the past is a dangerous animal, regardless of whether he's working in postproduction or Pamper production. Beware his call.

In the meantime, I'll check the on-line editions of Sri Lanka's *The Island* and *Daily News* newspapers before deciding whether his disinformation effort is having any discernible effect.

Tim Goodyer, editor

A B C
AUDIT BUREAU OF CIRCULATIONS

BUSINESS PRESS

PPA

POST PRODUCTION

Tuesday, 7 pm, and the end of a hard day's editing. The program goes out at eight and the producer wants to change a sensitive line of narration - disaster. Not with O3D. The artist is ready. Simply recall the channel strip library you set up for him - EQ, compression, aux levels to on-board 32-bit reverb are all set at the touch of a button. Automix will take care of any slight level changes. Job done.

RECORDING

Monday, 9 am, and last week's session needs more work. It's mid afternoon by the time you've re-created the previous mix. Unless, of course, you have O3D. In seconds you've recalled the exact position of every motorised fader, channel mute, bus level, aux send and multi-effects parameter as well as four band parametric EQ and a dynamic processor on every input and output.

Montreux Cinema Festival

SWITZERLAND: The 23rd-28th April saw the 10th International Electronic Cinema Festival take place in Montreux. Alternatively held in Montreux and Chiba City, Japan, this festival is now a spotlight on HDTV productions worldwide.

Ostensibly a bridge between standard television and the cinema, high-definition television seems to find more outlets in the world of cinema at present, rather than that of broadcast, and things will not be easier with the advent of digital television and advanced television systems. Six countries were represented with the majority of the 27 entries coming from Japan. Possibly not surprisingly, five of the seven awards went to the same country (including the Grand Prix Astrolabium for *Osamu's Summer*), with the other two going to Germany and an American-Japanese-French production.

The credits for sound only ever mentioned one or two tracks (if at all), with the exception being *Osamu's Summer* which used four. The productions that I saw did not shine in terms of audio and a lot of EQ'ing was done in the cinema auditorium in order to make dialogue more intelligible and give more depth to the music.

This would appear to indicate that most of the attention is still going to the picture, even though HDTV is trying to compare itself to the cinema. The mixing techniques for high-definition soundtracks should certainly be inspired from film and the studios should owe more to dubbing stages than small control rooms.

There still appears to be an on-going love affair with picture rather than looking at the medium as a whole and realising the effect of sound and image combined is far greater than the sum of the two parts. Film-makers have known this for a long time.

The decision-makers behind HDTV are going to have to come to grips with the fact that sound and vision together are going to make things happen faster—good pictures are not enough on their own.

Terry Nelson

Asian action

HONG KONG: The ceremonies accompanying the end of the British 156-year sovereignty of Hong Kong and its return to Chinese control found a custom American Meyer Sound public address system in the thick of the action—and the rain. The system was designed and operated by the founder and MD of Australia-based System Sound, John Scandrett whose main concerns after the audio quality and functionality were to remain unobtrusive in an event being witnesses by 40,000 people and covered by the majority of the world's television media. The audience for the harbour canto-pop concert that followed the ceremony commanded a further live audience of around 10,000. Other musical activities involved the Hong Kong Philharmonic Orchestra and Hong Kong Chinese Orchestra with orchestral mixes being fed directly to the

BBC for world broadcast.

Meanwhile the Chinese capital Beijing—as well as filling the world's largest public square with 100,000 revellers in recognition of Hong Kong's new status—has filed reports of a particularly successful Pro Audio, Light & Music China show. Focused on the Chinese market, PALM saw 16 other countries displaying and demonstrating their technology to 19,832 attendees. The 1997 event paves the way for 1998's PALM show which will enjoy a merger with China Sound & Light and is scheduled for 9th–12th May and will take place at the Beijing Exhibition Centre.

Further south in the Asia-Pacific region, Jim James's Singapore-based EastWest operation has changed its name to reflect its focus of operations on media relations, event management and direct marketing. Included in EastWest's portfolio is the imminent Pro Audio & Light Asia event in Singapore.

IIR Exhibitions. Tel: +65 227 0688. Fax: +65 227 0913.
EastWest Public Relations, Singapore. Tel: +65 222 0306.

▲ **UK-EUROPE:** Recent landmarks in the development of high-sample rate recording include the first 24-bit, 192kHz stereo session. The recording of a number of choral pieces took place at Queens College Chapel in Cambridge, England, on 26th June and involved a selection of Sondore omni microphones, custom dCS A–D convertors feeding a customised Genex GX8000 MO recorder (six of which were recently used for a 20-bit, 48-track location TV recording by the A[2]D mobile of Puccini's *Manon Lescaut* at the Glyndebourne Festival). The session was undertaken by Dutch classical production company, Kompas in collaboration with dCS to provide electronics giant Samsung with suitable material for an approach to the DVD consortium over audio standards. A full report will follow in *Studio Sound*. A 4-channel 24-bit, 96kHz recording of the Montreux Jazz Festival was conducted by MTX Digital Mastering of Switzerland. The setup again used dCS 904 A–D convertors this time feeding a Pyramix Virtual Studio workstation. The Pyramix system is already 24-96 capable and also provides the platform on which the recordings will be edited. The artists being recorded included Eric Clapton, Marcus Miller, David Sanborn, Eric Sample and Steve Gadd; the recording will be used for a forthcoming DVD audio release.

dCS has announced its 972 digital-to-digital convertor (pictured above) which allows down-conversion of high-quality digital audio to have a narrower wordlength and-or lower sample rate. Interfacing is comprehensive and includes AES-EBU, dual-AES, SPDIF and SDIF2 and conversion may be either synchronous or asynchronous. The 972 complements dCS' 902D and 904 A–D convertors which are fully 24-bit, 96kHz capable.
dCS, UK. Tel: +441799 531999.
Merging Technologies, Switzerland. Tel: +41 21 946 0444.

◀ **MALAYSIA:** RTM, Malaysia's national broadcaster, has set up a new recording studio at its HQ in Kuala Lumpur. The facility, which was designed and built by Roger Hayler of Hong Kong-based ACE, will be used exclusively for music tracking. The equipment listing includes an SSL 4000-series console, Genelec 1035B main monitors, Genelec 1030 close-field monitors and 1038 foldback speakers. Even the vocal, drum and string iso booths boast Genelecs—1031As.

www.americanradiohistory.com

▲ UK: The Forge studio has opened in Oswestry, Shropshire in the all-new complex of pro-audio company Systems Workshop. The studio will run alongside the company's sales and distribution operation, and its sound reinforcement hire and installation service in the listed building which dates back a couple of hundred years. Originally a Smithy, the building has been reconstructed from the ground up on a prime site in the centre of the ancient market town. Designed by Sean Zoega with Dynaudioacoustics M3 monitoring, the studio houses an Amek Rembrandt console, 32 tracks of Tascam DA-88, and the sort of outboard rack that includes only what the UK distributor for Tube Tech can. 'I decided that if the dream of my own studio was to become a reality it had to be done properly and we've put ourselves firmly on the map in the North West of England with a sensibly priced control room and large studio area, but at a standard more normally associated with far more expensive facilities,' says Systems Workshop MD Phil Beaumont. 'The complementary studio, sales and sound reinforcement operations is already proving to be a strong combination—studio clients get to use gear that they otherwise wouldn't which leads to sales, the sales operation can book the studio to serve as the demonstration suite to end all others, and if studio clients want to play some gigs we can take care of that as well.'
The Forge. Tel: +44 1691 658550

Alan Parsons

UK: Alan Parsons takes over as Vice President of the EMI Studios Group from Martin Benge. Who has returned to Australia for family reasons. Parsons, who spent many years as an engineer at Abbey Road during the 1960s and 1970s before pursuing a career as a producer and artist, becomes responsible for Abbey Road, Town House and Olympic Studios plus four Manor Mobile trucks.

'I was absolutely flabbergasted when Martin Benge contacted me about the job, I fully expected him to be there until he retired. I had a long think about it and decided that basically the opportunity would not arise again—I'll be 50 next year and I think the timing is absolutely right,' says Parsons.

'I realise that I have a lot to learn in the world of business and I'm very determined to make it work. However, I firmly believe that a lot of running studios successfully is about relationships and common sense and having worked on both sides of the glass, I think I'm able to see things from quite an objective standpoint.'

Some initial parallels have been drawn between the Parsons appointment and George Martin's figurehead role at Air Studios. Seeing it as a valid comparison, Parsons comments that 'putting some-one who's in the public eye at the helm can do no harm at all. It also plays a significant part in attracting foreign work, and with the London studio business being so international at the moment, it makes very good sense,' he adds.

DVD and multichannel surround formats are areas that he is particularly enthusiastic about. 'Having had experience with surround sound both now and in the 1970s, I believe it deserves to be a commercial success this time around. If there's anything I can do to popularise it, I'll try my damnedest to do it,' asserts Parsons.

Benge will return to Australia in August where he will operate as an industry consultant.

Calling Parsons 'the obvious candidate', Benge says he wanted someone who came from the creative and technical side of the business rather than the financial planning-business management area. 'I'm absolutely delighted that Alan has accepted the position. His background as a producer and engineer plus his early history at Abbey Road is going to stand him in very good stead. Also his prominence in Repro, the British Producers Guild and the APRS give him a broad perspective of the industry, and the fact that he has been studio client himself puts him in a unique position. I'm sure he will do a great job.' Patrick Stapley

■ Tokyo-based Imagica has ordered a second AMS Neve Logic 2 console for August installation in its No.5 MA Studio. The console will be used for surround-sound and high-definition television production work. Other Japanese action sees two Fairlight MFX3 Plus 24:24 systems being delivered to post house Qtec, a member of the Pioneer group; national broadcaster NHK purchasing a 38th Fairlight FAME system for its Studio CC500; and Fuji Television's Tokyo-based Tele Pro has also subscribed to FAME.
General Traders, Japan.
Tel: +81 3 3291 2761
Fairlight ESP, Australia.
Tel: +61 2 975 1230
■ Cologne's Sound Studio N has chosen a Sony OXF-R3 'Oxford' digital console to partner an existing SSL 9072 analogue desk. It is the fifth Oxford sale to date. The decision accompanies the appointment of HGA's Neil Grant to install both the console and a pair of Boxer T-5 main monitors in Studio B. The new room will provide tracking facilities for music production ranging from jazz to pop and radio broadcast.
Sony, Europe. Tel: +44 1256 55011
Harris Grant Associates, UK.
Tel: +44 1753 631022
■ California's Monterey Post has installed an Oram BEQ Series 8 console in its main recording room. The studio, which has enjoyed considerable success with animated features such as Superman and Batman, is currently working on Pinky and the Brain. Elsewhere in Burbank, Warner Brothers Post Production Services has announced the order of two AMS Neve Logic Digital Film Consoles for use in long-form television production. The identical consoles offer 88-inputs, 72-faders, assignable joysticks and are intended for two-operator use. The confirmed DFC order contradicts earlier reports of an order for two SSL Axioms.
Meanwhile, the California post take-up of KRK monitors continues with five pairs of M7000B/S and three pairs of shielded KRoKs going to Universal Studios; Mad River Post taking two pairs of shielded M6000/S; Creative Cafe taking five pairs of M9000B/S and M6000/S; and Digital Kitchen taking a pair of M7000Bs.
Oram Professional Audio, UK.
Tel: +44 1474 815 300
KRK Systems, US.
Tel: +1 516 249 1399
■ London's Precinct 23 has installed a Soundtracs Jade console. Installed by Don Larking Audio, the 40-channel Jade has VCA and moving fader automation, and assignable dynamics and will serve a variety of work for owner Ronin Productions, including TV commercials, remix work and Ronin's own recording career as 23 Skidoo.
Soundtracs, UK.
Tel: +44 181 388 5000
Don Larking Audio, UK.
Tel: +44 1234 772244
■ Chinese state broadcaster, China Radio, has ordered APT digital audio

codecs for coverage of the hand-over of Hong Kong to China this month. The DSM100 codecs will serve some 8,000 journalists and ensure quality worldwide broadcast facilities. APT's ADK200 digital audio coding systems are also being used for prerecorded announcements on Hong Kong's airport rail link. Other East Asian sales include a slew of Focusrite signal processors to Thailand, among others, the Kantana Group film production company has taken two Red 7 mic preamps; Pink Panther music studios has taken a Red 2 dual EQ, Red 3 dual comp-limiter and a Red 6 preamp-EQ; and Peter Pan mastering has taken a Red 6, Blue 315 and Blue 330.
Audio Processing Technology, UK.
Tel: +44 1232 371110
Focusrite, UK.
Tel: +44 1494 462246
■ California-based producer Bruce Botnik has bought eight Martech MSS-10 mic preamps. Botnik's preferred setup involves a variety of microphones including Neumann M50s and M49s, and AKG C12s; dB Technologies AD-122 and Studer D19 A–D convertors; and Sony PCM-3324 and Genex GX8000 recorders. The MSS-10s have already been used to record Jerry Goldsmith's score for The Edge and the opening theme for Spielberg's The Lost World with more film work to follow.
Martinsound, US.
Tel: +1 818 281 3555
■ Belgian Radio and Television has ordered two Amek DMS consoles for its newly-refurbished radio studios at Broadcasting house in Brussels. BRTN intend to have the installation up and running in September when the 72-input, 24-fader digital consoles will be used for live programmes, such as networked sports broadcasts, where their reconfigurability is regarded as invaluable.
Amek, UK. Tel: +44 161 834 6747
■ Chicago-based independent television broadcaster and 'cable-caster', WCFC-TV38, has installed a new Euphonix CS2000P console. The move follows viewing another Euphonix broadcast installation, WFLD Fox 32, and the desk will handle live production and postproduction duties. A further CS2000 has been installed at Stockton, California-based Studio-C as part of a complete studio redesign which also includes the purchase of Manley Mu compressors, LA-2A limiters and Mid EQs; and three Alesis ADAT XT MDM recorders.
Euphonix, US. Tel:+1 415 855 0400.
Manley Labs, US.
Tel: +1 909 627 4256.
Alesis Corp, US.
Tel: +1 310 558 4530.
■ Czech broadcasters Radio Bonton and Czech Radio have taken delivery of DDA consoles. A 24-input FMR desk has gone into Radio Bonton while a DCM has gone into Czech Radios Studio Macac—both were supplied by Mediatronik and are for production applications.
EVI Pro Audio Group, UK.
Tel: +44 1562 741515
■ Vancouver's Ocean Studios, owner of seven DAR SoundStations, has upgraded two more to the Sound-Station's Sigma specification.
DAR, UK. Tel: +44 1372 742848

July

SMPTE 97
Darling Harbour,
Sydney, Australia.
Tel: +61 2 9976 3245.
Fax: +61 2 9976 3774.
Email: conference@peg.apc.org

International Television Festival & Market of India 97
World Trade Centre, Mumbai
(Bombay), India.
Contact: Kavita Meer.
Tel: +91 22 215 1396.
E-mail: saicom@bom2.vsnl.net.in

New Broadcast Standards and Systems
Eighth Residential course at
the University of Surrey
Contact: Neil Sharp.
Tel: ++44 171 344 5421.
E-mail: nsharp@iee.org.uk

Lecture. Cochlear Implants
Conference room, Baden Powell
House, South Kensington, London SW7.
AES, UK.Tel: +44 1628 663725.
Email: AESUK@aol.com
Fax: +44 1628 667002

Pro Audio & Light Asia 97
World Trade Centre, Singapore.
Tel: +65 227 0688.
Fax: +65 227 0913.

September

PLASA 97
Earls Court 1, London, UK.
Contact: Marcus Bernie
Tel: +44 171 370 8231
E-mail: marcus.bernie@eco.co.uk
Net: www.harvard.co.uk

MIDEM Latin America & Caribbean Music Market
Miami Beach, Florida
Contact: Jane Garton
Tel: +1 33 1 41 90 44 39
Email: jane_garton@midem-paris.ccmail.compuserve.com
US tel: +1 305 573 06 58
Net: www.midemcannes.com

IBC 97
RAI International Exhibition and
Congress Centre, Amsterdam
Tel: +44 171 240 3839
Email: show@ibc.org.uk
Net: www.ibc.org.uk/ibc/

Infocomm Asia
Singapore International Convention &
Exhibition Centre
Tel: +1 703 273 7200
Email: icia@cia.org
Net: www.usa.net/cia

19th Annual Satellite Communications Expo & Conference (SCEC 97)
Washington Convention Centre,
Washington DC, US.
Tel: +1 800 288 8606.
Fax: +1 913 967 1900.

TVLink Brazil 97
International Trade Mart,
San Paolo.
Tel: +44 181 910 7841.
Fax: +44 181 910 7813.
Email: zoe.cano@reedexpo.co.uk

AES 103rd Convention
Jacob K Javits Convention Centre,
New York, US.
Tel: +1 212 661 2355.
Fax: +1 212 682 0477.
Net: www.aes.org

October

18th Nordic Sound Symposium
Bolkesjø, Norway.
Contact: Seem Audio:
Tel: +47 66 98 27 00.
Fax: +47 66 84 55 40.
Email: soundsymp@nrk.no
Net: www.nrk.no/soundsymp/

7th Intermedia 97 Music Expo
Hala Ludowa, Wystawowa Street 1,
Wrocław, Poland.
Tel: +4871 481821.
Fax: +4871 481451.

International Audio, Video, Broadcasting and Telecommunications Show (IBTS 97)
Milan Fair, Porta Metropolitana
Pavilions, Milan, Italy.
Tel: +39 2 481 5541.
Fax: +39 2 49 80330.
Email: MC1703@MCilnk.IT

Asia Cable, Satellite & Broadcast 97 (ACSB 97)
Putra World Trade Centre, Kuala
Lumpur, Malaysia.
Tel: +6 03 264 5663.
Fax: +6 03 264 5660.
Email: acsb@mfsb.po.my

Reproduced Sound 13
Hydro Hotel, Windermere, UK.
Tel: +44 1727 848 195.
Fax: +44 1727 85 0 553.
Email: Acoustics@clus1.ulcc.ac.uk
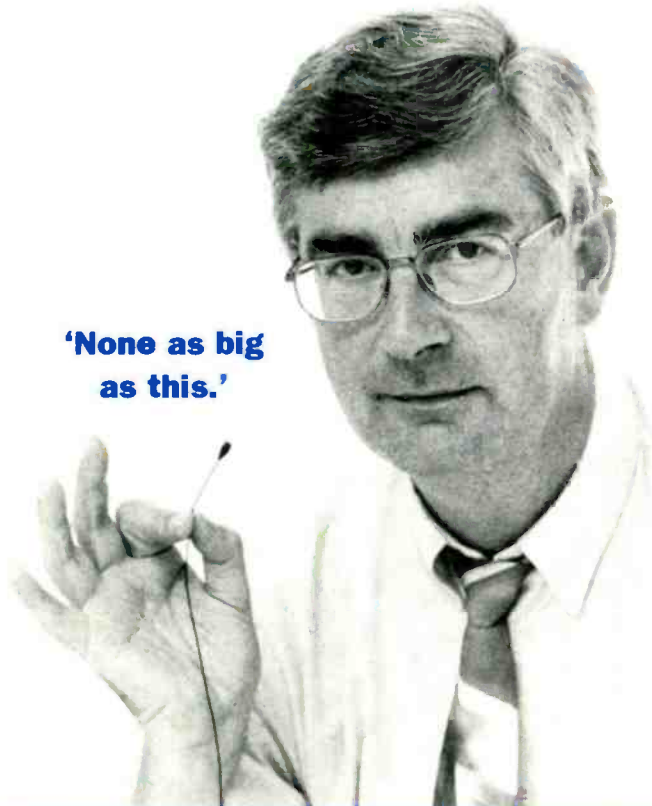
The National Vintage Communications Fair
Hall 11, NEC, Birmingham, UK.
Tel: +44 1392 411565.

Broadcast India 97 Technical Symposium
Chavan Centre, Mumbai, India.
Contact: Kavita Meer
Tel: +91 22 215 1396.
Fax: +91 22 215 1269.
E-mail: saicom@bom2.vsnl.net.in

Otari PD-80

The PD-80 represents the latest step in Otari's ongoing push into the crowded nonlinear recorder-editor market. **Rob James** investigates

THE OTARI PD-80 Digital Audio Workstation has arrived. And in a section of the market which seems to have more players than a casino. Otari's workstation package is a compact, hardware solution consisting of a 3U-high rackmounting unit, the PD-80, a neat little controller, the CB-164, and a standard display VGA monitor which was not supplied for review. Up to seven additional PD-80-A expansion modules can be added to the setup and all can be controlled from one CB-164, giving a total of 64 tracks. To come, there is a film dubber version, the PD-80-D, which comes with a front panel controller and brings a number of extra features to the party. While it can use the usual SCSI hard drives it is billed as a magneto-optical disc recorder and uses one of the new 'overwrite' 3½-inch Fujitsu drives which in this application is capable of replaying or recording eight data streams at once. There is front panel space for up to two of the M-O drives. Up to 16 consecutive punch in, punch outs are allowed. Disc capacity is 640Mb per single-sided disc of which 635Mb are available for audio. This gives a total of around 85 track-minutes depending on sampling rate. Recording is 16-bit, sampling rates are 32kHz, 44.056kHz, 44.1kHz, 47.952kHz or 48kHz. The GPIO connector provides a bi-phase input which allows the PD-80 to slave to film equipment at various pulse per frame rates. It also allows external equipment to control commonly used transport functions and provides tally outputs for indicator lamps.

The PD-80 front panel contains nothing more than a mains switch, headphone socket and volume pot, and one or two slots for the M-O disc or discs. The rear panel is busier, to say the least, with four channels of mappable AES-EBU I-O and Time Code I-O on XLRs, Word clock and Video sync I-O on BNCs and a SCSI connector. Analogue I-O is on two 25-pin D-connectors. The convertors are 20-bit 64x oversampling A–Ds and 20-bit 8x oversampling D–As. Other D-connectors take care of RS422 (Sony P2) machine control, GPIO and biphase, and the connections to the controller, VGA monitor and additional PD-80s. Optional boards are available for connection to machines using TDIF or ADAT interfacing. The remote controller can be up to 30m from the PD-80 or further if the CB-64 is provided with a source of 9V power. Connectors on the controller are a 25-pin to the mainframe, a standard IBM PS-2 mini-DIN to allow for a computer keyboard, the 9V power socket referred to earlier and a phones jack and pot. The controller is well laid out with eight bright bar-graph meters, a miniature QWERTY keyboard for labelling and a chunky jog-shuttle wheel with a good feel. Standard illuminated tape transport keys are in the logical place in

the bottom right-hand corner with locates above. At top right are four small buttons which handle paging through multiple PD-80s, VTR control, Chase and varispeed.

The rest of the keys are 'full sized' with a good use of colour to highlight certain functions. If you wish, the remote control will emit a discrete 'click' on each key press. You can also decide if you want the machine to beep on errors.

On powering up, you are presented with an opening screen with a number of pull-down menus and a box with a neat scrolling waveform graphic and time, date and software version information. The menus are accessed using the function and-or yellow cursor keys to select an item and the green ENTER key to confirm the choice. The 3½-inch M-O discs have to be formatted before recording can take place. This is done via the Disc menu. Housekeeping facilities leave little to be desired and cover physical and logical formatting of discs, verifying, copying and so on. The defaults held in backup RAM are changed under the General menu (which is also where you can set the time and date). Project and Library menus cover the creation, opening and management of projects and libraries respectively. The two other interesting modes of operation are selected under the Utility menu for some reason. The final menu is Power which enables you to put the machine into mac-like 'sleep' in a power-saving mode, useful if you want to leave the machine for a while. Touching any control 'wakes' it from where you left it.

In Otari-speak an individual collection of recorded sections of audio and the associated data is referred to as a Project, each recorded sound is called a Cue, a collection of Cues is a Library and a Cue placed in a Project is termed an Event. An Event consists of a set of instructions to the PD-80 which determine how a cue is to be played.

The display is clear and reasonably clutter free. At the bottom of the screen are four groups of boxes. The first group shows the name of the selected Cue, its start time, its sync point and end time. The second group shows Mark A and B times, the duration between them and incoming time code. The next group shows sample rate, frame rate and where the current reference clock is coming from. The last group has two status boxes for Chase mode and varispeed rate.

In normal use, the window occupying the middle of the screen is the Project window. At the left of this window are track number boxes. To the right of this are track name boxes which change colour accord[ing to sta]tus. When tracks [are in play they turn] green, when mute[d red, and when the] input yellow. Mo[...]

a row of buttons under the record selects which in turn are under the meters.

The rest of the window width is the scrolling track display. Events are coloured according to status. The normal colour of an event in replay is orange or brown. These colours alternate on successive events which makes identifying joins in tracks easy, while recording the events being created are shown in red. A selected event is yellow and an event selected to be moved is white. When half of a stereo pair event or one element of a multitrack event is selected, the other parts are displayed in yellowish grey. If editing is performed all members of the group will be affected. Events can be ungrouped for individual attention.

Above the Project window, a further window displays the Library list associated with the Project or Event list. The list windows can be expanded to fill the whole screen. Lists can be sorted on various criteria. Events can be edited numerically using the EMODIFY key. Libraries from other projects can be accessed and events added to the current project.

At the top of the screen are eight boxes which indicate what the white function keys on the controller will currently do. There are several pages of Function key assignments which are accessed using the FUNC UP and DOWN keys. The two keys adjacent to the jog wheel, labelled 2ND and 3RD, act like shift keys to modify the action of other controls—for example, when in Guide mode which allows the display to be scrolled without playing audio, holding down 2ND gears up the action of the jog wheel whilst 3RD gears it down.

The ZOOM IN and ZOOM OUT buttons allow the display resolution to be changed from less than 2s across the full window to over 2⁻ minutes in 11 stages, the minimum step size changes in proportion from 0.1 frames to 1s. Again, ZOOM IN and OUT are on dedicated buttons. Waveform display can be toggled by the associated WAVE button.

Comprehensive edit functions are provided together with a plethora of Audition modes. Most editing is carried out by marking two points with the large yellow A and B buttons. The space in between the marks becomes the edit region. Editing normally only affects the selected event or events but using the 2ND key you can, for example, slip all events on all tracks. If you decide you have got it wrong there are 16 levels of Undo-Redo available on dedicated buttons. A handy edit mode is the Gate function. This will chop up an event into separate events by cutting out silence; this is very useful where tracks with silence are transferred from another machine and you want to move things around. The Gate can be set to operate at thresholds of -60dB, -52dB, -45dB or -38dB which should cover the majority of situations. As before, if you overdo it, there is always Undo.

Punch-in and punch-out recording can be performed across any or all of the eight tracks together, either manually or automatically. Auto in-out times can be programmed with the AB markers or numerically. With the times set the auto punch in-out is initiated by pressing 3RD REC and PLAY while in Stop.

The locator functions are also impressive. You can locate the start and end of a project or individual Events or Sync points. There are

five locator memories available on dedicated buttons and a total of 100 memories. An unusual addition is the first five locations can be locked and may be used to trigger external events via GPIOs when in Project Programmed Play.

DSP functions are included for time compression and expansion, pitch shift and reverse. These time-domain effects are all off-line, although sections can be auditioned in real time before committing to the process. A choice of algorithms is provided for the pitch shift and time stretch.

Response on hitting the PLAY button is variable. From a standing start it is almost instantaneous, but after winding back a minute or two there is about a 1s delay, presumably whilst buffers are loaded. There is reverse sync play but this is initiated by pressing REWIND while holding down 3RD which is a bit counter-intuitive. Why, I wonder, did they not make it 2ND and PLAY? Varispeed can be invoked in play or record and covers a ±12.5% range.

Machine control is versatile with a good selection of frame rates and modes for time-code master and slave operation or RS422. When multiple PD-80s are synchronised at word clock accuracy you cannot use Chase Sync or Varispeed. The varispeed isn't much of an issue, but the inability to chase lock with multiple units can be limiting. There is a 'workaround', but this is somewhat inelegant.

In addition to the 'normal' mode of operation there are two other unusual possibilities,

Cue Quick Play and Project Program Play.

Cue Quick Play enables the numeric pad buttons on the right of the controller to be 'loaded' with Cues selected from any library for instant playout when the relevant button is pressed. Four pages of assignments can be saved and recalled for a total of 48 sounds. This has obvious applications in theatres and for quiz shows, and so on, or maybe even railway station announcements.

Project Program Play is, to my knowledge, unique in a general purpose DAW. It enables a completed Project or Projects to be replayed at predetermined times of day (triggered from the internal clock) or external time code. The playback can also be initiated manually. You can have up to 18 Projects per page and there are four pages for a total of 72 possible Projects. Each Project can contain up to five locate points each of which will send a GPIO trigger to operate external devices. This is, potentially, an extremely powerful tool for a wide variety of hitherto rather neglected applications—multi-track audio installations in theme parks and museums, for instance

Otari has an excellent pedigree in pro-audio. Their analogue multitracks are a semi-universal workhorse and the Radar digital multitrack has a good following. The company has aimed the PD-80 at the widest possible audience. It believes it will find friends in TV, radio, theatre and many other applications. There is however, one glaring omission for sound-for-picture users: there is currently no auto-conform function nor is there a diskette drive which would be required to load and save the EDLs. The company say this is on the wish list. The inability to chase sync with multiple machines in wordclock sync also makes the PD-80 awkward in this field.

This is, however, a well thought-out and versatile machine with a few limitations for certain applications. The learning curve is fairly shallow and the interface is pretty intuitive. In addition to the usual applications for a DAW, the PD-80 could be used for multitrack audio in theme parks or museum A-V installations, as an automated playout unit for radio commercials or any application which requires playout of various sequences at specific times. ∎

Soundscape v2

When the Soundscape workstation first appeared, its designers openly favoured functionality over flash. With its new version, Soundscape presents a serious challenge for those systems whose designers had other priorities as **Rob James** discovers

SOUNDSCAPE ENTERED the DAW market in 1993. Since then the company has established an enviable reputation for its products. This has been reflected in serious numbers of workstations delivered to facilities and individuals all over the world. Soundscape's version 2 software together with the required SSAC-1 accelerator card is a major upgrade to the system and extends its capabilities in a number of ways. There is also a new interface available, the SS8IO-1.

For those people unfamiliar with version 1, a little explanation may help to appreciate v2. Soundscape is built around the principle of using dedicated hardware for the audio recording, editing and manipulation with a standard IBM clone PC used as the display and control interface. Dedicated hardware controller fans are catered for via MIDI with software for the JL Cooper CS10-2 and MCS-2 and Penny & Giles MM16 and DC16.

If the PC is fitted with an RS422 card, Remote Device Control (RDC) software is provided to control external machines using the Sony P2 (9-pin) protocol, otherwise Soundscape can be an MTC (MIDI Time Code) master or slave. Multiple instances of the RDC can be run together to control several machines, limited by the number of RS422 ports.

Also included is an AVI player that will work in conjunction with video capture cards such as the Fast Electronics AV master or Miro DC30. Because Soundscape is so undemanding of PC hardware it is perfectly feasible and

practical to have a fast PC running picture (on a separate monitor) at the same time as Soundscape. This makes for a cost-effective Sound for Picture solution.

The basic building block is a 2U-high rack-mounting unit together with a PC interface card. Each interface card can connect to two rack units; a fully expanded system is currently eight rack units which will give a total of 96 tracks (simultaneous data streams). In v1 each unit gave eight tracks with two inputs and four outputs.

Each rack unit can be fitted with one or two fixed or removable hard-disk drives for audio storage. These are referred to as SDisks in Soundscape parlance.

The philosophy behind Soundscape is to place all the burden of dealing with the audio on proprietary hardware which allows the use of a comparatively modest PC for control. Practically, this means for v1 you need a PC capable of comfortably running Windows 3.xx and for v2, Windows 95. Soundscape uses standard (and cheap) IDE interface hard disk drives rather than SCSI drives. Until recently IDE drives had data bandwidth limitations which restricted their use in most DAWs. Soundscape's designers took the view that by using these cost-effective drives and designing their system in a way which does not demand high-performance disks, they would have a better option than the alternative approach of utilising expensive SCSI drives for maximum disk performance. The largest single drive currently supported is 4.294Gb. Larger drives, like the seriously quick and cheap Maxtor 5.1Gb, can be used, but Soundscape can currently only 'see' 4.294Gb. Under v1 software, if two drives are fitted to a unit they are seen as one drive. In v2 they can be addressed individually.

The primary user-interface is the main screen, this is customisable so you only need to display the windows you need for a given operation.

The EDL module is a separate application that uses the same hardware to conform material which can later be loaded into Soundscape.

Most of the terminology used by Soundscape is musical in origin thus the main display is referred to as the Arrange window. Raw audio recorded into the system is termed a Take. A complete Take or part of a Take can be placed on virtual tracks in the Arrange window. These are then referred to as Parts.

Tools are selected by using the mouse (or other pointing device) and then clicking on the Part you wish to affect. The mouse can be 'loaded' with four tools at one time. One each on the left and right buttons and one each on left and right with the ALT key held down. Tools are grouped in nine pages or Toolbars accord-

ing to the type of operation and you can customise the contents of each toolbar to taste. The parts to be played are coloured according to the physical track or output to which they are assigned. Only one Part can be played from each output at any one time and overlapping parts on a single track will be muted.

Here are just a few examples from the comprehensive list of tools on offer. Normalise creates a new take from the selected Part or Parts with the level adjusted so the peak is actually 100%.

DC Removal works in the much the same way but applies a 10Hz filter.

One welcome and extremely powerful tool is the Noise Gate. This can be used with a variety of parameters to break up a long part with silent gaps into a number of smaller parts. This can be very useful when, say, a complete commentary track has been recorded in one take and you want to see where the voice actually is.

Transport control is taken care of in the Tape Transport window which has the usual tape recorder-style buttons and displays time and so on, together with the current locator positions. As with most other functions there are keyboard short-cuts which duplicate the on-screen buttons.

The track display is static with a moving Play Pointer. When this gets to the right-hand side of the screen, the track display is redrawn. Time display can be SMPTE or Bars:Beats:Ticks. Resolution of the time display is down to 100th of a frame or 192 ticks to the beat. Editing resolution is down to individual samples. A waveform file can be generated for each part by using the Waveform tool. Once generated this file is stored so subsequent display of waveform is near instantaneous.

At the heart of the upgrade is the SSAC-1 card. This adds further DSP to the existing Motorola 56001 in the form of the new Motorola 56301. This chip offers significant advantages to the designers of Soundscape over the earlier device due in part to its re-locatable code structure. The speed and efficiency of instructions is greatly increased. This is evidenced by a huge extension of real-time processing capabilities and the way in which these can be assigned and re-assigned at will. The SSAC-1 also has a TDIF connector which can be used to connect to any suitably equipped recorder or mixer such as a Tascam DA-88 or a Yamaha 03D. Alternatively it can be used to connect to the other new piece of Soundscape hardware, the SS8IO-1 interface.

Instead of the output assignment arrangements in v1, there is a new digital mixer with a user-definable channel structure which exists in its own window. Different configurations can be created, saved and loaded and the mixer has total recall of all parameters including the settings of any real-time plug-ins. Each channel can have a number of real-time processes running concurrently. The limits on this are currently what will physically fit at the chosen screen resolution plus the total amount of DSP processing available. A scroll bar for the processing is under development which will mean the effective limit is the available processing. The mixer has 16 internal buses per SSAC-1 card. These can be used to distribute or group signals to be sent to internal effects processes or to a master fader. Any

Screens are highly automatable

Output or Send element can be routed to a bus and the input to a channel can also be a bus. A bus can be assigned as the input for multiple channels allowing flexible signal distribution. All mixer elements are, in effect, plug-ins. The EQ elements in the initial release are 2-band fully parametrics. If you want four or six bands, simply drop two or three 2-band EQs into the strip. Signals can be routed so you can record a mix of tracks back into Soundscape with effects included, if required.

Each element uses a certain percentage of the available DSP power. The total allocated percentage is shown at the top of the mixer page. For example a 2-band EQ uses 2%, a mono peak meter 0.3%. A processing intensive plug-in—a chorus-flanger from the new Audio Toolbox plug-in takes 10%. If you exceed the available processing power it is possible to simply mute mixer elements—they do not have to be removed. Thus it is possible to create a mixer which has everything you might need for a particular project but exceeds the available processing and simply activate the components you actually need to do the job.

As a result of these innovations the segment based EQ (which was the only way of achieving EQ in v1) has temporarily disappeared but will return as it is seen as a highly desirable feature. The v1 Merge function has been replaced by a full Mixdown function through the mixer. A simple Combine function using the mixer level and pan controls is under development.

The v2 software is fully 32-bit. Currently this means it will run only in the Windows 95 environment, but drivers are being developed for Windows NT. Since the hardware and software layers are separate in Windows NT this will open up the possibility of Soundscape becoming cross-platform in the future.

The biggest change to the record and playback capabilities is the number of simultaneous records is raised to 12 tracks per unit with buses from the two available in v1 and the number of playback tracks is raised to 12 per unit from eight. Via the TDIF port there are a further eight physical I-Os in addition to the original 2-in 4-out.

As mentioned earlier, each disk in a unit is now treated separately. The whole SDisk filing structure has been radically improved. It now resembles Windows Explorer but with additions specific to audio use. File names can now be 64 characters with a 128-character comment field on the SDisks. Takes are now stamped with the original time code and frame rate. Fields are provided in the Take format for default virtual track, physical track, volume, EQ, reverb, cue point, a 'read only' flag and an archive flag that will be used in v2.01. There is a proper folder-subfolder filing system which makes keeping track of multiple projects far easier. Folders can exist across several drives in the system allowing all files used in a project to be dealt with simultaneously. Arrange and mixer files can now be saved onto SDisk so a complete project with all the required information can reside on, say, one removable drive. This will make moving projects via 'sneakernet' a much easier proposition. Any PC files can now be copied to SDisk and backup can be to any logical PC device, tape streamer, magneto-

optical, Jaz or whatever. Software is also provided for backup to Audio DAT. A DAT backup can now span over more than one tape. In addition the hardware and software can be networked using standard PC networks with TCP/IP protocol.

Software is provided for recovery of inadvertently deleted files, however the usual caveats apply....

The limit on the number of virtual tracks is now increased to 256, formerly 64. Multiple units are now supported seamlessly in one window. In v1 it was necessary to have multiple iterations running. .WAV files can now be imported or exported in batches.

The other new item of hardware is the SS8IO-1. This unit provides eight balanced analogue I-O on XLRs using Crystal Semiconductors 20-bit convertors from a TDIF source. It also provides ADAT I-O (format conversion) and 8-channel metering. The metering is part of the ADAT interface circuitry. However, since the unit allows one input format to be assigned to more than one output format, you can have metering without an ADAT in sight. Sampling rates supported are 44.1kHz and 48kHz and there is a connection for external superclock or wordclock. The board also has a 512 channel expansion bus.

The impressive changes in v2.00 are only the beginning. The accelerator card is laying the foundations for major future enhancements. Already in development is an expansion chassis which will take up to multi DSP or I-O cards connected using the 512 channel expansion bus. Multiple Soundscape units will connect to the same chassis and be able to share the DSP resources.

There are already a number of plug-ins available for Soundscape.

The CD-R software combines Soundscape software with the acclaimed Point CD package for writing Red Book standard CDs. Reverb comes courtesy of Wave Mechanics and provides a comprehensive set of reverb effects. There is a Time module that provides time-domain effects and an Audio Toolbox with Chorus and Flanger effects in real time. More plug-ins are on the way.

Slated for v2.01 software are Punch in-out on the fly, play looping and more mixer elements. Various software enhancements are planned aimed specifically at sound-for-picture users. These include a count-in beep generator for ADR, M-S encode and decode, and various editing tools.

Also under development is a software mix automation module which will allow record and replay of dynamic mixer changes and an additional software module that will make all mixer parameters available via MIDI. Soundscape is also destined to support the new generation of advanced hardware controllers like the JL Cooper MCS3000 and Mackie HUI with full editing functions as well as mixing.

At the beginning of 1997 Soundscape announced a change in pricing which reflected the move to v2. The base system price for v2 is the same as the old v1 price. The accelerator card is, reasonably enough, an additional cost. For those who do not need the extra facilities there is now a v1.18F S (fixed software) entry-level system at reduced cost which can be upgraded later for approximately the difference in buying v2 at the outset.

Obviously any manufacturer hopes existing users will upgrade to the new version. Soundscape had estimated a 75% take-up for the upgrade. In fact, the company says, even before release this had been comfortably exceeded. Some measure, I think, of the level of user satisfaction with the system.

What all this adds up to is a self-assured DAW which can be configured for many specific tasks. If you take the time to set it up properly for a particular job it can be a satisfying and productive tool. It is easy to save complete configurations for specific tasks and-or individual users.

With v2, Soundscape is well on the way to achieving a very neat trick many others have tried to pull off—a robust system with a low entry level cost which is truly scalable at reasonable cost. It is now possible to envisage a very serious facility-wide solution to production process problems using a range of Soundscape units specified according to task. The modularity of the system makes it easy to change configurations as requirements change without rendering units redundant. The new networking capabilities are the icing on the cake. ■

HHB CDR800

The CD-R ball is now rolling, with HHB being the latest to add its experience to this gathering momentum. **Zenon Schoepe** test drives the CDR800 and wants to keep the keys

THE DEMOCRATISATION of CD-R has been a wondrous thing. Having attended one of the earliest launches of this type of technology in some out of the way Parisienne hotel suburb, I remember being impressed by the achievement but bemused as to why anybody would want to go to so much trouble and such expense just to roll their own CDs. In reply we were told that here, at last, was a medium that was convenient and universally acceptable that would become the interchange currency of the studio-to-record company chain and beyond and an invaluable aid for the postproduction conveyor belt.

I was unconvinced mostly because DAT was happening and looked altogether more exciting and practical, not to mention more financially accessible. When the reports came through on the questionable reliability of the early CD-R media I felt vindicated. However, I wasn't alone in thinking that if they ever tied the CD-R thing down, and sorted it out, and made it cheaper, which, of course, we were promised, then its potential was enormous. Well it has been sorted, and it's now cheaper. Considerably cheaper.

The HHB CDR800 is currently the cheapest professional CD-R recorder available. As such it could represent the final push that CD-R needs to tip it over into truly widescale acceptance.

There are a number of things that have to be clarified about this box. For starters, the observant will have noticed more than a passing resemblance to a certain Pioneer PDR-05 CD recorder—and they'd be right. What you're looking at here is one of those rather clever HHB rework jobs where the company has taken a fundamentally good piece of gear and tweaked it, and realigned it, to better serve the requirements of professional users.

HHB has done this sort of thing before with considerable success in DAT machines and now clearly thinks the time and technology is right for CD-R. It's not accidental that Pioneer has been active in lasers for some time and has been about as enthusiastic about Laserdisc as you can get.

I'm not going to dwell on the PDR-05 connection too much as the CDR800 is not simply a rebadge—there are a number of significant differences between the two that should be mentioned. For example, the original Pioneer CD-R runs on consumer CD-R blanks and not your standard CD blanks which it will not recognise. This is tied in with the PDR-05's consumer stance and the presence of SCMS which tries its best to prohibit bedroom pirating.

The CDR800 uses standard CD-R blanks and is not limited by SCMS, allowing copy protection to be encoded as Inhibit, No Inhibit and Once Only copy.

Internally the CDR-800 uses a twin power supply arrangement, a copper chassis

together with improved convertors over the Pioneer PDR-05 so HHB is clearly gunning for the attentions of the Golden Ear Brigade. More obvious differences include the 19-inch rack mounting and the presence of balanced XLR analogue inputs and an AES-EBU input.

A word or two about the transport: described in Pioneer parlance as a Stable Platter Mechanism, it is different to most CD-R machines because you put the blank in label-side down. This is actually quite awkward to do if you have established an elaborately sad and flamboyant routine for removing a CD from its case and placing it into a machine's tray. With the Pioneer mechanism you have to twist your wrist.

However, the benefit of this orientation is that the disc is better supported resulting in a claimed improvement in LF response while more practically it means that the laser is firing from above rather than from below. You don't have to be that clever to know that dust and dirt tends to fall to the floor rather than the ceiling which means that the optics are less likely to be afflicted by some of the gunging up that can occur over time. I'm for anything that improves reliability on a write-once medium.

The CDR800's rear panel is fairly well populated with unbalanced phono inputs and outputs, co-axial and optical SPDIF I-Os, an AES-EBU input and balanced XLR

analogue inputs switchable for -8dBu/+4dBbu operation. These together with the line inputs use the front panel RECORD LEVEL and BALANCE pots.

The front panel is pleasantly uncluttered by comparison and once you connect button to function, the CDR800 emerges as a simple machine to operate—so simple that I initially suspected the use of multifunction keys. But there really aren't any.

Aside from the regular transport controls (the forward and back track-search keys function as manual search when held down in Play or Pause), you are presented with nicely isolated RECORD and RECORD MUTE switches, the latter preset at 4 seconds. You also get the facility to program skip playbacks, but more significantly on a write-once medium, you can also record this skip data on to the disc providing, of course, that the destination CD player can recognise them.

The display is not that enormous and features lots of small bits of legending that you initially have to search around before you can work with it.

Among this machine's many attributes is the ability to translate DAT, MD or CD IDs automatically into disc track starts (you can also enter them manually) and the presence of an in-built sample-rate convertor that puts down 44.1kHz from anything from 32kHz to 48kHz. The latter is totally invisible to the **page 20 >**

user and requires no activation or selection—the CDR800 just does it.

How the machine actually records illustrates its versatility but on first contact is also the most confusing aspect of it until you understand what the choices mean and concentrate on the ones that you think you'll use the most.

Basically there are five recording modes —one analogue input mode and four digital input modes. Of the five modes two are 'traditional' manual modes—the aforementioned analogue mode and one digital mode allow the machine to be run manually from the transport controls—and the remaining three are so-called Digital Synchro modes and effectively variations on a theme. Synchro modes instigate CD-R recording from the arriving digital input and comprise a 1-Track mode in which the machine records a single track and then goes in to standby; an All-Track mode which records a string of tracks and then stops; and an Auto-ID mode in which the machine creates tracks in line with incoming index info from the digital source. The last of these represents the full-clone mode and is likely to be the most useful.

The Finalize process is performed on a push of a dedicated button, the machine prepares the disc and then you release the PAUSE button and away it goes and writes it in stone. You can also check various time related parameters from a TIME switch including the total remaining recording time available.

There's provision for a rather generous (approx) 7-second fade-in and fade-out which is instigated from the supplied infrared remote and this additionally activates Play Repeat and programmed playback functions. I did not take to the remote control because it's a tiny thing with the sort of squidgy buttons that characterise svelte modern hi-fi systems and apart from duplicating the main transport controls and recording functions it's main benefits lie in its direct numeric track access keys. It's a fab and tidy remote for listening back but I found it hard to accept that I would want to trust it while attempting tricky compilations. It's almost irrelevant because you can do the business quite comfortably from the front panel which is as clear and positive as can be. You can also peel out the main transport controls plus Track Write via a rear panel 8-pin parallel remote socket which should permit the construction of a remote that you could really thump.

I missed the ability to enter track sub-indexes from the front panel while recording which is something you can do with the Marantz CDR615. The omission of anything other than a phono output for this device, no matter how sweet and beautiful it sounds, could be construed as an oversight although you would expect to buy a CD-R machine to record on, not to playback from.

Even so, it is difficult not to be impressed by the sound quality of the CDR800, and while I will not be drawn on the nuances of the low-end response *vis à vis* the Stable Platter Mechanism, it's a fine and wholesome sounding piece of machinery that puts down on to CD what you send in. Should there be more?

The appeal of the CDR800 spins around the fact that it's a dedicated audio CD-R with no pretensions to doing anything else. It's pro. And for a CD-R machine it's also surprisingly stress-free to use —you don't have to babysit it, its modes are intelligent, and it'll even park itself.

On features the balance is about right, pricewise (£1,299 + VAT in the UK) it's outstanding. The CDR800 is a major advance for the cause of the CD-R. There's going to be a queue. ■

LA Audio Millennium series

LA Audio's Millennium-series currently includes a cost-effective dual mic-line preamp, dual compressor-gate, dual 3-band parametric and multieffects unit. **George Shilling** balances price and performance

DESPITE THE NAME, it would seem that there is nothing American, glamourous or West-Coast about LA Audio. The parent company is SCV London, and the units are made in the UK. The Millennium-series incorporates the MLX2 dual mic-line pream-plifier, GCX2 dual compressor-gate, EQX2 dual 3-band EQ, and MPX1 multi-effects processor. Unfortunately the latter shares its name with the latest Lexicon multieffects unit: I don't know why manufacturers can't think up proper names for their products. Why not a Tweaker or a Knobulator?

Perhaps because the Millennium-series consists of seriously boring professional pieces of equipment, with nothing humourous about them at all. That is why they are finished matt aluminium and grey paint. All the knobs are black and rubbery like sensible shoes, with good markers indicating their position. The buttons are small and black. Consequently, and despite low prices, the image generated by the Millennium-series is that of no-nonsense gear, designed and built to high specifications. And I really like the look: it is neat and tidy, and in no way distracting. The manuals are also grey and black: small format booklets which straightforwardly and comprehensively

explain every feature. Applications are suggested, concepts explained, block diagrams presented. There are even graphics of the front panel for noting settings—marvellous news for Total Recall fans.

Each unit is a standard 1U-high rackmount, a surprisingly shallow 15cm, compact and light. Power supplies are onboard affairs, which is good news. No power switches are provided, but that's okay: I hate walking into a studio where half of the outboard gear is powered down. A green LED glows on the front panel indicating the presence of power. Connection is via a standard IEC mains lead. Units are preset for the voltage of the country where the unit is marketed. Costs have been saved by omitting any external voltage switch, although internal adjustment is possible.

All units feature outputs at -10dBV on balanced TRS (tip-ring-sleeve) sockets—stereo jacks to you and me—and XLR sockets at +4dBu. There are no fiddly rear panel switches for level changing. Most comparable units feature similar inputs, although the MLX2 and MPX1 have mic inputs on XLR and line and DI on jacks. All units except the GCX2 include clip LEDs which light a sensible 3dB before maximum level.

The MLX2 is a very simple and straightfor-

ward unit with all the connectors on the back except DI inputs, which are on the front. There are buttons for input selection, phantom power, and phase which is useful and not featured on most budget consoles. An input gain knob has a good range and the output pot has 20dB cut or boost with centre detent. The filter is 2nd order (12dB 8ve) variable high-pass adjustable from 25Hz to 250Hz. This is useful in many circumstances, but particularly for reducing unwanted noise when recording vocals. The DI input is buffered by a high impedance amplifier. The signal then bypasses the input gain pot and goes straight into the line amp. This is useful for electric guitar for example, where if you connected directly to a microphone input much of the HF content would disappear due to loading down of the signal. The unit sounded clean and neutral, and is ideal for connection directly to any type of recorder.

Each channel of the GCX2 features a compressor and a gate, and both channels can be linked. The back panel includes a side-chain insert jack socket for each channel. This is again a TRS jack, but not balanced: usefully, the input signal is sent on the ring so you can externally process it before returning it on the tip. The compressors feature

MLX2 dual mic-line preamp

GCX2 dual compressor gate

EQX2 dual 3-band EQ

MPX1—multieffects

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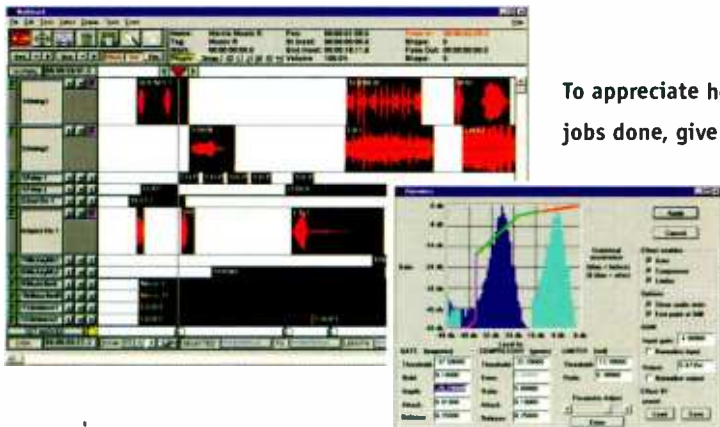
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A Euphonix Company

THRESHOLD, RATIO and GAIN knobs, there is a button for slow or fast auto attack and release, and one for bypass. A meter displays gain reduction. The gates' controls simply comprise threshold and release knobs and bypass switches. An LED lights to show gate open. A STEREO LINK button defeats all the Channel 2 controls and LEDs, and control voltages from both channels. inputs (or side chains) are combined. I would have preferred to have retained independent control of all bypass switches and especially the gain controls in this mode. The two auto modes for the compressors worked well in most situations, but I felt that the slow release was a little too slow. Also, I would have preferred separate switches for Attack and Release times, as implemented on the latest unit, the MPX1, since I often like to combine a slow attack with a fast release. However, at this price, this is a minor criticism. The gates work very well, with attack time quoted as less than 1ms, and range as -80dB. The attack is excellent, without incurring a horrible click on most programme material, and the release time range of 30ms to 3s is more than adequate. In Bypass mode, all sections. LEDs glow as normal, except more dimly, which is helpful.

The EQX2 is a dual 3-band EQ that can be linked to make one 6-band mono unit at the push of a button. Unfortunately, LA Audio has put the button on the back panel, which is no use at all when you have mounted the unit in a rack. A green LED glows on the front to indicate Mono mode, but that is of little consolation. Each band has a generous 15dB boost and cut, and a wide-range sweeping frequency control. The latter is only marked to indicate the frequency at each end of its range—a number in the middle would have been helpful. Between these knobs are tiny Q (bandwidth) knobs. These small black plastic-topped pots are barely bigger than the little push-buttons, and it is quite difficult to see where they are set. The settings are powerful, going from a narrow notch of twelfth-octave to three octaves. Each channel includes a clip LED, BYPASS button and 75Hz high-pass filter which remains operative even when the bypass is in, so there is a slight potential for cocking-up. No shelving capability is included, but the wide ranging Q control almost obviates

the need for this. Sound quality and clarity is excellent for a budget unit, and at this price who could resist having two units for stereo 6-band operation?

The MPX1 is the latest and most complex model in the range. It is a mono unit, specifically designed to cope with any processing you might need between vocal or instrument mic and recording unit, be it analogue or digital. First there is a mic-line-DI input section with all the features of the MLX2. This is followed by a compressor section which incorporates a simple single-knob expander to reduce extraneous noise. The compressor boasts separate FAST and SLOW switches for attack and release auto settings, plus a DS switch which inserts a boost at 8kHz into the side chain. The 4-band EQ section has fixed frequency shelving above 12kHz and below 80Hz, with two fully-parametric sweepable bands in between. There is a BYPASS switch, and also a 12kHz low-pass filter. The output section has a LEVEL TRIM knob, and meters for output level and gain reduction. the latter activated by either the compressor or the expander. The meter LEDs are dimmed when the compressor is bypassed. The back panel is quite a surprise, with no less than 11 jacks, as well as XLR input and output. The sockets allow you to patch separately into each of the four sections of the unit, allowing connection of the elements in any order. There is also a side-chain connection for the compressor, and a Link socket to connect two MPX1s for true stereo-linked compression. There is now a resurgence of do-all units such as this, many reminiscent of the old A&D Vocal Stressor, but this is certainly the best-value one I have come across.

A 5-year warranty is included with all models, which is unheard of in the studio gear market. Build quality is very high —despite their diminutive size they feel solid, and all the controls feel satisfying. Astonishingly, for what you get these boxes are very, very cheap. They are not top-of-the-range by studio standards, but are aimed mainly at the undoubtedly lucrative and recently expanding project studio market, where I am sure they will succeed. In most situations, these units will quietly, and boringly, do whatever you ask of them. I just hope they get noticed. ∎

Studio Sound July 1997

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Trantec

Three years after Trantec used a relatively narrow bandwidth with its S5000 radio mic system to offer preprogrammable synthesised tuning across 16 channels running simultaneously in the one television band, the company has taken the concept a step further with a dynamic means of simple local reconfiguration. Using computer controlled test routines Trantec can now digitally set up the new circuits in around five seconds. Three new systems have been launched—the VHF S3500 and S3000 (synthesised over a range of 165MHz to 225MHz) and the UHF S4000.

The fixed frequency S3000 uses PLL synthesised tuning and is available in handheld (Audio Technica capsule), lapel and headworn variants as well as a dedicated gui-

tar pack. LED indicators denote power, AF peak, RF level and diversity channel (A or B) while other features include variable gain output. The variable frequency S3500 gives 32-channel choice. Frequencies are contained within four organised banks allowing, for example, Bank 1 to contain deregulated channels 173.8 to 175.00, Bank 2 the fixed site channels, Bank 3 the independent programme makers, and Bank 4 a combination of all these. Features include a multifunction receiver LCD showing frequency, RF level and vu, and diversity operation with front mounted nondetachable telescopic antennae. Available in lapel and headworn mic versions and a guitar pack, two handhelds are offered fitted with dynamic or condenser Audio Technica capsules.

The S4000 is programmed with 16 intermodulation-free frequencies contained within TV channel 69 for operation in the UK but can be configured with up to 32 frequencies to increase its capacity for the worldwide market. Available in lapel, headworn, guitar pack and two hand-held versions (Audio Technica dynamic or condenser).
Trantec, UK. Tel: +44 181 330 3111.

Focus on Blue

The Focusrite Blue 300 mastering controller has been developed in response to the requirement for high-quality routeing and monitoring control in mastering.

It provides analogue and digital source selection, precision monitor control with speaker and headphones selection, recording path routeing with inserts and high-quality stereo peak metering. An oval-shaped remote connects to a 2U-high

Sennheiser radio microphones

Established at the forefront of radio mic manufacturers, Sennheiser cannot afford to stand still. **Dave Foister** brings us up to date

IF YOU'RE SERIOUS about radio microphones, there aren't too many people to consider buying your kit from. High on the shortlist is Sennheiser, who has done at least as much as anybody to bring radio mic technology to the situation it finds itself in today, and whose equipment is at least as likely as anybody's to be found doing the job where quality and reliability count.

The Sennheiser UHF range recently saw a revamp, with a new model of receiver complementing the extended range of microphones including the big hand-held SKM 5000 and its associated selection of capsules. This microphone is a familiar sight on TV and elsewhere, partly because it does the business but also because it looks good. The sleek body has a slot for the power supply pack, with two options offering dry cell or rechargable operation. The transmitting frequency is selected from 16 preset channels on a screwdriver rotary switch inside the power supply well, and one appealing aspect is that once assembled it is virtually tamper proof. The interlocking of the power supply and the body is like a puzzle out of a Christmas cracker, and if you don't know which bits to slide sideways you'll never get it apart. Similarly, the audio gain and high-pass filter switches are hidden underneath the screw-on capsule, so the only external control is the on-off switch next to the stubby aerial. The state of the battery pack is shown on a small row of LEDs, which remain lit even when the microphone is switched off.

Sennheiser is particularly proud of the range of capsules which screw on to the top of the SKM 5000. This currently comprises four condenser models, offering a selection of polar patterns. The head supplied with the standard kit is the ME 5005 supercardioid, and this can be swapped around with the 5004 cardioid, the 5009 wide cardioid and the 5002 omni. Of particular interest is the new vocal capsule, shortly to be available and currently a special order item, which first saw action at this year's Eurovision Song Contest.

Sennheiser is aware of the fact that purchasers of a hand-held radio systems expect to be able to achieve a typical hand-held mic sound, of the kind they would normally get out of a dynamic. Unprepared to compromise the high specification of the UHF system, and conscious of the acoustic effects of a body filled with transmitter as contrasted with the almost hollow body of a typical dynamic, Sennheiser set about tailoring the response of the supercardioid capsule to deliver the same characteristic. The result is a capsule that has all the quality advantages of a good condenser microphone but with the liveness and presence of a dedicated vocal model. Indeed it's a credit to the

system as a whole that such differences can be demonstrated as clearly as they are here. It will hardly need pointing out to those who have experience of a modern top-end UHF wireless system, but all the old difficulties and frustrations of radio working have been largely eliminated, and the audio performance of the whole chain is virtually indistinguishable from that of a good quality wired microphone.

The other relatively new component in Sennheiser's system is the EM 2004 UHF receiver, a single-channel diversity receiver intended for use with the whole range of transmitters. Its operation is simplicity itself, with a screwdriver rotary switch to select one of 16 frequencies to match the transmitter, a single horizontal LED bar meter switchable to show RF signal level or deviation (effective audio level within the received RF) and adjustable squelch. A headphone socket is provided, and operational information is given by a pair of LEDs showing which diversity channel is currently working.

Two short wire antennae are provided for attachment to the rear panel BNCs, and for

many purposes these are quite sufficient. A more elaborate system can be built up with a variety of optional remote aerials, and splitter sockets are fitted for linking two receivers on the same pair of aerials. Final audio output level is adjustable up to nominal line levels, and of course is partly dependent on the gain setting within the microphone itself.

This is a system in the best traditions of Sennheiser radio microphones. It's very simple to set up, virtually immune to fiddling by the performer using it, and the quality is quite superb. I ran the system with its omni capsule in the studio alongside some pretty grown up wired microphones, and the results were effectively indistinguishable—not only is the capsule performance the equal of a good studio condenser but the RF link does nothing to compromise it. Exploiting this with the new vocal capsule can only strengthen Sennheiser's already enviable position at the top of the wireless tree. ■

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Garwood ADA8

Garwood's system addresses the problems presented by multichannel radio mic setups **Terry Nelson** engages with an effective option

ONE OF THE PROBLEMS facing multichannel radio microphone system users is that of multiple aerials. It was okay in the days, when radio mics were the exception rather than the rule—one or two lead microphone channels meant four antennae at worst, assuming you were using diversity systems. But with today's requirements, you either put up with a forest of antennae or you do something about it.

That 'something' usually resolves itself into being an antenna distribution system, where all the aerial feeds to the various receivers are fed from a common antenna system, itself fed from aerials that tend to be more sophisticated than those generally found with receivers. This technique has two distinct advantages over the 'forest' solution: firstly, you can now package the receivers and distribution system neatly into racks without a rats' nest of cabling and

clutch of antennae and secondly, the quality of transmission is generally improved by the improved aerials and the fact that they can be optimally placed for reception. This latter fact benefits everybody as no one system is fighting for the 'best position'.

Following the success of the Radio Station in-ear monitoring system—which uses a channel-specific ground plane antenna (the RA-8)—Garwood Communications has expanded its range with the ADA8 antenna distribution system. The unit can be used with any 50Ω antenna but the RA-8 is recommended.

The ADA8 comes in the form of a 2U-high rack chassis in the traditional Garwood maroon livery, with the front panel blank of all controls except for two pilot lamps for Power and Phantom. The distribution amplifier is designed for both diversity and non-diversity 50Ω radio microphones, providing eight channels of distribution for the former and 16 for the latter.

The rear panel carries antenna input connectors for each channel (A and B) together with two Link connectors. These allow up to four ADA8s to be connected together for large systems or alternatively, in the case of non-diversity systems, the A Link connector would be connected to the Antenna B input for 16-channel use.

The RA-8 antenna looks like a model of the lunar module and features a central cone-shaped body with three short aerials as legs plus a vertical aerial. This ensures uniform coverage over a wide transmission area. Garwood recommends siting the aerial at a height of about 2m and away from metal structures.

The ADA8 has obviously been designed as a 'set-and-forget' unit—from the point of view of ease of installation in a rack, to putting the power switches where they cannot be knocked on or off inadvertently. The phantom power facility can be set internally to 5 - 15 - 20V and is factory set for 15V (as required by the MA-10). It should be noted that phantom should definitely be set 'off' when aerial amplifiers are not being used.

Installation of the ADA8 is extremely quick and simple, all cabling being on the rear. However, for systems with less than eight or 16 channels, BNC 50Ω terminator plugs must be used with the unused inputs.

The ADA8 is frequency specific: you must specify the frequency range that you work in when ordering. This has the distinct advantage of cleaning up the transmission no end as you are not picking up unwanted garbage along with your radio systems. The downside is that you are locked into your system and cannot use the unit with radio systems that use a different frequency range.

The frequency-specific side of the ADA8 (and shortage of time on my side) meant that I was unable to actually test the unit in operation (my systems being outside the frequency bandwidth of the review unit). However, I was able to get some user comments from Mick Shepherd of Hand-held Audio, one of the UK's largest suppliers of radio microphone systems and fully experienced with all the problems posed by 'going wireless': 'We now have eight systems or more and we have been very pleased with the results. The signal does get cleaned up a lot, though the frequency-specific nature of the ADA8 does mean that we cannot always use the support band's radio mics with it (when doing concerts). The fact that units can be cascaded for up to 32 diversity or 64 non-diversity systems makes it very flexible for setting up multichannel radio installations.

We discussed the frequency-specific side of the ADA8 in a bit more depth: 'Depending on where you are, one channel tends to get used fairly consistently (UHF channel 69) and this really is not a problem compared to the advantages gained. It also has to be said that few synthesised radio systems, where you can dial in different frequencies, can actually change channels. You still have to take systems that are appropriate for the countries where they are to be used.'

In short, the Garwood ADA8 antenna distribution amplifier offers a hassle-free solution for improved reception and easy installation for multichannel radio systems. Well worth looking into. ■

master module. The Blue 300 has eight stereo analogue inputs which can be selected on two independent stereo signal paths, with six digital inputs added using the Blue 260 D–A convertor. This 20-bit device uses static logic (no DSPs) and complements the Blue 245 A–D convertor which features six selectable digital inputs and a fully regenerated AES output.

Focusrite Blue series 315 and 330

The company has also upgraded its Blue 315 isomorphic EQ and Blue 330 mastering compressor-limiter. The former benefits from more EQ shapes including two bells and two shelves plus shelves can now be switched separately to bell mode for a maximum of four simultaneous bell curves. The low band has also been extended. The 330 sports a quieter and lower distortion class-A VCA circuit and a lookahead limiter achieved with an audio delay.
Focusrite, UK. Tel: +44 1494 462 246.
Net: www.focusrite.com

Wheatstone desks

The newly designed TV-1000 console for live TV includes a combined version of the company's Bus-Minus and eight Mix-Minus systems to feed IFBs and has claimed 'unlimited' input source to bus routeing capability with logic-follow for each source. It also provides comprehensive, intelligent monitoring capability with its Mute-Link feature, which enables foolproof studio speaker muting and an on-air tally display. Using the TV-1000's Mute Groups, operators can quickly move from live 'on-line' control to network feeds. Every input module and master output on the console has dedicated metering and the instant recall of switch settings and fader positions is possible via Wheatstone's Event computer, an optional feature on the TV-1000.

Other standard features on the desk include: individual channel compressor-limiters, user-defined aux send configuration (stereo or mono), stereo sub masters, comprehensive solid-state switching, alphanumeric displays for each monitor module's selected source, European logic

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Pictured right: SCM10 PRO, SCM20SL PRO, SCM50A PRO, SCM100A PRO, SCM150A PRO, SCM200A PRO, SCM300A PRO (left to right).

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Drawing on 20 years of experience in professional monitor loudspeaker design, ATC has developed the ultimate transportable active monitor system, the new SCM20A Pro. The radical cast aluminium cabinet houses ATC's legendary bass/midrange driver combining a 150mm bass cone with a 75mm midrange soft dome, and a 25mm tweeter along with independent 200 watts RMS and 50 watts RMS power stages with a phase corrected active crossover.

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Sony PCM-R700

Finding a balance of functionality, reliability and cost in a DAT recorder can be tricky. **Dave Foister** investigates the new Sony

IF A FORMAT as young as DAT can be said to have spawned any classics in its short life, there's no doubt that most of them carry a Sony badge. The company that brought the delights of Digital Audio Tape to the world has been assiduous in finding it a home in every area of what we do, from basic project studio decks to Walkman-style portables to grown-up location recorders to serious time-code-capable mastering machines. Few models have been anything other than a success, so a new one is inevitably of interest.

Here, then, is the PCM-R700, big sibling of the PCM-R500 with which it shares all but a few of its features. It's pitched squarely at the pro market, with integral rack ears and a back panel full of XLRs, although its sleek looks are more suggestive of a machine intended to blend into the living-room hi-fi. One or two of its gadgets, such as an automatic fade in and out function, also seem to belong more to the semi-pro application, but are not there at the expense of any professional features—the only omission that anyone could complain about is time code, but that is reflected in the price.

The new cosmetics make for a very friendly machine indeed. The display incorporates the

kind of join-up-the-dots alphanumeric window familiar from MD and DCC, which makes it highly flexible, and able to convey menu information without resorting to the indecipherable jumbles of upper and lower case half-characters which have you reaching for the manual every time you need to do something more complex than putting it into play. The ease with which this can show what's going on means that the whole display is less cluttered, as little dedicated light-up panels are less necessary.

The transport controls, too, are unusually elegant in operation, with concealed illuminated symbols in their surfaces which light up brightly in an apparently blank button to show what's happening. The result of all this is a deceptively simple front panel, whose logical layout and lack of clutter hide a wealth of functionality.

Partly responsible for this is the menu system for the less frequently required functions. This is called by a single MENU key; clearly-identified parameters are selected with the transport's shuttle ring and have their values altered with what looks like

a jog wheel before being set with, you guessed it, the SET button. Items covered by this include time and date formats, contents of the AES-EBU data stream (at last it's becoming acceptable to include IDs in this as well as SPDIF), and the threshold for the Auto ID function, which unusually is adjustable in 1dB steps rather than the more familiar 10dB lumps. It's even possible to set the length of blank space introduced by the Record Mute function and the length of silence required to trigger an auto ID.

This is typical of the thought and detail in the machine. A feature I haven't seen before is an automatic end search if Record is selected while the machine is sitting in a section of unrecorded tape; if only more recorders had this we'd see the end of the amateur-hour tapes coming in with no A-time and no program numbers on them (why do so many people find it so difficult to understand DAT's little foibles?).

The audio quality is superb, and can benefit from built-in Super Bit Mapping, although this is switchable. Full pro I-O spec is included, with screwdriver presets for matching analogue levels to the rest of the system; nominal level as supplied gives —20dBFS for +4dBm input with the front-panel controls at their centre detents. Digital signal handling includes Sony's broadcast studio version of IEC958 (SPDIF) allowing additional subcode transfer and SCMS override when used with other similarly equipped machines.

The main bonus on the PCM-R700 compared with the 500 is 4-head off-tape monitoring with appropriate switching; it also adds the aforementioned fade functions and a switch for locking out the front-panel keys. Both machines have facilities for wireless remotes and sockets for both parallel and serial wired remotes.

Subcode handling is comprehensive, complete with time and date stamping, and includes niceties like rehearsal of ID positions before writing and moving IDs after they've been recorded, all helped by the Shuttle wheel. The central dial is not a jog wheel at all but is purely for data entry, both within the menus and for specifying a particular track to find. One locate memory is also provided.

Sony has no doubt done it again; the PCM-R700 ought to be a winner. Perhaps every manufacturer's dream is to produce the DTC1000 for the 1990s—the current state of the DAT art in a stylish affordable package that everybody knows like an old friend. Maybe this is it. ∎

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NEW TECHNOLOGIES

functions and machine control ports with fader start, lighted tallies for all switches, and 'on console' power supply status indicators.

Wheatstone's SP-8EX television audio console features a source selector overbridge which expands the console's source capability to over 300 inputs. Like its predecessor, the SP-6, the SP-8EX incorporates many of the features of the company's higher-end desks such as a complete switch selectable internal logic structure, mix-minus capability, and group muting. The SP-8EX also has a dedicated relay-operated machine control port for overseas clients, as well as traditional US-type opto-isolated control ports.

The company also produces a range of custom furniture for radio stations, which the company describes as 'highly stylised' and 'based on a curved style of architecture'. The U-shaped control room counter array of the new line includes a sweeping curve toward the interview area, and the studio interview-host table is kidney-shaped. The furniture includes a substantial amount of storage space, as well as generous wire raceways, and hinged wire termination panels. Audioarts Engineering, a division of Wheatstone Corporation, has introduced the R-5 on-air radio console which the company claims is simple to install and does not require disassembly. Among its features, the R-5 has two pairs of Sifam vu meters for stereo programme and audition, and all switches are LED illuminated for easy identification and low maintenance. It includes Audioarts' Simple Phone mix-minus feature, which simplifies a station's multiple call-in operations; full function tape remote controls; on-off button nomenclature; and four microphone preamps. The new console also provides electronic switching throughout and two inputs per channel.
Wheatstone, US. Tel: +1 315 452-5000

Sennheiser dynamic

Introducing a new generation of Sennheiser stage mics the dynamic super cardioid MD425 has a capsule that is shock mounted on a rubber membrane and features an impact protection shield and an acoustically transparent stainless steel inlet basket.

Weighing 320g, the mic has a noiseless on-off reed which can be locked and a rugged quick release clamp.
Sennheiser, UK. Tel: +44 1494 551551.
Sennheiser, US. Tel: +1 203 434 9190.
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Elation KM-201

The Russian mic must now offer more than military styling and an indecipherable manual. **Dave Foister** approaches a new one

THE RUSSIAN MICROPHONE industry is becoming a major force in the market. I imagine many Western companies would give their R&D budgets to achieve the kind of impact and growth the Russian manufacturers have managed. The novelty may have worn off, but in the cold light of day the microphones still look good, and have genuinely expanded our choices.

A new offering like the Elation KM-201 has therefore, perhaps, lost the advantage of the early arrivals in that it is no longer a curiosity. Now we must approach it without surprise, and without being patronising, expecting it to perform to professional standards and stand up to established competition in its own right. Commendably, it emerges unscathed from the scrutiny; like many of the Russian models, this microphone

would be impressive whatever its origins.

The Elation is a modular system; one slim preamplifier body accommodates a variety of different capsules offering a range of polar patterns. The standard kit has the body plus three heads in a classy oak box, the superior finish of the exterior promising much from its contents.

The capsules fix to the body by means of screw threads, and these score over some other such systems in being far less delicate. Some microphones of this type seem more inclined to mate cross-threaded than properly aligned, resulting all too often in the uncomfortable nerve-jangling mangling of soft metal and the eventual stripping of the threads. The Elation heads line up positively and the broader threads lock down securely first time.

The capsules themselves are a little bigger than on some systems, with a reassuring chunkiness and particularly clear engraving of the associated polar pattern.

The three capsules supplied with the kit cover the expected trio of cardioid, hypercardioid and omni, and because of the apparent robustness of the components changing them around during a session is a far more practical proposition than it is with some systems. Swapping the heads is in fact the only adjustment possible on the

KM-201; it has no filters and no pad, claiming an impressively high SPL handling capability combined with low noise.

The only other item supplied in the kit is a basic plastic stand clip, complete with thread adaptor, which, although it looks cheap and cheerful, almost as though it was not actually designed for this microphone, grips the body well and supports the weight without drooping.

So far, the impression throughout (with the possible exception of the clip) is of quality and good design. Happily this also extends to the performance of the microphone, which exceeds the expectations. This is undoubtedly a good general purpose workhorse, but in many situations it can be more than that.

Used on individual instruments within a horn section, the kind of role a microphone like this is likely to be playing a lot of its time, gave more than the routine raw material, with an extra bite that I put down to extension of frequency response rather than lumps and bumps in it. Confronted with the kind of awkward oddities I get presented with on a regular basis, like bass clarinet, it continued to deliver the goods, with a completeness to both ends of the spectrum that left the easily-compromised sound intact. It surprised on vocals, with either the cardioid or the hypercardioid head in place; any expectations of a small sound in proportion to the size of the microphone itself were confounded by a big, full delivery coupled with a good resistance to popping. It's never going to replace any of the big classics as anyone's favourite vocal microphone, but it is certainly capable of doing the job.

Determined to test both dynamic extremes, I fitted the omni head and stuck it inside a kick drum for some overdubs, expecting to have to rescue it once I'd heard the resulting mess and replace it with something more suitable. Amazingly, it produced a particularly clear result, with its lack of proximity effect helping to avoid muddiness. The punch of the drum was tangible, and there was no indication that the microphone was struggling in any way.

I was provided with a pair of 201s, so was able to try various stereo setups as well as individual instrument miking. Spaced omnis worked particularly well, as drum overheads (still no headroom problems) or for orchestral section pickup. I also used them on a grand piano with impressive results.

Everyone now acknowledges that there are cheaper alternatives to many of the established microphones, models that deliver the required sonic performance at a fraction of the cost. Some have their trade-offs, most commonly a build quality that doesn't inspire confidence in their ability to withstand the rigours of the studio for years; the Elations seem to suffer from no such drawbacks and can be wholeheartedly recommended. ∎

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Barth unveils digital radio station

Barth has shown an EELA Audio digital mixing system called D.MAX that combines digital mixing with matrices. The system is aimed at a number of broadcast applications, including a journalist's workstation, sound transfer and the smaller self-op on-air studio. Up to 32 mono or 16 stereo signals, analogue or digital, can be routed digitally to either two or six control channels, providing one or two stereo outputs. There is a 7 x 24 stereo routeing system on the output as well as the 32 x 12 router on the input stage. A range of convertors and digital I-O modules are available, allowing the system to be tailored to individual requirements.

A modular approach also extends to the control surfaces, which are compact desk units containing faders and switches. More unusually, D.MAX has a serial interface which enables a high degree of automation via a modem. According to the company, the serial connection allows remote control via a 9600 baud modem and even with this low bit-rate, there is the capability to access all functions without time limitations. As a result, remote station automation can be extended to the point where it even includes the level meters to DIN specification over a standard telephone line.
Barth, Germany. Tel: +49 40 229 8883.

ARX

The DI-4 quad direct box-stereo line mixer allows unbalanced to balanced interfacing with gain or padding, ground lift, stereo mixing, plus aux sends for monitors and effects with a stereo aux return for signal rerouteing into left and right buses.

ARX Ambience 5 speaker

The Ambience 5 compact loudspeaker is said to be suitable for installations and is constructed from a lightweight high impact resin cabinet with a heavy duty protective steel mesh. Frequency response is claimed as 50Hz–20kHz ±5dB. The box uses an 8-inch low-frequency driver and 1-inch high-frequency dome radiator. Available in 8Ω and 100V versions with XLR or Speakon connectors.
ARX, Australia. Tel: +61 3 9555 7859.
Net: www.arx.com.au

Drawmer MX30

'Budget' dynamics often shave too many corners off expensive cousins for professional use. **Zenon Schoepe** finds an exception

THE RELEASE of the MX30 as an affordable Drawmer dynamics processor is not entirely without precedent. The company has produced a cheaper unit in the past in the form of the LX20 expander-compressor. However, this new box benefits from some of Drawmer's technological advances over the years and the sort of manufacturing techniques that permit well speced units to be put together for alarmingly small amounts of money. The MX30 distinguishes itself by being quite unlike any other Drawmer box before it and interestingly there is talk of a series of units all following along similar lines.

With the majority of affordable dynamics units, compromises manifest themselves in a number of tell-tale ways. Usually the box offers all the basics but is denied the little useful extras that can make a difference and may also feature glaring reductions in functionality. With the MX30, rather than strip away existing functionality Drawmer has approached the whole business of dynamics control afresh.

Billed as a dual-channel, stereo-linkable compressor, limiter and expander, these are indeed the functions that the MX30 performs but you are effectively presented with only five pots and one switch per channel with which to operate it. But before dismiss this box as too simplistic and too low-end for you, remember that it is a Drawmer. The fact that it's cheap should be regarded as bonus.

You also get some typical Drawmer LED metering of gain reduction and output level, the latter being switched to read a channel's input level when its Bypass is selected. Additionally there is a STEREO LINK switch which throws total control of both channels over to Channel 1 for stereo processing, the only exception to this is each channel's individual Bypass. Back panel connectors are supplied on unbalanced jack and balanced XLRs.

Cosmetically the MX30 is a deviation from traditional Drawmer all-black livery—presumably to differentiate itself in some way from its more expensive brothers. It looks fine, the quality is spot on, the LEDs are bright, the switches have a longish throw and feel positive, and the tidy pot caps have the usual clear position pointer built in to their seats. It's also very quiet.

This has got to be the most simple and effective box in this class that I have ever tried. Despite the apparent paucity of controls, it does compress, it does gate and it does limit extremely well.

I was sceptical about the one-pot gate and encountered some undulating programme that was giving the fast release setting a bit of hard time, but the SLOW RELEASE switch cured the problem.

However, it's the compressor and limiter sections that are likely to take most people's fancy as their versatility and results belie the relatively limited degree of control. You'd

Gating is programme adaptive, expander-like and lower ratio on close to threshold signals and it is controlled on a single THRESHOLD pot covering -70 to +20dB. There are LED indicators for above and below threshold signal levels and the section is defeated by turning the pot fully clockwise. Fast and slow release times can be selected on a switch, again with a LED.

The compressor section combines ratio and soft-knee characteristics that manifest themselves as a softish front-end becoming harder as more is piled in. To control this the THRESHOLD pot is variable from -40 to +20dB, while the RATIO pot kicks in at 1.2:1 and then trots over to an unspecified hard limiting value very much greater than 20:1.

Limiting is handled by a single OUTPUT LEVEL pot driving a circuit that resorts to additional gain reduction should the threshold value be exceeded for more than a few milliseconds. Limiter activity is signalled by a LED and is preceded by a GAIN MAKE-UP pot for the compressor section offering ±20dB of attenuation.

expect a brand box like this to do the job but you'd only hope that it also sounded good. The MX30 is as happy at clamping down dialogue gently as it is beefing up drums and pumping whole mixes. The extreme settings are delightful, abusing the limiter on its own produces a fabulous 'phwaatt' to the leading edge which is very reminiscent of more expensive units in the company's range. And it just will not overshoot.

Dual-channel operation for individual signals is flexible but I particularly appreciated the complete hand-over of control to one channel for stereo processing. It's quality performance.

This is an amazing box that represents amazing value for money. Admittedly its lack of control means that it makes decisions for you and it won't replace fully variable devices because it's not completely foolproof. But few dynamics devices are and that's often part of their charm. This is a smart repackaging of current clever technology that is very accessible. Highly recommended. ∎

NEW TECHNOLOGIES

28-bit A–D convertors

The Stage Tec Nexus digital routeing and interconnect system's new microphone inputs now feature 28-bit A–Ds on 4-channel cards claiming 22dBu maximum input level and 150dB dynamic range, with adjustable gain 0dB–70dB. In addition to the 24-bit output for the Nexus coupling field system, there is also an unprocessed 28-bit direct output, which the Cantus mixing console attached to the system can use.
Stage Tec, Germany.
Tel: +49 951 972 2525.

Audix dynamics

Latest in the OM series of dynamic vocal mics, the hypercardioid OM6 claims a low-frequency extension not typical of other dynamic mics. Together with a flat frequency response the company claims the OM6 can approximate the sound and performance of more expensive condensers. The OM6 joins the OM3, OM5 and OM7 dynamics all of which major on high SPL handling and good off-axis rejection.

The D4 dynamic is intended primarily for kick drum use and has a new capsule designed to capture high SPL instruments with extended frequencies below 100Hz. The mic is flat down to 63Hz, albeit with a slight bump at 80Hz, and a gentle roll-off below 40Hz. From 80Hz to 1kHz it is said to be extremely linear. It measures 4-inches long and 1-inch wide.
Audix, US. Tel: +1 714 588 8072.
Net: www.audixusa.com

Low cost AirWave

Pacific Research & Engineering's low cost AirWave on-air broadcast console has a mainframe that is fully enclosed welded-steel, but modular. Optical-isolated logic interfacing for remote and machine control is built into the modules for plug-and-play interconnection to other studio equipment.

Other standard features include a preamplifier module containing five high-performance microphone preamplifiers with phantom power, balanced patch points, a stereo program-1, program-2 and monaural output module, a dual remote-line selector module, a time of day clock and event timer, and an easy to use telephone module. AirWave also provides headphone system monitor facilities for control room, studio, co-host and guests; four illuminated Sifam meters; assignable A–B input control logic; built-in cue-speaker and operator's room headphone amplifier and built-in studio talkback microphone.

The company's QuikBilt modular studio cabinetry features high pressure

July 1997 Studio Sound

Joemeek VC2

It's only a matter of time before Joemeek outboard is as famous as Joe Meek himself. **George Shilling** reckons the time is near

I HAVE LONG BEEN a fan of Joe Meek's productions. I even went to a Joe Meek Appreciation Society event a few months back. But until recently I had not tried any of the Joemeek range of outboard equipment, designed by Ted Fletcher. From what you see in some magazines, you would think that the most important thing about the

Joemeek range is the colour of the front panel, but could I recreate *Telstar* in my spare bedroom with the VC2 valve channel? (Maybe.) Would it blend in visually with my other rack equipment? (Nope.)

Being a mono unit was an impressive start —some of Joe's best productions were mono. The VC2 comprises a microphone amplifier, a compressor, an enhancer, and an output stage with gain make-up. It is not only the paintwork which makes the appearance endearing. There is a lovely big old-fashioned looking VU meter with a particularly attractive metallic strip on its scale. The knobs are black, and look like they came free on the cover of *Electronics Today*. There are handles for pulling it out of the rack. And all the pushbuttons are bright red. On the back there are XLRs for mic and line input, an output and a -40dB output for connection to an external mic amp, although I have no idea why you would want to do this! There are jack sockets for an insert point, a mix input and TRS balanced output. There is also a mysterious shiny gold-plated unlabelled phono socket. No don't be daft, it's not a digital output, it is for stereo linking two units together. The manual waffles away about 1960's and 1970's technology, and boasts about such features as an unusually high overload margin. Reading it, you feel more and more like Ted has revived some long lost ancient British craft, and you can almost imagine him hand-building these in some chaotic workshop full of ancient test equipment and components. I was hoping to smell the valves when I turned it on.

No such luck, but I did like the yellow-white illumination of the VU meter. I had some vocals to record with a session singer: no time for faffing about! Fortunately I found the unit quick and easy to set up. Partnered with a standard Neumann U87 I was astonished at the clarity of the sound. After setting a rough level I pushed in the compressor switch. The singer pointed out that it sounded a bit compressed—he wasn't

wrong! There are two modes, a threshold knob, and knobs for attack and release times. I had switched on with mode 2 engaged. This is for people who crave the sound of compression, and want to hear it pump. I reverted to mode 1. This was smoothness beyond compare, and we got on with the job in hand. The compressor uses photo-electric technology, which many prefer over the more recent and more common VCA technology, and on many different types of program it does work very well. However, there is no graininess, even when at the fastest settings, so if you are after crunchy extreme compression effects for percussion you might have trouble achieving them with this kind of unit, wherever you put the wide-ranging attack and release knobs.

The Enhancer section is included to add a bit of sparkle. This is not a subtle thing either—extraordinarily powerful, with a wide range of adjustment. A drive knob is set to make the dim green LED glow brighter on signal peaks, but not red which indicates overload. Then you can ease up the enhance knob to taste, tweaking the Q control to affect the length of the high frequency harmonic after the sound that created it. The manual wisely advises caution when using the Enhancer, but used subtly and carefully it is quite appealing. All Enhancer effects are a matter of taste—I am not a huge fan of any of them—but I must admit I was led into temptation by this unit.

In terms of layout, the front panel is fairly sensible. No scale is indicated on the compress knob, because, as Ted states is the manual, there is no clearly defined threshold. For my liking there are a few too many LEDs which don't really help much, but just add a bit of confusion. Perhaps if the buttons used had a deeper travel it would be easier to tell at a glance which modes are in use.

The tube is in the make-up amplifier section, adding a characteristic warmth. I was impressed with how low the noise level was.

The VC2 is most enjoyable to use, and something slightly unusual. I gather it is selling well in the States, and it is not hard to see why: characterful retro looks combined with a wonderfully smooth sound. I don't think it would be historically accurate for recreating Telstar, but is all the same a useful tool for modern recordists. ∎

Contact

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TL Audio Crimson 3013

As a solid-state version of a valve EQ, this new parametric offers a rare opportunity: **Dave Foister** exclusively compares technologies

ONE LOOK at the TL Audio 3013 and you know exactly what to expect. TL's consistency of approach has led to a readily identifiable product range that, besides the mixers, comprises three distinct sets of outboard units. Identified by their front panel colours (not original but still effective) the three ranges offer known common characteristics despite the crossovers between them.

The flagships belong to the Classic valve series, with the Indigos being simpler, more affordable valve processors. The Crimsons are all solid state, most being functional clones of the corresponding Indigo model, and on their initial launch won many friends immediately for their combination of features and quality in affordable packages. The Crimson 3013 differs from the others in being derived not from the Indigo range but from the top-end EQ-2 valve equaliser, with which it is functionally identical. Perhaps surprisingly, TL quotes identical specifications for the two, apart from boring things like size and power consumption. This provides ammunition to both sides of the valve argument, those who say the difference is a musical thing that can't be measured and those who say the whole valve thing is a heap of hooey anyway.

Apart from its circuit topology and its front-panel colour, the 3013 is identical to the EQ-2, so anyone who read about that in *Studio Sound*

allowing fine adjustment of two closely-spaced frequencies where most EQs would only have one band available. I would have liked to see a couple of the bands offering shelf operation, as it is. Q is fully variable on all bands and each has a gain range of ±15dB. Each band is individually switchable in and out of circuit, as are the two filters, swept 12dB/octave high and low-pass circuits with a range much greater than usual—this isn't just surgery, it's amputation.

The most surprising feature on the EQ-2 is also carried over to the 3013: the STEREO button. Anyone who's listened carefully to the phase between stereo channels through two mono equalisers, or even looked at them on a vectorscope, will appreciate the dangers of even a small difference in settings, particularly of the centre frequency. Here the two channels can be ganged together and operated from the lower controls (for some reason Channel A is at the bottom) with much closer tracking than could be achieved by eye.

The preamps are clean and quiet, worth using in their own right—and since the EQ can be completely bypassed this is a possibility—with a detented output GAIN knob and a peak warning LED to complete the picture. It would however perhaps be going too far to suggest buying the 3013 on this basis alone. The difference in EQ feel between this and the
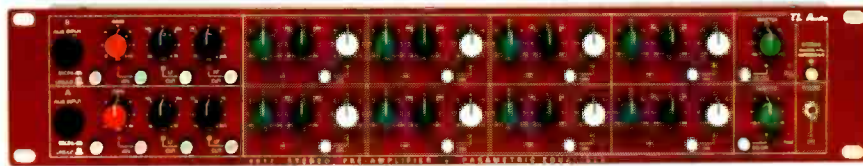
(May, 1996) can skip the next bit. Essentially it's a 2-channel, 4-band, parametric equaliser, with the additions of stereo linking and onboard microphone preamps. The phantom-powered microphone inputs give a total of four inputs per channel, with the expected balanced and unbalanced line inputs on the rear panel and unbalanced instrument input jacks on the front—a typical TL feature. These have switchable gain before the shared GAIN control, which is calibrated for a microphone input, but has a centre detent for line-level unity gain.

The EQ has four bands, presented as two identical pairs, with the LF-lower mid sections having the same frequency range and the HF-upper mid sections similarly doubling up. I reckon an ideal equaliser would have four identical bands with either switched ranges or multiturn pots to aid fine control over the whole spectrum. Conventional wisdom separates the range into overlapping bands, but perhaps TL's approach here is a good halfway house.

EQ-2 is undeniable, although that's not to suggest an intrinsic superiority of the one over the other. Where the valves have a character that seems to add a musical enhancement, the Crimson version has a neutrality that sometimes will be exactly what is needed. Adjusting the controls produces exactly the effect you would expect, with no surprises; this is not a box of mysterious magic tricks, but a tool for the occasions when you know precisely what you want and expect your equaliser to deliver just that. It does it all cleanly and smoothly, never imposing its own ideas and never detracting from the music.

It's clear that TL Audio is not putting all its valves into one basket, despite being one of its most enthusiastic advocates. The Crimson range may not be expensive, but neither are they cheap alternatives; on this showing the solid-state boxes are designed with as much concern for musical results as the valve versions. Well worth checking out. ∎

Focusrite Green 4

From Blue to Red to Green, Focusrite's outboard has proven consistently good. **Zenon Schoepe** turns green on compression

ROUNDING OFF the the latest rash of Focusrite's rackmount processors, this dual-channel compressor-limiter is a major piece in the great Green jigsaw. In fact, it tidies up the range quite nicely, particularly as the combined processing channel and voice-box strip units both draw on sections that are available in fuller blown stand-alone devices.

bypassed and the pair can be linked for stereo—Channel 1 takes control although the filtering and Look-ahead switching remain independent on each channel.

Connectors are on balanced XLRs and there's also a link jack socket provided for slaving multiple Green 4s together for multi-channel work.

The Green range is undoubtedly cheap by Focusrite standards but its units lie fairly close to the top in price of many other manufacturers' product portfolios. This is particularly the case when you stack the Green 4 compressor-limiter up against what else is available for this type of money because there's lots and much of it excellent. Consequently the Green 4 has much more competition than any other Green unit and is likely to have a slightly harder life justifying itself.

One of its strengths is that it concentrates its efforts on compression and limiting only with no distracting add-on gate or expander sections. What is on offer is two stereo link-able channels of proper compression with separate limiting control and high-pass and low-pass filter side-chaining. The layout is straightforward enough but there are a lot of extras on offer.

The compressor has fully variable threshold, ratio (1.5:1 to 10:1), gain make up, attack (0.3–100ms) and release (0.1–4s). Nipping in the bud any arguments about whether hard-knee or soft-knee compression is best, the Green 4 offers both, selected with a switch, while the RELEASE pot can be bypassed by activating a programme-dependent Automatic setting. The filters sweep from 15Hz–10kHz and 65Hz–25khz and can be switched as a pair into the side chain. They're sharp enough to flatten a boomy kick drum or de-ess a particularly frothy singer.

Limiting is handled by a fully-variable Threshold and Release (20ms–1s) plus a switchable Look-ahead function that delays the front end relative to the side chain to give the limiter that little bit of extra time for instances where going into the red really would be disastrous.

The metering is the rather good sort already seen on other Green units defaulting to input level but switchable to gain reduction which is displayed as a moving single dot on the LED ladder. Each channel's processing can be

Despite the fact that the Green range has now been available for just over a year, I already feel comfortable with the presentation principles involved. There are no surprises in the Green 4. It seems very familiar. The same solid and quiet performance I now expect of them all.

This is a powerful dynamics strip. The limiter works best as an output protection device rather than the sort of thing you can pile in to at the end of the chain. It's not that it sounds bad when under load, it's just that it can be beautifully unobtrusive when left to deal with the occasional hot blip.

The biggest boon for the compressor is the ability to switch between hard and soft-knee characteristics. Hybrid designs are fine but from my experience their precise nature varies from manufacturer to manufacturer and some are discernably better than others. A soft knee is handy when you want to compress but don't want to be too blatant about it, while the hard stuff shines when the aim is to feature compression as a process. Both have their place and the ability to try both in the same unit expands the potential of the Green 4. It can be as subtle or as crude as you wish but it's always smooth.

I would describe the compression quality as being pleasantly granular at high processing levels, it's definitely a contemporary sounding box. However, I would say that its real strength lies in its skill at controlling really tricky stuff. It's a sophisticated device with surprisingly fine resolution and an uncommon touch. It's cure-all dynamics that you can throw anything at. The Green 4 provides exceptional degrees of control. About the only omission is the lack of an automatic mode for the attack time. Not essential, admittedly, but there are times when it is nice to have some of the thinking done for you and the results can also be subtlety different.

Yet another well rounded Green unit which stacks up well against the opposition. ■

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transform the CPP into a post-production playback unit. An RS422 serial port links to other equipment remotely while a BNC socket is provided for synchronisation purposes.
Nagra, Switzerland. Tel: +41 21 7320101.

Mix-Genie v2

Currently being tested is a new chip controlling the Mix-Genie automation section of the 3G Fireball automated live desk. In addition to storing and recalling all aux mutes, channel mutes, channel and group EQ enables, aux master mutes and all fader positions, v2 software permits independent MIDI control over each switch and fader for real-time fader and switch changes when running on a sequencer such as Cakewalk Pro Audio V6 which the manufacturer recommends.
3G, UK. Tel: +44 1702 420645
Email: 3GAudio@compuserve.com

Prism DScope

Featuring high-quality analogue and digital measurement and signal generation, the DScope Series III also offers jitter and related digital interface measurements.

The interface measurements complement those available on the DSA-1 hand-held digital signal analyser. The system does not require cards to be fitted inside the host computer and runs under Windows 95 or NT. Measurement electronics are housed in a small external unit which occupies the same footprint as a notebook computer.
Prism Sound, UK. Tel: +44 1223 424988.
Net: www.prismsound.com

FAR out

Belgian monitor manufacturer FAR points to the benefits of having passive and active versions of monitors in its range for instances where one version better suits the circumstances. The CR10 and CR20 passives are available in active form (CR10A and CR20A) and both are shielded as standard. The company claims that the CR10 is surprisingly small for its low frequency extension and power handling while the CR20 employs a double voice-coil kevlar driver and achieves 92.5dB sensitivity.

FAR's largest monitor is the 3-way CR100 which has a 15-inch woofer and

Amek System 9098 Dual Compressor-limiter

The high-end System 9098 line continues to grow with the addition of this compressor-limiter. **Rob James** puts it through its paces

RUPERT NEVE is arguably still one of the most respected designers working in the pro-audio field today. Since selling his controlling interest in the Neve Group in 1975, he has worked with Focusrite and, more recently, exclusively with Amek. He received a Grammy award this year in recognition of his contributions to the industry. A new dynamics unit with the Rupert Neve seal of approval and embodying many of his design features has to be taken seriously.

The standard of construction of the 9098 unit is excellent. One huge PCB carries most of the circuitry with a separate, completely enclosed, power supply and satellite boards carrying the pots and switches.

The rear of the unit is one of the smartest I have seen. I-O connectors for both channels and side chains are on XLRs. There are two push buttons to set the operating level on each channel at either -4dB or -10dB. Adjacent to the IEC mains connector and voltage selector are a pair of binding posts with a strap across them. If the unit is used in an installa-

tion with a separate, technical, earth the link can be removed and the safety and technical earths connected separately.

The circuit design embodies several Rupert Neve 'trade marks': the main outputs are transformer coupled using his 'tertiary feedback' design. The side chain inputs are electronically balanced. Frequency response is flat (±0.2dB) 20Hz–20kHz with the -3dB points quoted as <10Hz and >120kHz.

The front panel is covered in the same material as the rear. The two channels each have identical controls apart from a 10dB pad switch for the meters which is grouped with the Channel B meter buttons and the STEREO LINK switch which is on the right under the illuminated mains switch. All the switches have associated LED indicators except the four black meter source select buttons. The meters can be switched to monitor Gain Reduction, Input level, output level or side chain. The compressor and limiter sections have separate bypass switches. The compressors have rotary controls for Threshold, Output Gain, Ratio, Attack and Release. Switches select Hard Knee

and Auto release time. The limiters are wonderfully sparse—one pot sets the limiting level, a second the release time. There is one pushbutton marked FAST ATTACK and a LED which indicates limiting is taking place. I know some people like variable attack time on limiters, but if the presets are well chosen, as in this case, I prefer the simplicity.

The STEREO LINK switch does what its name implies, all parameters apart from Output Gain are then controlled from Channel A and the Channel B controls and switches are rendered inoperative. The exception is Ambiance where both switches must be depressed if using Ambiance on stereo material. Beneath the large vu meters are the little source-select buttons and a switch that selects Side chain in. There is also a further switch labelled AMBIANCE for each channel.

The compressor and limiter have the character you would expect from a Rupert design. This can be summed up as 'musical'. Extreme settings can be used for specific and predictable effects. More gentle settings produce unobtrusive control, silk socks rather than support hose, if you know what I mean. They both smooth out bumps but which would you prefer?

The effect which really sets this unit apart lies behind the enigmatic AMBIANCE switches. When ambiance is selected the output of the channel is the difference between the input signal and the compressed and-or limited signal. If there is no compression or limiting taking place the output is, to all intents and purposes, silence. When gain reduction is happening all the dynamics controls exert an effect on the resultant signal but some are effectively reversed from their usual roles. In particular the OUTPUT GAIN control changes the balance between the compressed-limited signal and the input. With careful adjustment of the various controls, background noise and 'ambience' or reverb can be reduced almost invisibly. It can also be set up to reduce the background noise in gaps in wanted signal. This process would, of course, more normally be carried out using an expander or gate. The ambience effect is not, however, quite the same, either in the perceived change, or in the way the change is effected.

I played about with this for some time and managed to considerably reduce background hiss on an old voice recording without introducing unwanted artefacts. Similarly a reverberant lead vocal was rendered virtually 'dry'. If you have no other reason to audition the 9098 comp-limiter the Ambiance effect should provide one. ■

< page 40 7-inch kevlar midrange. Described as a compact 3-way, the DBW80 has a soft-dome midrange for improved stereo imaging while the DBW100 improves on its bottom end. Top of the tree is the DBW200 3-way system with two 15-inch low-frequency drivers and high sensitivity. FAR speakers can be bought in a variety of different finishes ranging from wood to high gloss paint.
FAR, Belgium.Tel: +32 4 259 7412.

Spirit Powerpad

With a width of only 22cm, the Spirit Folio Powerpad powered mixer has a stereo 30W amp that can be switched between mix and monitor outputs and ten inputs. It has four mic inputs, two stereo inputs, and each mono input has 2-band EQ, and a postfade aux send. Phantom power is switched globally, stereo inputs are equipped with switchable RIAA preamps and a 2-track return is provided together with peak and vu metering, a headphones output and clip protect.
Spirit, UK. Tel: +44 1707 665000.
Net: www.spirit-by-soundcraft.co.uk

EAW

Designed to optimise the performance of EAW loudspeakers the MX8600 close-coupled digital signal processor incorporates a number of exclusive EAW authorised parameters and factory presets. It supplies dozens of programmable memories, protected by a security lock-out function, while MIDI permits external control and linking of master and slave MX8600s via a personal computer.

Each unit has two inputs and four outputs with one output configurable as a direct subwoofer output. Each output has 3-band EQ, delay, digital level controls and variable high and low-pass filters that can be set for 12, 18 or 24dB/octave slopes.

The LA212 is the latest addition to the Linear Activation Series and combines a 12-inch cone driver in vented enclosure with 2-inch exit compression driver on a proprietary wave-guide. Long term SPLs are said to reach 120dB. An asymmetrical enclosure enables the LA212 to be used horizontally as a low-profile stage monitor or vertically stacked for PA applications. The new enclosure is aimed at applications that do not require the low frequency output of the larger LA215.
EAW, US. Tel: +1 508 234 6158. ■

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July 1997 **Studio Sound**

Dynamics processors

Managing the dynamics of an audio signal can be the difference between success and failure.
Studio Sound explains how they work, what's on offer, and profiles some perennial favourites

IF YOU WERE ABLE to ask Joe Meek his opinions on dynamics processing you might—just—learn the single secret of 1960s winning record production. In the 1990s, managing dynamics is a mature art still capable of making or breaking a production—here laid bare by *Studio Sound*'s confidantes. Whether dynamics processors should be subtle in operation or be the key to a producer's 'sound' is a debatable as any significant production issue. In classical circles, protecting the dynamic range of a digital recording system can be as important a consideration as the choice of mics, their placement, their preamps or the recording medium. Whichever way, the process and processors cannot reasonably be ignored.

George Shilling: One of my favourite compressors has to be the Fairchild 660-670. It's preamp magic on acoustic drums adding a vintage fruitiness that cannot be achieved with any other unit. A friend supplied me with the spec sheet of this unit which explains the mysterious attack and release modes, but generally I tend to 'suck it and see' when setting up. On vocals with a fairly slow setting (Mode 3 or even 4), the Fairchild brings a lovely warmth and smoothness to the sound. It does not, however, suit everybody's voice, and sometimes I revert to my favourite workhorse, the Urei 1176. This unit is just superb, and

works with any voice and microphone combination. The black-front-panelled 1176 is supposed to sound better than the aluminium version, but I have never A-B tested them, and can't claim to notice any difference. I'll never understand why Urei stopped making it—you can only get the stereo 1178 now. It's just about quick enough to use on bass guitar if there is no dbx 160x to hand, and is smooth enough for brass and pumping on drums. The dbx 160x is top dog for bass guitars—I never use the Over-easy mode, but in Normal mode it's brutal and punchy, and also useful for rocking guitar sounds

The Valley People Kepex II, is my favourite gate, its hysteresis curve can be adjusted, and you get no nasty splat as the gate opens when you process a bass drum (unlike certain other commonly found gates I could mention). There are no key filters, but somehow, you don't usually need them—they just work.

Zenon Schoepe: Like many others I have remained impressed from first contact with the simple control of Drawmer's DS201. It still defines what a gate should be to me—unobtrusive and just clever enough. It's about as close to a standard as things get. However, there is always room for something a little smarter and BSS' DPR-502 introduced a couple of twists including an attack transient enhancement feature and the ability to fire a

MIDI note by a convoluted process from each channel's envelope. It were great at the time.

BSS' DPR-901 dynamic equaliser falls somewhere between the realms of dynamics and EQ by applying multiband frequency-selective compression and expansion to give an effect that is rarely blatant but is often a life saver. Nothing else goes quite like it and it's excellent on vocals. For sound signature there is always the Joemeek compressor, but Calrec's RQD6400 packs two stereo compressor-limiters into 1U along with the best brick-wall limiter I have heard. You can slam into it and back out again without really noticing.

Fairman has done the Fairchild 660-670 tricks for the modern day with its TMC, but at a considerably lower price than the old stuff now commands. This is a superb piece of machinery that exudes feel-good factor and thumping, thick musicality. At the opposite end of the scale, Drawmer's first truly budget dynamics box the MX30 shows what can now be done for very little money. However, I'd be happy to be forced to use only TubeTech's beautiful LCA23 compressor and limiter. It's so fine sounding yet so versatile that I can't think of many instances where it wouldn't triumph.

Rob James: You won't find any valve devices among my favourites, simply because I've never worked much in the areas which benefit from their characteristics. Over

Drawmer DL241

Aphex Expressor

Aphex Compellor

dbx1066

< page 45 the years my most constant companion has been the AMS Neve 33609 and its predecessors. I was never that keen on the compression but the limiting can be set and forgotten unless you go really mad. The other analogue device I had a real crush on was the Aphex Dominator—it's not ideally suited to all circumstances but you can use ludicrous amounts of limiting without it being obvious.

More recently my affections have shifted to digital devices. The tc electronic Finalizer and M5000 mastering dynamics both blow me away; they can be used to make programme considerably louder than anything else I've come across without running foul of the PPM police. I also have a lot of fun with the built-in dynamics in Yamaha's O3D and the AMS Logic 2.

I am not a big user of conventional expanders or gates, but the Rupert Neve designed Amek 9098 has this magic 'ambience' mode which is not gating or expansion but does a similar job—much better—in my opinion. The only other expander I've come across so far I would consider using is the LA Audio 4x4² with its neat Selective Noise Reduction mode.

The last two devices are the Dolby Cat.43a and Cat.430 noise reducers. These use Dolby A and SR respectively and are relatively broadband devices which operate rather like expanders, but not quite. If the material you are working with is bad enough there is no real alternative, but unless you are damn careful, the artefacts can be worse than the noise.

Dave Foister: I can unashamedly say that my favourite compressor is my trusty old Rebis RA303 Complimex. As the name suggests, it's got the full set of compressor, limiter and expander, it's stereo, and it has just the right number of controls and indications on it to make it flexible without slowing me down. I always get what I want out of it and the quality is superb.

A more familiar model from the past, and one of my earliest encounters, is the Audio Design 760, a masterpiece of quality page 49 >

Drawmer 1960

TL audio Dual Valve preamp compressor

Joemeek compressor

and simplicity which might surprise those who are young enough to believe AD only does digital, unlike us old codgers who remember the Scamp rack. One of the longest-standing dynamics experts is dbx, and the recent 1066 is, perhaps, the best dbx to date. Again the balance between versatility and complexity is about right, and the required treatment only ever seems to be a couple of knob twiddles away.

Sharing dbx' advantage of in-house VCA design is Aphex, and although the name is known for exciters, there's a whole range of dynamics processors, often with a twist in the tail. I especially liked the Expressor, a very powerful compressor capable of drastic measures as well as subtle control and with the added bonus of an HF expander to re-inject a bit of sparkle into over-compressed material.

You can't go far wrong with Drawmer, a byword in dynamic control, and my favourite is the all-singing all-dancing M500, where the luxury of digital control and a preset library can save huge amounts of time, albeit at the risk of making you lazy. If you want to get down to the nuts and bolts it also has just about the most detailed control I've ever seen, as well as having probably the biggest selection of processes of any dynamics package around. Similarly reliable is Focusrite, whose Red 3 is about as smooth a compressor as you're likely to find. I'd also single out the BSS compressor, the DPR402, which impressed me with its unusually informative displays and simple mode of operation.

It could be said that a love for vintage valve compressors helped set the ball rolling for the tube revival: I remember being most amused in the vinyl days when engineers would proudly boast that their recordings were all modern and solid state, conveniently forgetting the ubiquitous Fairchild valve compressor that almost every cutting engineer used to put everything through. Naturally, compressors have featured strongly in the new valve equipment ranges, and TL Audio's C-1 is a particularly good example, offering a controllable valve sound in a flexible yet simple package. TL's espousal of the tube has done much to promote serious interest in its return, and the C-1 represents the company's pragmatic approach very well. Another early entrant to the new valve market was TubeTech, whose original Pultec EQ clone was followed by the LCA 2B compressor, a real throwback to the delights of tube dynamic control. ■

Lydcraft TubeTech LCA2B

DPR502 MIDI noise gate

BSS DPR902 dynamic EQ

Dynamics processors

John Watkinson discusses the basics of compression, limiting and gating, and their audible artefacts

Fig.1a: Compressor can work from input and output

COMPRESSION HAS a number of applications. The dynamic range of real sounds can be extremely wide and although high quality microphones can handle them, reproduction of full dynamic range requires large and powerful loudspeakers that are not always available or practicable. Where more modest equipment is in use, the use of full dynamic range may result in low-level sounds being lost in noise or high-level sounds overloading. Or both.

In pop recording, compression, even analogue tape overload, may be used as an effect. Tape recorders designed for unskilled users will usually incorporate compressors or automatic level controls so that the recording level control can be dispensed with. Many radio listeners, meanwhile, use cheap receivers that are really only suitable for speech. Often these are used in the work place where ambient noise intrudes. On the AM bands, background noise is extremely high. Consequently it is a practical necessity to compress music signals prior to AM

broadcast in order to optimise the transmitter modulation. One of the consequences of compression of this kind is that the programme material appears subjectively louder for a given peak power. In a competitive world some broadcasters use large amounts of such compression because it makes transistor radios of limited power seem louder. Generally once one station has taken this step others feel compelled to follow suit. As a result compression is also widely used in FM radio even though the use of FM allows a wider dynamic range.

Compressors work by analysing the signal level and using it to control the gain. The relationship between input and output dynamic range is called the compression ratio which is adjustable. **Fig.1a** shows that the compressor can work from the input or the output signal. The instantaneous level signal cannot be used as the rapid gain changes produce sidebands. The gain has to change slowly to prevent this distortion.

When the input level is increasing and is in danger of overloading, it is important to reduce the gain promptly. This is known as the attack of the compressor and the speed of attack must be a compromise between overload and distortion. Some compressors can change the attack automatically as a function of the slew rate of the input. However when the level falls after a peak the gain should be restored slowly. This is known as the release or recovery of the compressor. Separate controls are usually provided.

A threshold control only allows compression above a certain level. When manipulated in conjunction with the compression ratio control a variety of transfer functions can be obtained. **Fig.1** also shows some examples. At **Fig.1b** the threshold is set low so that uniform compression of most of the dynamic range is obtained. At **Fig.1c** the threshold is raised so that the compression characteristic has a knee. At **Fig.1d** the knee is set high so that most of the dynamic range

50

Fig.1b:

'KNEE'
Fig.1c

Fig.1d:

OUTPUT LEVEL
INPUT LEVEL
NOISE IS 'GATED' OUT
Fig.2a:

Fig.2b: Ducking

MIC SIGNAL 'DUCKS' BOTH OTHER SIGNALS
Fig.2c: Cascading gates

is unchanged but peaks are prevented from overloading. In this configuration the compressor has become a limiter.

Limiting is often used prior to recorders to prevent occasional transients entering saturation in analogue machines or clipping at the end of the quantising range in digital machines. The intended dynamic range is entirely uncompressed and provided the limiting only acts on brief transients it will be largely inaudible. It is also possible to place an equaliser prior to the compressor control so that the compression is frequency dependent.

The gain variation of compression is frequently audible. In particular, the human voice and classical music sound unnatural if compressed. The timbre of speech and many instruments, particularly piano, changes with level, and compression changes that relationship. The dynamics of compression are also audible. Because the signal can only have one gain at any one instant, instruments can amplitude modulate each other. A blow on a tympanum or an organ pedal note has its impact reduced by compression which also depresses the level of the rest of the orchestra. This then slowly recovers as the compressor releases. During release background noises and ambience appear to grow in level.

In multitrack pop-music production the
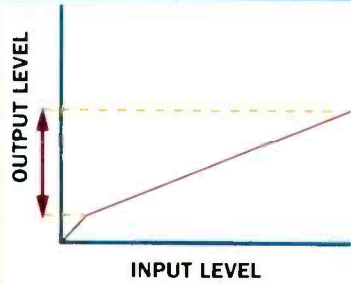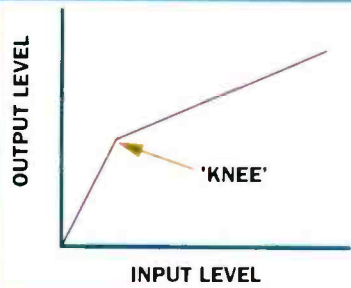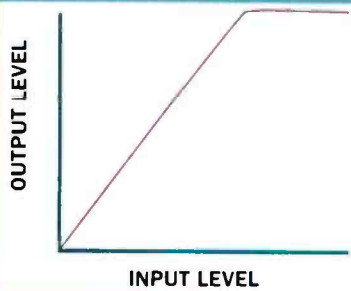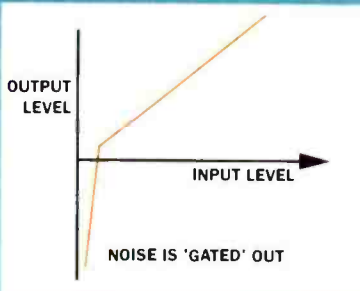
compressor can be used on individual tracks. In this case the compression of one instrument does not affect the level of other sounds in the mix and audibility is reduced.

The better the quality of the monitoring system the more audible compression becomes. When listening on a transistor radio the inherent distortion masks the changes of timbre and the background noise will be below the noise floor leaving only the amplitude modulation.

When using compression on stereo or surround sound signals, it is important to prevent the gain control process affecting the spatial characteristics. In practice the level of all of the channels must be analysed, but only a single gain control signal is produced which affects each channel equally. Compressor-limiters are often supplied in a 2-channel configuration. The channels can work independently for multitrack work or can be ganged for stereo applications so that the gain changes do not affect the balance.

Gating is another application of automatic level control. If a compressor is designed with an inverse characteristic shown in **Fig.2a** it will heavily attenuate low-level signals. The result is known as a noise gate because the noise floor of the input signal will be pushed down. Gating can also be used as in **Fig.2b** where one signal controls the gain of another. This may be used in radio stations or public buildings so that the announcer's voice automatically reduces or 'ducks' the level of any background music. Gating may be cascaded as in **Fig.2c** so that emergency messages will take priority over normal announcements and the background music returns to its normal level when neither type of message occurs. □

The Dynamics Outboard Listing

Alesis
3630 dual comp, limiter gate
Amek
System 9098 dual comp limiter
Aphex
106 Easyrider 4-channel comp
108 Easyrider twin-channel comp
320 Compellor dual comp-leveller
622 Logic dual gate-expander
661 Expressor single-channel comp with 'Tubessence'
720 Dominator II stereo limiter
ART
Pro VLA dual valve comp
ATI
Pro⁶ channel strip
Audio Technologies
Tubelink dual valve comp
Avalon
AD2044 dual opto comp
Behringer
MDX1200 dual comp, limiter, expander, enhancer
MDX2100 dual comp, limiter, expander
MDX2400 4-channel comp-limiter
MDX4000 dual 2-band comp, leveller, limiter
MDX8000 dual 4-band comp, leveller, limiter
XR1400 4-channel expander-gate
XR2000 dual expander, gate, ducker
XR2400 4-channel expander-gate
Bellari
RP583 dual valve comp-limiter
BSS
DPR-402 dual comp, de-esser, limiter
DPR-404 4-channel comp, de-esser
DPR-422 Opal dual comp, de-esser
DPR-504 4-channel noise gate
DPR-522 Opal dual noise gate
DPR-901 II single/dual 4-band frequency-conscious dynamic equaliser
Cairec
RQP3200 single-channel preamp, comp, expander, gate
RQD6400 dual comp-limiter
Cloud
CX-335 stereo comp-limiter
Crane Song
STC-8 dual class-A comp-limiter
Demeter
VTCL2 valve dual comp-limiter
dbx
Blue 160S dual comp-limiter
160A single-channel comp-limiter

163A single-channel comp-limiter
165A single-channel comp-limiter with peak stop
166A dual comp, limiter, gate
168A dual comp, de-esser, spectral enhancer
172 dual expander, gate
262 dual comp-limiter
263A de-esser
266A dual comp-gate
1066 dual comp, limiter, gate
900-series modular processing system
Digitech
VCS-1 dual comp, de-esser, noise gate, limiter
DNA
Dictator stereo peak limiter (studio and broadcast versions)
Dymand dual comp, expander
Drawmer
1960 dual valve comp
DL241 dual comp
DL441 4-channel comp
DS201 dual noise gate
DS301 dual expander, noise gate
DS504 4-channel noise gate
LX20 dual comp-expander
MX30 dual comp-limiter
Fairman
Tube Master Comp dual valve comp
Tube Recording Channel channel strip
Focusrite
Blue 230 dual broadcast comp-limiter
Blue 330 dual mastering comp-limiter
Red 3 dual comp-limiter
Red 7 single-channel mic preamp, comp
Green 3 voice channel
Green 4 dual comp
Green 5 channel strip
GML
8900 single-channel comp-limiter
Groove Tubes
CL1 valve comp-limiter
Höf
Dynamic Master stereo leveller, limiter, comp, expander
Joemeek
SC2 stereo photo-electric comp
VC1 single-channel mic preamp, comp, enhancer
VC2 single-channel mic preamp, comp, valve amplifier
VC3 single-channel mic preamp, comp, enhancer
Jünger
d01 dual digital comp-limiter
d02 dual digital comp-limiter with 20-bit A–D convertor
d03 dual digital comp-limiter with sample-rate convertor
Klark-Teknik
DN500 dual comp, limiter, expander
DN504 4-channel comp-limiter

DN510 dual noise gate
DN514 4-channel noise gate
LA Audio
GCX2 dual comp-gate
Classic Channel dual mic preamp, EQ, comp
Classic Compressor II
4x4² dual comp-gate
Manley
LIM 10/2 stereo comp-limiter
AMS Neve
33609 dual comp-limiter
Penny & Giles
PP10 Audio Multiprocessor multichannel software-based digital processor
Prism Sound
Maselec MLA-2 stereo mastering comp
Presonus
Blue Max single-channel comp-limiter
ACP-2 dual comp, limiter, gate
Rane
DC24 dual comp, limiter, expander, gate
MC22 dual comp
VP12 voice processor
Smart Research
C2 dual comp
SSL
FX384 stereo comp
SPL
Loudness Maximise
De-esser dual de-esser
Summit
DCL-200 dual comp-limiter
MPC-100 single-channel mic preamp, comp
TLA-100A tube levelling amplifier
Symetrix
425 stereo comp-limiter
488 8-channel comp
501 single-channel comp-limiter
528E single-channel comp. de-esser
562E dual expander-gate
601 single-channel digital voice processor
602 stereo digital voice processor
628 single-channel digital voice processor
SX208 stereo comp-limiter
tc electronic
M5000 multiple-channel software-based digital processor
Finalizer dual digital mastering dynamics processor
DB Max digital broadcast processor
TL Audio
C-1 dual valve comp
Indigo 2021 dual valve comp
Indigo VP-2051 valve voice channel
Crimson C-3021 dual comp
Crimson 3051 voice channel
Tube-Tech
CL1B single-channel comp
LCA2B dual comp-limiter

Smart Research C2

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Winning
formula

People build up a reputation, knowing that eventually somebody will come along to take it from them. This extends to the TV coverage, as different production companies and different broadcasters battle to provide technical facilities or transmit the entire season

As the 'most international' of all modern sports, the television broadcast of Formula 1 racing requires a high degree of cooperation between the host nations' broadcasters and the high-tech F1 circus. **Kevin Hilton** reports from the Canadian Grand Prix

FORMULA 1 Grand Prix motor racing elicits strong emotions: there is no middle ground. Its devotees regard it as a supreme challenge between skilled drivers steering vehicles created from the finest aerodynamic design powered by the best engines. For the dissenters it is merely a monstrous, noisy, tedious, full-size, real-life version of Scalextric. Either way it is a immensely popular sport: for those who can't make it to each venue round the world, television coverage is essential and TV stations bid enormous amounts to transmit the whole season, giving over large amounts of prime time to accommodate it.

F1 has a worldwide TV audience of 400 million viewers, a figure only surpassed by more occasional events like the World Cup and the Olympic Games. Controlling the TV rights to the sport is Formula 1 Promotions and Administration (part of FOCA), a body headed up by Bernie Ecclestone who recently doubled his salary to £54 million a year. FOCA strictly controls coverage of the GP season and is currently working on a 'shadow' project that will eventually result in multichannel digital TV coverage. My attempts to obtain more details met with unreturned phone calls, while those technicians involved turned out to be bound by non-disclosure agreements.

The popularity of F1 and the reasons why it is such a lucrative business are obvious. Each season is eagerly awaited by the fans; each Grand Prix throws up its own issues as the drivers and constructor teams vie for their respective championships. This year is no exception. The reigning champion, Britain's Damon Hill, is now driving for the regarded but still unproven Arrows Yamaha team, following his acrimonious split with current world champion team Williams Renault. His rival, twice champion Michael Schumacher (Germany), is still struggling with the unreliability of his Ferrari V10 engine. With these top drivers hampered in different ways, the best of the rest have a chance, opening up the races, and race for the championship.

This year has also seen the arrival of teams bearing the names of two illustrious past drivers: Prost Mugen Honda (after the French champion of 1985, 1986, 1989 and 1993) and Stewart Ford, fronted by the Scotsman who, up until 1987, held the record for the highest number of GP wins. It is not insignificant that this accolade was taken from him by Alain Prost. But F1 is like that: people build up a reputation, knowing that eventually somebody will come along to take it from them. This extends to the TV coverage, as different production companies and different broadcasters battle to provide technical facilities or transmit the entire season.

UK viewers are still getting used to the new style of ITV, which has taken F1 away from the BBC this season. While Fleetwood Mac are still reeling from the loss of royalties (their song 'The Chain' was used as the theme music for the BBC's coverage), the voice of F1, commentator Murray '...and here comes Schumacher—no, it's Hill' Walker (the man who once said that he did not get things wrong, he just made predictions that were instantly proved incorrect) was tempted to join the other side, providing some continuity.

While each production company will tailor its coverage in terms of presentation links, expert analysis and pit stop coverage, always a source of fascination for the faithful, the core of the programmes, audio and vision of the race itself, is supplied by the host broadcaster of each country. Due to the nature of Grand Prix, this means multiple cameras and a large number of microphones, mixed with footage from onboard the cars to give a driver's-eye view of everything. F1 is about speed, noise and excitement: getting this over to the viewer is the most important job for a broadcaster.

Round 7, 15th June 1997, Montreal, Canada. French Canadian driver Jacques Villeneuve, leading hope for Williams, arrives on his home turf, hoping to win the Grand Prix staged at the track named after his father. While the race will be broadcast live on both CBC (Canadian Broadcasting Corporation) and the French language service SRC Sports, as well as around the world, the presentation is being produced by the Canadian Formula 1 Association itself.

Hired to act as Director-Executive Producer for this exacting TV production was Michel Quidoz, a veteran of 13 Grand Prix who recently retired from SRC Sports after 38 years with the channel. 'This race has been broadcast on both the English language network, CBC, and the French language channel, SRC Sports, for the past 14 years,' he explains. 'This is my first year as a freelance and I am responsible for all 28 live feeds. We provide a clean host feed, with provision for commentary voice-overs to be added and commercials to be inserted.'

Like any major sporting event, there is a build up to each Grand Prix, not only with the usual punditry on who may win, but also coverage of the qualifying laps prior to the race itself. To make sure that no car is missed in Canada, 29 cameras were positioned around the track, backed up by helicopter shots and 14 in-car cameras, bringing the total to 44. The in-car vision and audio is provided by FOCA and transmitted from the cockpits over RF links.

'The 14 in-car cameras are allocated to the major teams and come to us from FOCA as a three-video feed,' explains Quidoz. 'I use these to complement my own shots, trying to match where they are with the ground cameras and the helicopter, which is under my direction.'

Because of the enormity of the job, elements are broken away from the main production and given their own supervisors. While Quidoz concentrated on the main business of directing the race, another director handled **page 56 >**
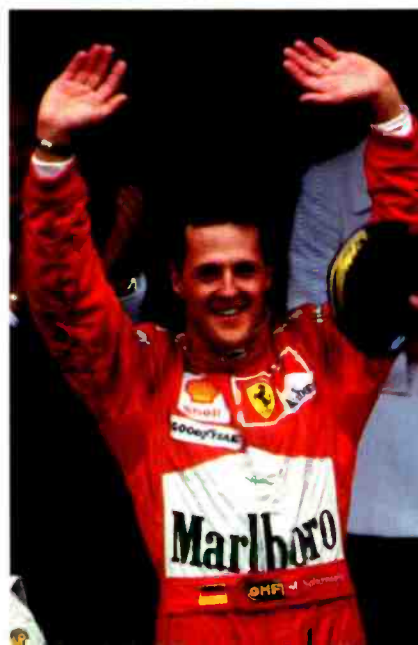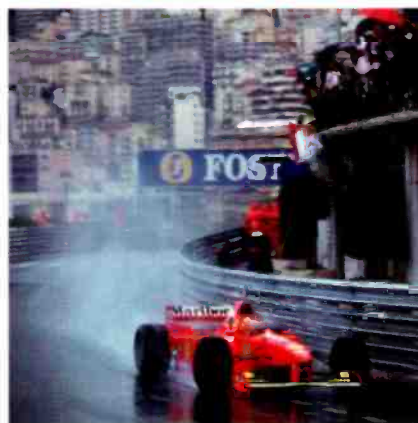
Plan of the Gilles Villeneuve circuit showing camera and microphone positions. The table shows camera and mic allocations.

< page 55 replays and in-car shots while a third worked specifically on the helicopter shots. RF feeds from the pits and cameras in the garages of two of the main contenders. Ferrari and Williams. To ensure that there was no confusion on the talkback. it was arranged that Quidoz' two colleagues could over-ride when they spoke. There are two switchers, one with me on and another with the other two directors. When the replay/in-car director spoke, he would over-ride me. as would the other director to him. One over-rides the other.'

Any TV production needs to know exactly what is happening and what will happen next but it is crucial in motor racing. The fans get decidedly aggrieved if they miss any of the action, something that has been underlined in the UK by the commercial ITV network taking

over coverage from the advertisement-free BBC. After the Melbourne Grand Prix. the first of the season. irate viewers complained about the breaks from the action so that people might be persuaded to buy something. By contrast Canadian viewers are used to this. as both CBC and SRC Sports are allowed by the government to obtain revenue through advertising —and that includes during Grand Prix races. which can include up to six minutes of commercials.

Aside from the insertion of advertisements. all effort is made to ensure that not an overtaking. a spin off or any wheel bumping is missed. All camera operators are accompanied by spotters to alert them to what is coming up. Race consultant James Robinson keeps in touch with the TV directors over RF

Microphone positioning			
Camera 1	441	Camera 11B	CK-8
Camera 1B	*568 + ATM25	Camera 12	MKH60
Camera 2	441	Camera 13	CK-8
Camera 3	568	Camera 14	MKH60
Camera 4	MKH60	Camera 14B	*568 + ATM25
Camera 5	CK-8	Camera 16	MKH70
Camera 5A	421	Camera 16B	2xECM77
Camera 6	421	Camera 17	441
Camera 7	568	Camera 17B	*568 + ATM25
Camera 7A	421	Camera 18	568
Camera 8	MKH60	Camera 19	441
Camera 9	568	Camera 20	568
Camera 10	441	Camera 21	568
Camera 10B	*568 + ATM25		
Camera 10B(2)	568	*568 for car approaching camera	
Camera 11	568	ATM25 as car passes cameras	

communications. enabling everyone to know exactly who is coming round the bend next.

SRC Sports may have lost the rights to produce the Canadian Grand Prix bit. as is now common in independent produc- page 58 >

tion, the corporation was still involved by providing the technical facilities, Quidoz explains. 'The situation today is not like a production with its own facilities. When F1 Canada decided to keep the rights to the race, it still asked SRC Sports to provide official technical support for our show.' SRC's input consisted or all technical personnel and the OB trucks that made up what was designated as the Pool, where the clean race feeds were produced.

When stereo television was first proposed during the late 1980s, sport was cited as one of the areas that would benefit from 2-channel sound. In recent years some broadcasters have gone further than this, adopting surround techniques to encircle their viewers with the action. Which is why it comes as an initial surprise that the Grand Prix is a mono affair. 'It's not stereo yet,' confirms Quidoz, explaining, 'because re-equipping would cost millions and millions. The CBC networks use satellites to distribute programming to all the stations in the Canadian provinces, which is something huge like 48 to 50 downlinks. At the moment it can't be justified to take public money to upgrade to stereo.'

Just because the soundtrack is mono does not mean that the F1 audio production is not an involved affair. Overseeing the sound and communications for the Grand Prix was audio engineer Serge Parent.

'There are about 34 microphones placed around the track,' he explains. 'We use dynamic units for cameras close to the track and condenser shotgun for cameras that are further away, which can be up to 1,000 feet.' All microphones correspond to a camera, some being mounted on or near cameras, others being placed on stands on their own. The majority of mics used at Montreal were Sennheiser and included 416s, 568s, MKH60s, 421s, MKH-As and the AKG CK8 electret shotguns.

As with any audio production, a Grand Prix soundtrack is built up from a foundation, with elements being added and removed as necessary. It is hardly a case of opening the faders and just transmitting the noise; microphones are placed next to cam-

eras so that the sound corresponds to the shot. And all of this is done by hand, with two operators working two separate consoles in the audio-video control truck. One worked a 40-input Studer 993 to create the overall race mix, which was based on two MKH60s mics mounted on the roof of the casino that sits in the middle of the racetrack. Spot effects were brought in as directed. Another operator handled ancillary audio, including the pit lanes, the helicopter and the FOCA RF feeds from the in-car microphones.

'Everything is mixed by hand,' Parent comments, 'rather than using a switcher to follow the video. There is a video routeing switcher that has a display of tally lights, which indicate which camera is currently going to air. The track operator will listen to the director and watch the lights, mixing the relevant channels in and out as things go along. We choose to do this because it is smoother than switching with the cameras, which can sometimes produce a distinct 'click' sound when the shots change. The operator will always cut the mic when a car is far away from the camera.'

The FOCA in-car mics came in on three lines to the second console, a 14-input Allen & Heath, which also dealt with the pits and the helicopter feed. Two OB vans, one from CBC in Toronto that fed the English language network, the other provided by SRC Sports for the French language network, formed the basis of the Pool. This also sent feeds to the attending international broadcasters, who were additionally provided with 30 booths containing headsets for commentators and consoles for feeds to and from their home country.

While this audio setup was an extensive one, Parent regards the communications system as being more complicated and, ultimately, more crucial to the overall running of the production. Intercom and talk-back was handled by a 50 in, 60 out RTS-Telex programmable matrix. 'The Grand Prix is the biggest production because of the distances involved between the cameras and the number of mics.'

The circuit itself is around 2km long, with cameras positioned around it, which called for secure communication between the remote sites and the control position. Of the 22 cameras around the track, ten were controlled by CCUs, with further links for the RF cameras. 'There are a lot of communication connections between the Pool and the international booths,' explains Parent. 'In addition there are telephones and fibre optics, which includes 24 audio lines.'

All of this went some way to capturing the drama and excitement of the Canadian Grand Prix. Michael Schumacher won the race and regained his lead on the championship board when the event was stopped due to a spectacular crash in which Frenchman Olivier Panis' Prost-Mugen-Honda hit the barrier 14 laps before the end. Panis escaped with the help of the considerable medical support which is a part of every Grands Prix, but damaged his legs. Jacques Villeneuve span off at the end of the second lap, blowing his dream of winning at his home circuit and prompting him to beat his helmet with his fists. The TV cameras caught everything... ■

www.americanradiohistory.com

The Sharp End

Car to pits comms are crucial in Formula 1—how it is done is a secret

There can be few more unfriendly places to place mics or cameras than on a Formula 1 car. **Dave Foister** gives the 'petrolhead' perspective on race broadcasting

IN THE WORLD of Formula 1 motor racing, nuts and bolts details of the on-car sound and vision arrangements are hard to come by. It's easy to see that capturing and sending these from the hostile environment of an F1 car represents a considerable technical achievement apart from contributing a great deal to the atmosphere of the television broadcast of an F1 race. But such matters are handled exclusively by FOCA (Formula One Constructors' Association), which plays all things technical very close to its chest. None of the actual F1 teams was in a position to help me very much with my quest for enlightenment, referring me instead to Formula One Administration, who clearly regards the arrangements as a trade secret—for the time being at any rate. This is a measure of the enormous power FOCA holds over F1, although before anyone cries foul it is worth remembering that without the determined efforts of FOCA, and supremo Bernie Ecclestone in particular, F1 would not have the extraordinarily high profile and worldwide TV following it currently enjoys.

Onboard microphones present far fewer aerodynamic problems in terms of effects on the car's performance than cameras, although it is believed that this year's microphones are a well-known recently introduced subminiature model that combines negligible size and weight with the necessary huge SPL handling capability. The noise of a 750BHP V10 at 17000rpm has to be heard to be believed, as does the thump it emits when gears are changed, which is akin to the explosion of a very large firework. RF transmission to FOCA's facilities is believed to be handled by one of the top players' current high-powered multichannel UHF systems, allowing easy frequency reassignment in the widely differing RF environments in the several territories the F1 circus visits. Actual allocation of frequencies for this and other areas of the broadcast spider's web is handled by FOCA's own specialist RF department. Similarly, RF communication between cars and pits is centrally handled, but not by FOCA itself; an interesting sideline is that while some forms of motorsport (British Touring Cars, for example) are able to provide feeds of their pit-to-car radio traffic, the nature of a Grand Prix with its complex pit stop strategies and other sensitive information means that

Cockpit information is communicated directly to the pits

< page 61 this is not possible in F1. It is common for the teams to log this traffic, however, along with the telemetry information which transmits comprehensive data about the car's behaviour to the pits every lap. Occasionally carefully screened snippets of this radio traffic make it on to video releases about the season's racing, underlining the action at crucial moments.

All decisions as to which cars will carry cameras and microphones for a particular Grand Prix are made unilaterally by FOCA immediately before the first practice day for the race. FOCA, in conjunction with the FIA (Federation Internationale de l'Automobile) sets the regulations for the car specifications, which can vary considerably from season to season in the interests of safety and fairness; one of the stipulations is that all cars must have camera mounting points in several standard places, and FOCA effectively turns up in the pits and fits the kit without the teams having any say in the matter. The most commonly used position is right on the top of the engine air intake, the place where it perhaps has least effect on the car's handling; a certain small latitude of final positional adjustment is allowed, and the actual mounting consists of two bolts and a wire.

Cars not carrying cameras carry compensating ballast instead to remove any weight penalty incurred by the chosen cars, but another consideration is the aerodynamic effect of a car-mounted camera. The aerodynamics of a modern F1 car are as critical to its performance as any other single factor, perhaps more even than the engine and tyres. The downforce generated by the wings at straight-line speeds is around three times the weight of the car, leading to the mind-boggling fact that the cars could run upside-down on the ceiling of a tunnel if up to sufficient speed. Huge importance is attached to the airflow over the front wing,

Camera mounting impinges on aerodynamic efficiency

which besides holding the front wheels down on the track also affects the behaviour of the rear wing and even the airflow into the radiators and hence the efficacy of the cooling system. It cannot be denied that placing a camera in this airstream, streamlined as the housings are, affects the aerodynamics, and the second most popular site is right in this airflow, on the side of the car behind the front suspension. To compensate for this, there are reported to be moves afoot to make all cars carry dummy camera housings, not just the weight, to eliminate the possibility of unfair disadvantage.

The clarity of the pictures from the onboard cameras says much about the behaviour of the cars themselves. An F1 car's suspension is so stiff that the tyres have more give in them than the springs, and this shows in the absolute steadiness of the image of the car in front of the camera—there is never any jerking or bouncing of the picture unless the driver hits something he shouldn't. It used to be the case that an onboard picture could never be sustained for more than a few seconds without breaking up; for reasons best known to FOCA this has improved dramatically this season, although the tunnel at Monaco will always present a problem.

The fact that any video releases containing sound or pictures derived from the cars also tend to come under the direct control of Formula One Administration further underlines the picture of an extraordinarily tightly-run sport with never an opportunity missed of capitalising on the effort put into its operation. This is further reflected in the heavy investment in new digital multichannel TV facilities which FOCA may well be among the first on the air with, as the format has clear benefits to the race enthusiast. If and when details become available we hope to explore the technology and its application to the continuing promotion of the F1 circus. ■

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Power
and corruption

Where most movies strive to achieve seamless perfection in every aspect of their production, one curious liaison between Eastern and Western cultures is identified by its chosen imperfections. Albert Lord gives **Richard Buskin** an insight into the appeal of the *Power Rangers*

HEN ALBERT LORD was first brought in by LA-based production company, Saban Entertainment, to edit dialogue on *Turbo: A Power Rangers Movie*, he expected to be doing just that. The job of cutting, inserting. and, perhaps, looping where necessary was on his mind. He certainly had not anticipated the scale of ADR necessitated by some of the production considerations until the job was underway.

'When I first viewed some tagged-together scenes in mid-January they really didn't have a movie.' he recalls. 'There was no beginning. middle and end. It was just a case of "Here's what we've shot so far". Because there was an accelerated expectation to deliver this film within a matter of weeks. there was a lot of concern about whether or not it could be completed in time so. they brought me in to start identifying what needed to be done. and what immediately became apparent was that the production sound had been recorded

under very difficult circumstances.'

Anyone with any experience knows that this kind of diplomacy usually spells trouble.

Massively popular around the world, the *Power Rangers* TV show is produced on two continents. The live-action martial arts scenes and animated robotics are shot in Japan, while everything else, including the dialogue. is taken care of in the States. Now. if you've ever watched the show and wondered why the spoken word doesn't appear to be in sync with the characters' body movements (even for those wearing masks). wonder no more... 'There's a high demand for product and not a lot of time in which to provide it.' asserts Albert Lord somewhat predictably. The next revelation, however. catches me off-guard. 'Part of the show's attraction is the kind of cheesiness that the action portrays—and there's an expectation for the sound to complement that.'

Can this be this right—on-screen antics benefiting from sloppily misaligned dialogue?

'That's the basic concept in Japan.' confirms Lord. 'I remember cutting dialogue for an amusement ride in Kobe which featured the underwater adventures of a submarine team. There were two robot characters and then there was this heavy animated face providing directions from the side of the screen. I took a week cutting the Japanese tracks and ensuring that everything was dead in sync with the animated facial movements. Then they brought in the producer and director to take a look at what had been done. but when I played back the ADR I was told it was all wrong—it was too much in sync.

'Apparently. Japanese culture wouldn't buy into this thing talking perfectly. If it's an animated object then it has to look wrong. otherwise it'll throw off the whole believability factor.

'In line with this way of thinking, and also because of the really tight schedule that I mentioned. the actors on the *Power Rangers* show have been trained and grilled to cut 20 to 30 lines of ADR in an hour. They usually just rip through it like crazy. and so when we came to do this movie it was very refreshing for them to know that they could take their time.'

With Twentieth Century Fox distributing the *Power Rangers* feature film. (as well as its predecessor, *Power Rangers: Race to the Volcano*). certain big-screen production values had to

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Left: The evil space pirate Divatox, Hilary Shepard Turner, and her two henchmen prepare to scheme be observed. The result was that dialogue and action ended up in sync with one another. But we're getting a little ahead of ourselves in the story here. After all, for the dialogue to be in sync with the action needs some dialogue, and when Albert Lord started out on this project he really didn't have a whole lot of that to work with.

'We sat there counting the lines that could be retained, and out of the whole movie I came up with a grand total of four that were usable,' he says. 'In effect, every single bit of dialogue—the live action as well as the talk-

'Cartoon characters all have in-your-face voices. There's no perspective, whereas with the live action we had to treat the microphone and ensure that it was a certain distance away. They bought a brand new Sennheiser 416 specifically for this project'

ing-head material—had to be redone, and that was down to the quality of recording. I mean, the sets were rife with production problems. They'd been trying to cut this movie with the pacing of a television show, and so the setup times had been really quick. As a result, you could hear footsteps, as well as the sound of those heavy costumes and masks that the actors had to wear and move about in.

The dialogue editor's job is to go in and find lines that don't have extraneous background noises, or, if they do, then at least ones that can be matched to the on-screen action, but this was impossible. Everything in the production was a problem. Not only the "helmet head" dialogue but also the on-camera stuff was done poorly. The actors sounded like they were talking with socks stuffed into their mouths—you couldn't understand a thing. It was obvious that we'd have to work on the entire picture, and at that point I said, "Wait a minute. When do we have to deliver this?" They said, "Well, the delivery date is in the middle of March...". That meant we had about a couple of months, but then we also had to do a temp dub in six weeks. I said, "There's no way that I can do this entire film—program it and then shoot it [dubbing lines to the on-screen action] and then edit it—in that amount of time". They wanted to know if there could be a compromise. I said, "Well, let me auto-assemble the production and isolate the lines on the principal character, Divatox". She's the bitch of the movie, and she was played by Hilary Shepherd who was really incredible on-camera.

'I used the Fairlight to auto-assemble the production sound, and isolated her lines for the first temp dub. We then brought in the actors to loop lines that were on-camera. However, you have to remember that the story was still being developed, because the editing hadn't been finished. They were still trying to figure out where the scenes would go, and so we

needed some cover and were adding words.'

As Lord could ill afford to lose any time at all trying to communicate his thoughts to the actors, the director who he had previously worked with on the TV show was brought in to direct the ADR sessions at the Saban studios while Lord himself supervised them. With all of the shows being produced there, Saban is a busy facility, so a studio had to be specially booked.

'Everybody treated this project as a big deal,' says Lord. 'I was the one who was determining if a performance was correct and the sync was okay, while the director's role was to interface with these young men and women; to get them to slow down but not freeze up. It was a great pairing. He was in the room with them and I was at the console making sure that the sonic qualities were right. To that end, being that this was now a movie, not a television show, we had to ensure that the microphone placement was actually correct.

'When they do animation it's a different miking technique than for live action. If you notice, cartoon characters all have in-your-face voices. There's no perspective, whereas with the live action we had to treat the microphone and ensure that it was a certain distance away. They bought a brand new Sennheiser 416 specifically for this project, and then we also used a Neumann 46 that was in the studio just to get a difference. For instance, some of the girls' voices didn't have the projection, and so shooting them with the 416 would lose some of the intimacy. There again, certain scenes involved them exerting effort and fighting, so we'd vary what we used and where we placed it.'

Nevertheless, the 416 was the principal microphone, and as every single one of the lines had to be replaced there were hardly going to be any problems with regard to matching production sound.

'Given the tight schedule, I didn't have the time to spot all of the groups,' says Lord. 'A lot of background sounds had to be recorded for this temp and they had to be done quickly, and so I had a wonderful set of actors who came in for a marathon session from 7 o'clock until 11. During that **page 66 >**

Albert Lord among his studio equipment: Sony XB12 Trinitron 35-inch TV; Soundcraft Ghost console; Fairlight MFX3 digital workstation; JBL 4412 and 6208 speaker systems; Urei bi-amp; Yamaha SPX90; MasterRoom reverb; Gain Brains; Sony SVP 5600 ½-inch VHS deck

time we did all of the group ADR for the movie. I'd beat them up but nobody complained. and it was a wonderful experience in terms of what I got back. It was done for the temp and made it through to the final, and that was also partly because we did a very meticulous edit job.

'Often. when you see a television show with MTV-style quick cuts. that's partly because the young actors are unable to repeat five lines without fumbling. I've worked on them and I can tell you. they'll do one or two lines and then they'll mess up. The editors therefore wind up having to cut back and forth for reactions or whatever. and so that jerky cutting style is a compromise between the visual pace that they're trying to establish and the ability of the young actors to carry several lines of dialogue. They haven't been trained as extensively as their older colleagues who can do a long monologue. or at least a paragraph, without breaking down.

'In this case, however. the *Power Rangers* movie didn't have that real fast pace unless there was a fight scene, and then we'd have to go in there and get grunts and other reactions to make it believable. It was really

Poolside Post

ALBERT LORD has his own home-based post facility in Northridge, California. Northridge was at the epicentre of the January 1994 Los Angeles earthquake, and Lord's entire house was lifted up into the air before returning to terra-not-so-firma in a condition that was somewhat different to how it looked about a minute before. It has since been extensively rebuilt, and, while Poolside Post is currently located in one of the living rooms, it will shortly be relocating to the outsized garage.

The PP setup consists of a 24:8 Soundcraft Ghost console with an automated MIDI controller to enable loading to and from a Sony PCM 800.

'I used to have an Akai 1214, which I would use for music demos and whatever else, but it didn't have quite enough inputs,' says Lord. 'So, we upgraded to the Ghost and I'm really happy with it. It's very clean, almost like glass. Very slick.'

A Fairlight MFX3 Plus sits at the heart of the system, while in terms of monitoring Lord has opted for nearfield JBL 6208's, together with a 35-inch-screen Sony Trinitron XBR for sound-to-picture work.

'It took me a while to get used to the JBL's because they're flat.' he says. 'Most of the other speakers that we've been using have got a pre-emphasis around the vocal range to provide brightness and clarity, and so my initial response to the JBL's was, "God, these sound kind of dull". However, after having worked on the Power Rangers movie, where I had to put in 15 hours a day, I appreciate the fact that there's no brilliance to them. I mean, after eight hours my ears were tired.'

For the cutting of effects there is also a

The Ghost console and MFX3 double for music tracking at Poolside Post. Albert Lord (3rd from left), producer Kris Coleman (far right) and members of group Profound Sound

secondary system. comprising a Fostex Foundation 2000 and PCM 800, with an HP1535 for backup and a Panasonic SV3800 for DAT downloading.

In addition, there is the usual smattering of FX gear, from Urei 1176s, an SPX90 and an Alesis Microverb II, through to eight Gain Brains that have been configured for stereo Foley dialogue, music and effects mix stems Aside from utilising his Fairligh on Turbo: A Power Rangers Movie, other films that Lord has recently worked on with this setup include Independence Day and Bad Day on the Block, as well as the upcoming releases Best Men and Eve's Bayou.

'The industry is getting to use a lot more small independent post facilities,' says

Albert Lord. 'There are large independent houses and there are also postproduction facilities on the lot, but the demand for product is so high that they're now looking to the likes of us for overflow work. People like myself have taken to working from home because otherwise, with a schedule that runs from 8 o'clock in the morning to 11 or 12 at night we would never get to see our families. We're literally living on the project and that has to affect the quality of living. Our union is now looking at ways of addressing that, and there's now talk of putting together a network of the small independents.'

important for the whole thing to feel right.'

Locked into a deadline agreed with the devil, both Albert Lord and his fellow ADR editor, Mike Garcia, had virtually to work around the clock. 'It was my mission to accomplish the goal,' says Lord. 'At the start, Clive [Mizumoto, Saban's Vice President of Sound Operations] told me, "I don't think we can do this, Albert," and I said, "Well, is that a challenge?", to which he responded, "Er... yeah".

"Okay," I said, "here's what it'll take: I'll need my Fairlight MFX3 Plus and I'll need a room". They said, "Fine. Here's the room. Bring in the Fairlight," and if it hadn't been for that we never would have done it. The speed of the Fairlight allowed me to do the work quickly, and I don't know how else I could have managed the files so efficiently. I cut on

'On camera there would be all of this lip movement, but when it came to redoing the lines in the studio the energy level was just a bit lower. You know, she [Hilary Shepard Turner] wasn't in costume, she didn't have the high heels and the whips'

an Avid for *The Preacher's Wife*, and it's not the same. The speed of the editorial between the two is so different. I've also used Pro Tools and the WaveFrame, but the hottest box out there with both the quality and the speed is the Fairlight... and I don't get paid by Fairlight to say this!'

Lord is clearly pleased with the resultant *Power Rangers* movie, and this is not only in terms of how the post crew rose to the challenge when, with changes coming in daily, they achieved all of the dialogue and sound effects editing.

'When Mark Ettel did the mix they put him under the gun as well,' Lord asserts. 'He had five days to do this movie—"This is a temp dub?" "No, this is a final"... So, he did the best he could. After all, there had to be compromises just to make this ordeal deliverable, and overall I'm happy with what we did.'

A large part of that pleasure was derived from the attitude of the actors. Stress and pressure really were key words when it came to the ton of ADR work that had to be completed, yet they proved themselves very easy to work with. That is, in terms of their willingness and cooperation, if not always their ability to understand direction.

'They weren't used to being given complicated directives,' he says. 'Like, "Stress the 'r' on this movement, because you've got a breath coming right behind it and I don't want to hear the air." They'd go, "What?" So, I'd have to explain, "Well, what you want to do is say the line, hold your breath and give me an exertion". They weren't at all used to that kind of acting in the studio, because normally there wasn't that much pressure to get everything in sync with the right energy.

'In that respect, what was particularly hard

Albert Lord

A GRADUATE in both Electronic Engineering and Broadcasting Arts, Albert Lord engineered and served as production director for a couple of San Francisco radio stations between 1973 and 1989. Thereafter, he worked for six years as a sound editor at world-renowned Los Angeles post facility, Todd-AO, where he took care of ADR, Foley and music on a huge variety of TV and film productions. He was also undertaking outside work as a sound designer, editor and recording engineer, as well as an effects mixer and boom operator on various TV sitcoms and game shows.

The list of awards is too long to list here, but suffice it to say that, just for his TV work alone, Lord has been nominated for six Emmys and won twice, for *Law & Order* and *NYPD Blue*. Since 1995 he has worked as an independent.

was the Divatox character, who had been filmed in close-up with these really exaggerated lip movements. As the big villainess she'd be shouting, "What do you mean, I can't do that? I want those Power Rangers now!". On camera there would be all of this lip movement, but when it came to redoing the lines in the studio the energy level was just a bit lower. You know, she wasn't in costume, she didn't have the high heels and the whips, and so she had to get back into that character. Anyway, at least they didn't have to reshoot any of her scenes, because by that time she also happened to be pregnant' ■

Passionate Hertz

Putting the hyperbole to the test in a recent recording of Gershwin's *Rhapsody in Blue*, producer John Atkinson made simultaneous 16-bit 44.1kHz and 24-bit 96kHz recordings. **Tim Goodyer** yields the inside story

DVD's 24-bit, 96kHz audio implementation is now a *fait accompli*. Major DVD player companies are behind the format, with it burnished as a part of the DVD-video specification for audio reproduction, alongside Dolby AC-3 and MPEG2.

But how does 24-bit, 96kHz perform when used as a production or postproduction medium? This question is being asked by more and more producers as analogue tapes from productions as recent as the 1970s are facing last-ditch rescue measures, maybe possible only for one more time. Does 24-bit, 96kHz production offer the same quality as the best of analogue? Are the advantages of digital technology in postproduction and archival really that concrete?

Having attended a number of demonstrations over the past year, the difference John Atkinson had heard between the current 16-bit/44.1kHz CD-DAT format and 24-96 was not something he could easily forget. Atkinson's past experience as a professional musician, combined with his current engineering and production work would have been enough to ensure his interest in exploring the issue further but his position with *Stereophile* (both the specialist American hi-fi magazine and its recording label) gave him the opportunity to take a close first-hand look.

The event was a recent recording session of pianist Hyperion Knight performing Gershwin's *Rhapsody in Blue* and other piano works in Albuquerque, New Mexico for the Stereophile label and here Atkinson arranged for parallel 24-44.1 and 24-96 recording chains in order to experience the new technology first hand.

Piano recordings are regarded by many as ideal vehicles for testing new recording technologies. Capturing the rich and complex harmonics of a solo Steinway Model D concert grand with accuracy is a challenge to the recording engineer's skill and resolve, especially when it is to be done in an acoustical environment.

Like many purists, Atkinson's sessions are engineered so that the performance space becomes an important element of the performance itself. He records almost all his productions in natural acoustic spaces. In addition, to fully explore the capabilities of a new technology whose promise is much better accuracy and resolution of the mic feed, every detail in the preparation of the session that could effect what that mic feed delivers must be scrutinised to maintain the highest possible degree of integrity.

The setup of the session was uncluttered and yet effective. With the piano placed and tuned in mid-stage, the main mics were carefully placed in the hall, about 10 feet above the keyboard and 9 feet away from the front edge of the stage. The outrigger mics were placed 8 feet in the air and about 15 feet apart. The surround channels were picked up by a pair of mics facing away from the stage but placed vertically coincident with the main pair. The minimalist mic placement techniques ensured the presentation integrity of the performance as well as the acoustic space.

In fact, Atkinson's technique is so purist, he didn't even use a mixer—by directly recording four sample-sync tracks to a Nagra-D, John balanced the final mix to 2-track stereo from his main B&K 4012 ORTF pair with B&K 4003 outriggers in his Sonic Solutions workstation to get the best combination of imaging specificity of the cardioids and the low-frequency linearity and accuracy of the omnis.

To preserve maximum signal integrity, Atkinson placed his mic preamps and analogue-to-digital convertors close to each other to avoid long analogue interconnects. The main pair and the outriggers were both fed to Millennia Media HV-3B preamps and the surround B&K 4011 outriggers were connected to a Forsell valve mic preamp. From the mic preamps, a custom multi split out individual feeds for connecting to a 24-bit dCS 900D and a 20-bit Manley Reference A–D running at 44.1kHz and three 24-bit dCS 902 A–Ds running at 96kHz for the surround recording.

The digitised mic signals ran from the stage box through 100-foot Canare AES- **page 70 >**
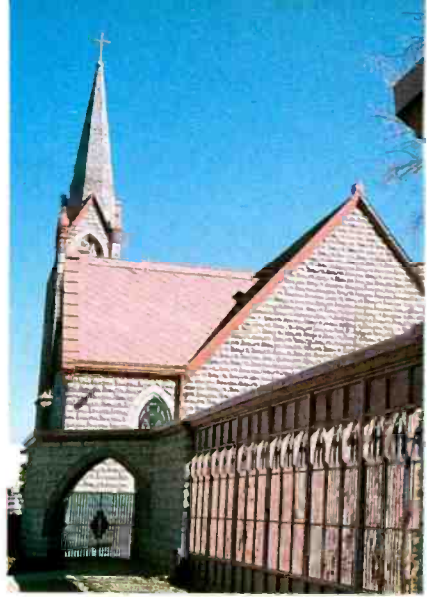
Top: The Albuquerque, New Mexico venue.
Bottom: Hyperion Knight at the piano

< page 69 EBU cables to the control booth where they were connected to the appropriate inputs of multiple sample-sync'd Nagra-D digital open-reel recorders. Since the Nagra-D is capable of confidence playback in real time. the playback head outputs were used for monitoring through 96kHz dCS Elgar digital-to-analogue convertors. the consumer version of the dCS 952 D-A.

The Elgars were chosen for monitoring the session over dCS 952s because of the convenience of the 48-bit precision digital volume control. To complete the monitoring chain. the analogue outputs from the Elgars fed a multitude of Stax professional electrostatic headphones.

Tape started to roll, watchful eyes and ears were concentrated. Everyone in the control booth was now at a heightened level of equilibrium, with a sensitivity to anything that could impact the recording. Take after take. the equipment recorded everything that was going on in the hall.

Like any other location session, from the 4 o'clock traffic jam outside to a broken piano hammer, all the usual gremlins came out to join the party. Three days later, 24 hours of recordings were laid down on tape with enough takes to produce not only the current CD release but ample material for a surround 96kHz DVD release as well.

In postproduction. Atkinson loaded the master tapes into a Sonic Solutions workstation with the cardioid tracks at full range to preserve the imaging characteristics. With the treble rolled off the omni tracks, they were blended together to give a 2-channel presentation of the recording offering pinpoint spatial cues and appropriate size and weight in the lower registers. A Meridian 518 Mastering Processor was used to redither the 24-bit to properly truncate it to 16-bit before the CD masters were produced.

Was there a significant difference between 44.1kHz and 96kHz? Was all of the meticulous attention paid at various stages of production worth the effort? Where were the sonic differences? How did the artist feel?

'Having been able to carry out similar comparisons with the two sets of *Rhapsody in Blue* cardioid master tapes,' Atkinson comments. 'I am astonished at the difference. The response of the B&K microphones we used for the rhapsody is about 10dB down at 30kHz, while tweeters rarely extend much above 30kHz. And my hearing cuts off above 16kHz. You would think, therefore, that extending the recorded bandwidth by just over an octave, from 22kHz to 48kHz, would be irrelevant.

'Yet in level-matched comparisons the 96kHz-sampled tapes had a palpability, an ease to their sound, that the 44.1kHz versions, as good as I thought they sounded in isolation. only hinted at. The clarinet, for example. sounded more like an instrument with a pipe of air vibrated by a reed and the player's embouchure. And the sense of the acoustic space captured on tape was more tangible. There is no doubt in my mind that the higher sampling rate gives a sound more true to the mic feed. A telling indicator was **page 72 >**

Second from top:
Positioning one
of the B&K mics.
Bottom: Engineer
Steve Lee sets up the
ranked Nagra-Ds

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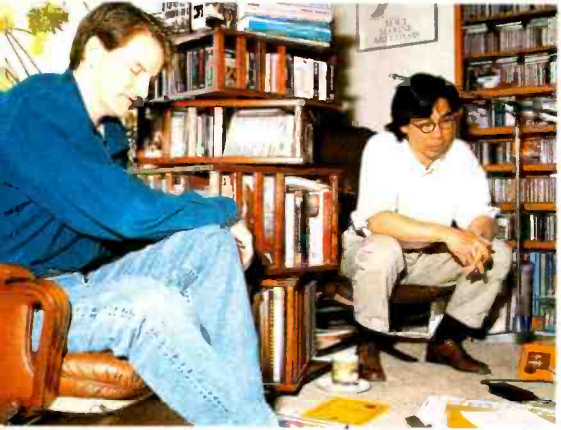
the deflated look on Hyperion Knight's face when I switched back to a 44.1kHz tape after listening for a while to a 96kHz tape while we were deciding what takes to use for the rhapsody master. If you only hear the CD-standard 44.1kHz sound, you won't be aware of what you're missing. But listen for a while to the same music with the higher sampling rate and you really resent it being taken away.'

Atkinson's comments sit well beside those of other industry leaders who have had first-hand experience of 24-96 technology for some time. Bob Ludwig of Gateway Mastering in Portland, Maine recently said: '...it is easy to hear the benefits of moving to 96kHz sampling frequency with greater than 20-bit dynamics. Upon hearing a comparison with 44.1kHz, 16-bit, one no longer thinks the music sounds "digital",

Left: Producer John Atkinson, Pianist Hyperion Knight, and Engineer Steve Lee listen to the finished recordings

it simply sounds accurate. The increase in sound resolution is not subtle to me. There seems to be some kind of perceptual hearing threshold that is crossed with the higher sampling frequency.

'I got to hear the sound difference between 44.1 and 96kHz many years ago when I used the Mitsubishi X-86HS digital tape machine. So far I have heard four different manufacturers implementation of the 96kHz format. In every case the 96kHz was preferable to recordings made on the same machine at 44.1kHz. It is important to remember that it is difficult to make these new convertors correctly. I have listened to A-B demonstrations using 96kHz convertors made by dCS and 88.2kHz convertors made by Pacific Microsonics. To me, these manufacturers are some of the best and most knowledgeable.

'At Gateway Mastering Studios we are looking to create as many DVDs as possible with 24-bit, 96kHz sound. We hope manufacturers will implement a 96kHz digital output on the DVD player so one can use a great digital to analogue convertor. I think listeners will be amazed by the final result.'

Bob Katz, who very successfully engineered most of the Chesky Records albums over the past 10 years, said: 'Regarding 16-44.1, 24-44.1 and 24-96 respectively, I have this to say: strained, constrained, unstrained; closed-in, more open, transparent; narrow, wide, spacious; and small, deeper and deepest.

'We have switched all our production over to 24-96. It simply sounds much better, with much higher resolution,' enthused David Chesky, the A&R force behind Chesky Records.

Perhaps most importantly, Hyperion Knight, the pianist at the Albuquerque recording session of *Rhapsody in Blue*, had this to add: 'I had heard from others about the superiority of digital sound at 96kHz, but I was still not prepared for the enormous differences in resolution, ambience, and dimensionality that I heard between the two master tapes. I hope that enough momentum will gather to force the major labels to adopt 96kHz as the industry standard—and I am glad that my Gershwin recording will have a 96kHz master tape ready and waiting for the day the right playback hardware becomes available.' ■

The Back Room Boys

Some are famous simply for being famous, others find fame through their endeavours, and then there are the Albert brothers. **Richard Buskin** talks to a pair of American producers whose success is in inverse proportion to their profile

FRANK ZAPPA, ERIC CLAPTON, Crosby Stills & Nash, The Rolling Stones, Jimi Hendrix, Joe Walsh, James Brown, Johnny Winter, Jimmy Page, Joe Cocker, Buddy Miles, The Allman Brothers. Ever heard of Ron and Howard Albert?

As producers, engineers, record company execs and studio owners in Miami, Florida, the Alberts have worked with all of these artists, and many more. They have some interesting stories to tell and have developed some innovative techniques from the miles and miles of tape that these guys have played with down the years.

'Making records has always been a wonderful experience for us,' opens Ron Albert. 'Everything's a hit in the studio. There's no such thing as a flop while you're in there, and, unless there's some personal crisis, we're all in a good mood, having fun and doing what we want to do.

'Having said that, the acts that sustain themselves are the ones with the true talent, and there's a real fine line between enormous talent and dislikeable personalities. It seems that the crazier they are, the more talented they are, and once you've figured that out and are able to deal with it on a rational basis, then you don't notice their lunatic shortcomings as much as their ability.'

Back in 1966, Howard was a session keyboard player. One of the Miami studios that he worked in was Criteria, and when an engineer there said that they were looking for an assistant, the elder Albert suggested his brother Ron. The job description was a little fuzzy, and so Ron soon found himself typing labels for tape boxes.

When Howard was drafted and sent to Vietnam, Ron got the chance to engineer. His first gold record was 'Rainy Night In Georgia' by Brook Benton. That was in 1970. Two years earlier Howard had joined him at Criteria.

'I'd been so far removed from what was going on in the States, that when I got out of the army Ron suggested that I also do some engineering,' Howard recalls. 'He gave me a crash course in how to use all of the different microphones, and we started doing sessions the same week.'

At first, the Alberts had different clients and would alternate behind the 4-channel console. It wasn't until a little later, when they had progressed to live recordings, that they began to work together, sitting side by side and operating separate halves of an 8-channel board.

'Things quickly went from 8-track to 16-track to 24-track,' says Howard, 'and whenever we had a large group the two of us would operate more or less like a tag team. One of us would be out there scratching a mic, and we'd each engineer and assist one another. Then, when we were mixing, Ron would take eight tracks and I would take eight tracks. He would take a string section and I would take a vocal and a horn section, and that's how we would ride it.'

'We carried that right on through to our console design,' adds Ron. 'In the early days of MCI's historic rise to the top we had them design a console specifically for us; that was the 500-series. They put the subgroups in the centre, so that Howard, sitting on the left, would have 16 faders in front of him and be able to access the subgroups to his right. Meanwhile, I had 16 faders in front of me on the right side and the same subgroups to my left. That way we could automate our mixes through four hands, and, being that Howard's right-handed and I'm left-handed, it worked really well. We designed several boards with MCI, and eventually that split console caught on all over the world.'

During the late-1960s, while recording acts such as Jimi Hendrix, The Grateful Dead and Frank Zappa at the open-air Miami Pop Festival, Ron and Howard used the original MCI desk which had been custom designed for them.

'Now that really was a unique console,' says Ron. 'It was built in three sections, and the centre section—containing the meter bridge and microphone channels—pulled out. That meant we could throw it into a remote truck and be mobile, even though we'd then have to shut the studio down as half of the board was missing... That console eventually ended up in Pink Floyd's barn in England.'

Around the time of the Miami Pop Festival, the Albert boys also worked with Zappa and the Mothers of Invention on their 1969 release, *Burnt Weenie Sandwich*. Next came *Weasels Ripped My Flesh*, a composite of live tracks and studio recordings from 1967 to 1969.

'Frank Zappa was an enormously talented musician,' asserts Ron, 'and he surrounded himself with enormously talented musicians. It was all about the music and the execution of it. Much of it was preconceived, at least in his own head, but then a fair amount was also improvised, and this was down to the extreme abilities of the players to take the material to a new level and to different places.'

Still, at that point Zappa was content to foster his own creativity in the recording area and leave the technology to the technicians.

'Back in the early days the musicians hardly ever came into the control room, not even to hear a playback,' says Howard. 'There would be playback speakers in the studio and they were happy to listen there. It's not that they weren't allowed into the control room, but most of them wouldn't even think of touching a console. In fact, the first artist who I can remember putting his hands on the console

was Stephen Stills...'

'...and his feet,' adds Ron.

'Stephen wouldn't be afraid to grab a fader and change a level or something but most of the others, while they might suggest this or that, would never actually physically touch the console. There wasn't a law against it, but it's just like you wouldn't walk out into the studio, pick up Eric Clapton's guitar and start playing 'Louie Louie'. There's been a demise in the industry since the musicians have started touching the console. Now they're doing everything at home!'

Alongside their recording successes, the Alberts can legitimately lay claim to the credit for having innovated many of the multi-miking techniques that are in use today.

'That's how we got clients to come to us,' explains Ron. 'We were creating drum and guitar sounds that no one else could create, and, because of Howard's ability to play keyboards, we also had a B3 sound that no one could create. It just so happens that we had a 9-foot Baldwin that sounded amazing, and we experimented with it and learned how to mic, how to EQ and how to limit. We were putting sounds on tape and then onto disc that had never been heard before.'

Nevertheless, both Ron and Howard are quick to point out that, back in the late-1960s and early-1970s, most of the artists who they worked with already had their own sounds before they entered the studio. It was therefore the Alberts' job to capture those sounds on tape, rather than to contrive specific effects.

'We were just trying to make everything happen as smoothly as possible, and to capture the magic,' says Howard. Ron meanwhile explains that the acts who they worked with, from the Allmans to Little Richard, had their musical foundations set firmly in live performance. 'They made records because it was a business and they could make some money,' he says, 'but touring and doing shows was their initial source of income.'

'We basically wouldn't change their music that much,' states Howard. 'We learned from Tom Dowd, and Arif Mardin, and Jerry Wexler that the real key to being a good producer was to make the group think it was their idea if you were going to change something, but otherwise just get things recorded. In fact, we learned that even more so from Crosby, Stills and Nash...'

As the 1960s turned into the 1970s, Ron and Howard recorded CSN's eponymous debut album, as well as the *Four-Way Street* double live set and the *So Far* compilation.

'With them we tried to be as transparent as possible,' says Ron. 'Especially with their vocals. Just get them down on tape. We've had the pleasure of working with such great singers and also with such great guitarists, and, although the magic inevitably comes, you can never tell when that will be. A great guitar player can produce ten hours of the worst crap you've ever heard—I mean, a player in a bar band might play better Eric Clapton solos than Eric himself, except for those magical ten minutes when he's on. The same goes for vocal performance, and so you have to be ready to capture the magic. Our philosophy has always been that tape is cheap. We record

every-thing, no matter how horrendous it is, because two seconds after the crap it may be magical. If you don't record it, that moment will pass and be forever gone.'

And do any particular artists come to mind in that regard?

'Stephen Stills, John Cougar Mellencamp... As producers we wear two hats: first we have to get the artists to the point where they produce the magic, and then, once we've got that down on tape, they can go away and we can spend days, weeks, months getting that magic to sound like it happened in five minutes.'

'At other times, it can actually happen in five minutes,' muses Howard. 'For example, there was Joe Walsh's lead guitar solo on 'Rocky Mountain Way'. We were getting ready to do that solo and he was warming up, and what ended up on the record was the first take warm-up. He didn't even know that we were recording him. He went to do a second one and we said, 'Wait a minute, Joe, you'd better listen to the first thing you did.' He said, 'What do you mean? What first thing?''

'What we'd often do was turn the tape machines around so that the artists couldn't see the red record lights,' says Ron. 'Either that, or before they came in we'd unscrew the bulbs so that they couldn't tell when we were recording. That way, the pressure wasn't on. We'd never use terms such as, 'Take 1. Okay, let's record.' In fact, we would sometimes calmly leave the control room with the tape machine in Record and rolling, and walk into the studio and just hang out with the guys. We'd half-way kid around, make some suggestions, do some arranging, and that would lead into 15 minutes of trying stuff out. They wouldn't even know that we were already recording. Eventually, they'd clue in to this way of working, but it was so informal that it still took the pressure off.'

The Alberts worked on five albums with Stephen Stills—*Stephen Stills, Manassas, Manassas II, Right By You* and *Stills Alone*. During that time, not quite sure as to how Stills' former

engineer had created what they describe as his '3-D acoustic guitar sound that sort of jumps out of the tape', they set about innovating their own.

'That was done with an old Urei 1176 and a Pultec,' explains Howard. 'We'd boost the bass, cut the bass, boost the treble and cut the treble at the same frequencies, and it seemed to work. In fact, even today, whenever we cut an acoustic guitar we still use that technique.'

'However, it's also down to Stephen,' adds Ron. 'He doesn't play with picks, he plays with his fingers, so acoustically he's always had a uniqueness. Additionally, he's playing these beautiful vintage Martins, and if somebody is trying to play the same stuff on a Guild or a Gibson it doesn't have the same effect.'

Another of the acts that greatly influenced the Alberts' early creative development was The Allman Brothers Band, and the albums *Idlewild South* (1970), and *Eat a Peach*, which was interrupted by Duane Allman's death in a motorbike accident on 29th October, 1971. Up to that point only three tracks had been recorded, but the band regrouped behind Gregg Allman and Dickie Betts, and *Eat a Peach* went on to be a massive seller in the US.

'As the Allman Band's reputation grew, so did ours,' says Ron, 'and as a result of our relationship with them a lot of other acts became aware of us. I mean, they literally lived in a camper parked on the studio property. They would get up in the morning, come inside and have breakfast...'

'They were great to work with, really easy,' adds Howard. 'In fact, it was through them that we ended up working with Eric Clapton. He was recording in one room and we were doing some Allman Brothers stuff in another, and we just sort of put them together and that turned into 'Layla'.'

Clapton had already been keenly aware of the burgeoning reputation of the southern guitar player named Duane Allman. Duane had been doing session work with artists such as Aretha Franklin and Wilson Pickett, and Clapton wanted to check out this

Crosby, Stills, Nash & Young; Derek and the Dominoes; Clapton tagged along with Duane Allman

guitarist who sounded as though he was equally comfortable with rock 'n' roll as he was with the blues, Tom Dowd, who had produced Cream, was now working on The Allmans' *Eat a Peach* with Ron and Howard, and he was the link between the two ace guitarists.

'We all went to see a live show by The Allman Brothers Band in Miami,' Ron recalls, 'and when we came back the additional baggage included Eric Clapton. Accounts vary depending on which person you speak to, but I don't remember anything being specifically preconceived.'

Some of my previous interviews with people who have recorded Clapton suggest that the only kind of problem that they have experienced has been in terms of cajoling him into the studio do some work. Once there, he'll get his performances down very quickly. However...

'We never had a problem getting him into the studio,' says Howard. 'We had a couple of problems getting him out of the studio...'

Whereas, when it came to sessions for The Rolling Stones' *Goats Head Soup* album, 'they were a little late arriving,' admits Ron. 'One day I think we waited about 20 hours for them to show up.'

'Yeah, that was the time when Mick Jagger hopped out of the limousine like a bunny,' recalls Howard. 'He hopped into the studio and kept hopping and hopping throughout the whole session. We never could figure out why.'

I think I can.

Getting back to Eric Clapton, Duane Allman, and Derek and the Dominoes' classic 'Layla', Ron Albert recalls that the song was never actually played as one piece. Instead, the first half—featuring Clapton's plaintive vocal—and the piano section were recorded separately and then assembled, with the half-time ending overdubbed on top of the existing piano track.

'The drummer, Jimmy Gordon, played the piano part on the second half of that song,' Howard explains. 'We stuck some microphones in the soundboard, closed the lid, and covered it up with a bunch of blankets. The guitars on that whole 'Layla' session were recorded right next to the piano on top of the piano, to be exact—with little Fender Champs. There were no big amps.'

'We needed those little amps for the sound that we wanted to get, but the band was so loud that Duane and Eric couldn't hear them,' says Ron. 'We therefore stuck them on top of the piano, so that, when they sat down, it was like having stage monitors at ear level. Obviously, there was a certain amount of bleed between the piano and the guitars, but that's

called rock 'n' roll!'

The multiroom setup at Criteria was responsible for a number of 'incestuous' recording relationships being formed.

'Near the end of our tenure at Bacteria —that's just what I call Criteria—we had built the facility up to include four studios,' says Ron. 'The Bee Gees would be recording in one room, Stephen Stills would be in another, and The Eagles in another. Now, it wouldn't be unusual for Stephen to walk from one studio to the other, and for us, within the space of an evening, to have made a Stephen Stills record and also have him guest appear on the Bee Gees' and Eagles' recordings.

'Many, many times, because of scheduling conflicts or perhaps somebody getting an idea in the middle of the night, Howard and I would be recording in separate rooms simultaneously,' says Ron. 'He, for instance, could be doing a vocal overdub in one room and I could be doing a guitar overdub in another —same artist, different song.'

Eventually, Ron and Howard Albert rose to the position of Vice Presidents and stockholders of Criteria, before retiring in 1983.

Why retire? 'We were rich,' Ron replies.

The hiatus lasted a few years. Then, in the mid-1980s, they decided to re-enter the business, but this time around they would not be 'babysitting the big groups.' Joining forces with artist, producer and songwriter, Steve Alaimo, who owned the disco-oriented TK Records, they eventually formed the independently distributed Vision Records in 1987. Today their enterprise also consists of a digital mastering facility, Master & Media, and a recording studio named Audio Vision. Meanwhile, the Alberts have kept fingering the faders, and, having had their fill of being company execs, they are now looking to spend a lot more time back behind the board.

You may have noticed that, during the course of this article, there have been references to their productions as well as to their engineering. Well, this transition actually took place during the mid-1970s, and it came about as a result of them opening their eyes as well as their ears to what was going on around them.

'With the Allman Brothers, Tom Dowd would leave the studio at about ten o'clock at night, and we'd stay there and record until three or four in the morning,' says Howard by way of explanation. 'The next morning he'd come in and say, "Let me hear what you did last night... Okay, that's great. Yeah, let's go on to the next thing." Then he'd stay another few hours and leave, and we'd carry on. So, a lot of the things that we cut or overdubbed were done when the producer wasn't even there. We did years of that, until we said, "You know, we're making these records and we're not getting the credit for them!" That's when we decided to be producers ourselves.'

'Well, at least Tom Dowd came to the studio to check things out,' adds Ron. 'Jimmy Miller, however, wasn't in the studio or even in the same state when we cut 'Heartbreaker' [during the Stones' *Goats Head Soup* sessions], and Bill Szymcyzk wasn't in the studio or in the same state when we made 'Rocky Mountain Way'. Bill's a great guy and we like him very much, but he just had a better manager than we did at that time.

'You live and you learn.' ∎

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Bavaria Film

Responsible for an impressive series of 'firsts' in film work dating back to 1931, Bavaria Film studios continues an aggressive technical policy today. **Yasmin Hashmi** gives a first impression of an unusual German film facility

APART FROM the occasional jazz gig playing trumpet at the local bier keller—as demonstrated by a foot-stomping set at Sony's Munich AES Convention party—Toni Ketterle's job is running the sound department of the second largest film studio in Germany. It's not a recent appointment, and his youthful looks belie years of experience beginning at Bavaria Film studios in 1960, and becoming head of sound in 1975.

The studio itself has been in business since 1919, and performed its first sound for film in 1931, later moving to its present site in a pleasant woodland area in the south of Munich in 1938. As we enter the main mixing studio, Studio A, Ketterle reels off a list of 'firsts' for which Bavaria Film can be justly proud: it generated the first magnetic 4-track recording in 1954, followed two years later by the first magnetic 6-track recording for the film *Oklahoma*. The first stereo production for TV began in 1968, and ten years later, the studio produced the first Dolby stereo mix for the German production of Milos Forman's *Hair*. Bavaria Film purchased the first Neve console with computer automation in 1981, and completed the sound for the cult Wolfgang Petersen film *Das Boot* the same year. As Ketterle is keen to point out, 'The Neve is still in use, but has new Uptown fader automation'.

Ketterle also notes that in 1988, the studio produced the first film soundtrack using a digital audio workstation—a Lexicon Opus—and was first in 1991, to use an Avid Media Composer nonlinear video editing system for editing a feature film entitled *Night Moves*. As far as tapeless audio is concerned, the past four years have seen the studio mainly using Avid's Audio Vision and Digidesign's Pro Tools for 4-track laying.

In all, the studio has 40 cutting rooms: six equipped with an Avid Media Composer 8000, four with Pro Tools III system, two with AudioVisions, and five with FED 8-track MOD systems. The remaining rooms employ 16mm-35mm Steenbeck flatbeds. There are five stages for ADR, two smaller TV mixdown rooms called Edit Mix studios, where a single sound engineer performs tracklaying and mixdown, and the Digi-Centre transfer room, which sup-

ports transfers between a wide range of film, tape and disc formats. There is also an impressive preview theatre that seats 50 people, supports all projection formats and has THX acoustics with Dolby SR or and Dolby SR.D.

The three main mixdown studios are named A, C and D, and each has a high-speed film projector. Studio C contains the 36-channel Neve console, and supports Dolby Stereo and Dolby SR.D, and Studio D has a 24-channel SSL 5000 with instant reset and moving fader automation, as well as audio replay from mono up to Dolby surround. According to Chief Mixer Michael Kranz, the 110m² Studio A 'is the largest mixing studio for post-production in Germany'. It also has THX acoustics, but its most prominent feature is an 80-channel Harrison MPC console with 48 mix buses, 256 x 256 routeing matrix and 32 x 8 monitoring. Of course, Toni Ketterle is quick to point out that 'this was the first studio in Europe to install such a Harrison', and also that about half of the Harrison MPC consoles in Europe are in Germany.

Kranz has been with Bavaria Film for ten years, and mostly works in Studio A. His credits include mixing for films such as *House of Spirits*, *Miss Smilla's Feeling For Snow* and, most recently, a new German movie called *Hunger*. Although the Harrison is occasionally used for two-person operation, Kranz normally mixes on his own, using tape or one of the two audio workstations, depending on the project. He worked on both the US and German versions of *House of Spirits* and *Miss Smilla's Feeling For Snow*, first completing the original main mixes, then the M&E tracks, then the German versions and finally the TV versions. In fact Bavaria Film has a large number of TV credits including crime and drama series as well as TV films.

Commenting on the Harrison, Kranz notes: 'Although there is no such thing as the perfect console, currently, for postproduction, this is a very good solution for film mixing'.

As far as working with tapeless audio systems is concerned, Kranz maintains that 'the big advantage of using workstations is that you can put your ideas into effect, and the ability to move or slip tracks independently is a very useful feature'. He argues, however, that the ability to save time using random access is not the main attraction. 'The most important thing is to be as flexible as possible,' he says, adding, 'What we have here is a good

solution, but like everyone else, we look forward to a common file format.

'Ideally, we would like to be able to take files from anywhere by networking within the studio, and also over a wider area for sound effects.'

Kranz would also like the workstations to support features particularly applicable to film, such as the ability to play in reverse. As he explains: 'Before automation, we did half the work backwards, and although the future lies in hard-disk recording with its random-access capabilities, in order to keep the feeling of a scene, to appreciate its timing, it is important to see it in a linear way, both forwards and backwards'. For this reason, Kranz is in no hurry to use a random access digital video player, although he does recognise its advantages for other applications.

With five ADR stages, it comes as no surprise that Bavaria Film owns a Dolby Fax system, which is mostly used for film work. The studio promotes the service as allowing actors in the German studio to be recorded in a studio in the US, with both ends being synchronised to picture, and the editor in the US giving directions back to Germany via the link.

The adoption of Dolby Fax helps summarise the continuing philosophy of Bavaria Film—it's a genuinely progressive studio equally prepared to use new or traditional technology in pursuit of the best results. ∎

CSR Studios

Emerging from the wilderness of isolation, South Africa is preparing to send a strong musical message to the world. **Zenon Schoepe** visits a multiroom complex in Johannesburg that looks set to play a part in this drive

IT'S HARD not to be moved by your first visit to South Africa. Most have heard so much about the place that they have formed an opinion without needing to go there, and it's one of the few places on the planet where until relatively recently you would think carefully before admitting you'd been lest someone judge you unfavourably.

Visiting South Africa challenges some pre-conceptions, reinforces others and tests your ability to assess afresh against an overwhelming backdrop of a philosophy that was fundamentally wrong. Even in its rain-soaked winter season the landscape and sheer scale leaves you breathless, tainted continually by the wasted potential of the country with its natural resources and the power of its workforce.

Now under black rule it is still difficult to accept it at face value without constant reminders of its past. South Africa undoubtedly suffered tremendously from its years of isolation and from the cruel regime that caused it. It is a country that has been cut off for so long that it has evolved almost independently of the rest of the world, and it is only just starting to recreate itself as the beginnings of the major force that it always should have been.

It would be wrong to suggest that this situation has in any way arrived, but there is still room for further transition. You are continually reminded of what it could be and the fact that what happens in South Africa will lead by influence and direction the rest of the African continent.

South Africa has deep-rooted music that is as varied and unique as its mixture of cultures and it is, perhaps, through this medium that the country will reach out and integrate itself with the rest of the world most quickly and dramatically. Promisingly, its South African Broadcasting Company, recently opened its own record label and Radio Park studio equipped with SSL and AMS Neve consoles, and there's also a small studio community the best known member of which is the ambitious and extreme, although now strangely quiet, BOP complex. Near the top of the band is recently expanded CSR studios in Johannesburg, probably the largest independent multiroom complex in the country. Developing alongside this is an admittedly low-scale, but still significant increase in independent studio activity founded on a bedrock of small-room operations.

The problem with South Africa is that it's been isolated for so many years and nobody even wanted to acknowledge that we existed until about four years ago,' explains CSR studios MD Chris Ghelakis. 'Things are opening up, there is interest in our music and we're getting better at starting to get on a par with

80

the rest of the world. I think we will be able to say something in the next couple of years because there's a lot of talent here that hasn't yet been been developed because of all sorts of restrictions.

'I've travelled quite a bit and been to a lot of studios and we are not playing on the same level as the rest of the world,' he continues. 'We record most of the albums in South Africa for around £4,500 [UK] with maybe twice that amount for a really major project every now an again. That's what we're doing and we're competing with albums from outside that cost a million dollars to make.'

Ghelakis adds that they do what they can but that large investments in recording would be difficult to recover and that exporting their music and getting foreign record companies interested is the key to the future.

The record company infrastructure is a curious one, but perhaps not that dissimilar to what goes on in other new territories. Dominated by the major labels, Ghelakis claims it is easier for them to import Western produce than survive on the local talent, but he nevertheless believes that the situation is changing with local labels being formed and slow turnarounds occurring in the machinations of radio.

'Only a year ago did they start talking about putting local music on local radio—if you switched on the radio all you'd hear is overseas music so what will the local consumer do? He buys overseas music,' he says. 'Funnily enough, since they started playing local music people have begun to buy it, but that had never happened in the past.'

A local music radio quota of around 20% is now being talked about, but Ghelakis says it should be greater: 'In a country like ours with the different cultures and diversity of music that we have, the local quota should be way higher than that, but the majors fight it because they think they're going to lose money. They say they're behind South African music and that they'll record it until they see the budget. They scream about spending 15,000 Rand on

recording an album when they've just spent however many millions on a new Bryan Adams record. That's what we've up against, but I believe that it's all going to change.'

Almost predictably, Ghelakis has his own record labels and production company running alongside the studio. These cover dance music, pop and R&B plus a studio with an engineer dedicated to the production of Township dance African music.

However, getting hold of the equipment and, more importantly, keeping up to date with developments over the years has been difficult, but he has used his frequent overseas trips as information gathering exercises.

'My main concern is that I don't want to buy anything that is obsolete because South Africa is very expensive—it's way more expensive here than anywhere else. There have been many cases where I have have imported equipment myself more quickly than the distributors.

'For the type of music we do it's important that we keep in touch and use the latest equipment—talent is one thing, technology is another and you have to have the technology to bring out that talent.'

His route to multiroom complex ownership came though the traditional route of starting as a musician, building a small studio in flat and working up from there by reinvesting the income.

'It's the same all over the world—you can always tell the good studios because they're the ones that are always working and looking to expand. You've got to be on top of things if you want to be good.'.

CSR is dedicated to South African music which it handles with four studios. Its premiere room, equipped with a Euphonix CS2000 bought a year ago and Genelec 1039 monitoring, serves as its fully commercial studio outlet. The remainder are occupied predominantly by in-house productions. A DDA DMR12 room running with a Fostex 1-inch 24-track is used solely for Township music:

'That's huge in South Africa,' Ghelakis explains. 'The studio has a sound of its own with these huge old Tannoys with enormous bass end and they love it.'

An Amek Einstein room concentrates on pop and dance while an all digital remix programming room centres around a Yamaha 02R. The Euphonix and Amek rooms run 32-track Tascam DA-88 systems and Ghelakis has already ordered four of the new DA-98s for his premier room, intending to move its DA-88s to run alongside the DDA—although he adds that the Township musicians have an ardent preference for analogue. There's also a Digidesign Pro Tools editing suite and he employs a total of six in-house engineers.

The last phase of building was completed in November 1996 when the 02R remix studio was put together and this completed a series of improvements and additions to the studio since its move to the present building in 1992. Previously home to Leading Edge studios —which had spent a fortune, according to Ghelakis, on the building of the main studio—the facility had also had a Synclavier suite and offered enormous potential when the site came up at auction.

'I bought the shell, it was a studio and they couldn't get rid of this place because everybody wanted offices, but it would have cost them a fortune just to rip it all out and start again. I was lucky and got a very good price,' he says.

The main room had originally been fitted with Ureis ('I've still got them, I can't give them away') and these were superseded by a host of different loudspeaker types that were wheeled in for trial.

'The worst thing about trying out all the speakers was humping them around because you've got to hear them in the room,' Ghelakis comments. He settled on the first pair of Genelec 1039s in South Africa. Smaller Genelecs are dotted throughout.

'I love the Genelecs and I'm going to get another pair—I want the 1035s for

Amek Einstein—32-track DA-88 systems are available in the main rooms

Digital remix and programming room with Genelec 1029s and subwoofer

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the Euphonix room and I'll put the 1039s in the Einstein room,' he enthuses. 'In the 02R room I've got the little 1029s with the sub-bass and that's one of the best things I've ever done. They're astounding. The distributors brought them in so we could hear them and they never took them back because I bought them. They're perfect for that kind of music and that size of room. For all this jungle drum and bass you need to hear the subwoofer because if you cut a record on an ordinary speaker you'll blow someone's hi-fi up.

'I heard the 1035s at Metropolis in London and they just blew me away. I've got to have them, they are the standard in serious rooms the world over.'

Studio rates at CSR are an eye-opener. 'We charge around £50 an hour for the Euphonix room—that's £500 a day, all in and with an engineer,' he admits. 'The trouble is if we charge too much we get too expensive for the record companies so they go to the back room guys with their little studios.'

The Euphonix room is attached to a 50-musician studio area with iso booths and this can additionally be connected and run into the live area associated with the Amek Einstein room. The arrangement also works in reverse and there is direct line of sight between the two control rooms.

'One thing you won't see much of in South African studios is lounge areas and restaurants, but that's because nobody is ever around for that long on a session. To be competitive we've got to work fast.' He adds that it's one of the reasons why he chose the Euphonix.

'The desk, once you get into it, is very easy to use and quick,' he explains. 'When you have a high turn-around of projects, like we do, with the budgets being so small, the guys can use it for the backing tracks, and then come back at a later date for the vocals, calling it all back up is so quick. Nobody books a three-month lockout here—nobody has ever done that. You go over a month and the record company has to change its underwear. We end up using that studio for something in the day time and something else at night. You push a button and you're there. Changing sessions on other desks takes you hours. All our outboard gear is on MIDI and we save all that in a sequencer song file. That way you can

The Township room. The music is 'huge' in South Africa according to Ghelakis

finish a session at six o'clock and the next one is starting at half past.'

He claims that it's also a board that is appreciated by all the engineers who use it. 'We like the sound, the computer-based technology and the way it is upgradable because that's where the future is,' he explains. 'It also doesn't get hot and it's modular from a construction and service point of view. It's a huge desk in a small amount of space which is important for us—it has 72 faders with loads of inputs per channel.'

But while Ghelakis is clearly aligning himself with international studio standards he's adamant that his ambition is not to carve a niche for himself on the international studio circuit: 'I don't really want international business I want to develop our local music.' he asserts. 'I want international engineers and producers to come here and work with our local musicians. That's what my studio is gearing up for. Having a South African act hitting No.1 in the UK or America, that's the pleasure. not some big-name Western artist recording a backing track here because nobody ever remembers that. Once that happens it will open it up for all the local musicians. Look at Sweden, they had ABBA and look at all the stuff that followed them and that's such a small country in comparison to us.

'South African musicians have a unique type of feel and rhythm. You can spot a South African record, you can hear it in the sound, it just needs to be refined and made more accessible to the rest of the world,' continues Ghelakis. 'The bass lines, the brass lines, the type of melodies are unique, but still too rural and they need to be Westernised. Paul Simon did it.

'We have a lot to offer particularly with regard to freshness. I've just been in Hong Kong and listened to their music—it sounds like England ten years ago—and the same is true of Japan. That's not Japanese music, that's Western music with Japanese lyrics. South Africa is so different.'

And that last sentence says it all. ■

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US: What is the going rate?

I know there's never a good time to bring this up, but it's time to discuss our studio rates again writes **Dan Daley**

IT'S THAT TIME of year again: that season in which we address the subject of rates. Actually, there is no one particular season to mull over this topic; rather, it's become one of those things that one prefers not to mention at all, so in a sense, no time is the right time. Any hubris about how far in advance a studio might be booked out is quickly turned to harmatia when the issue of how much those days and hours are going for raises its ugly little head.

Here's the deal on this side of the puddle: rates for recording studios emphasising music projects have been, at best, static, and more often than not eroding further. I'll tell you a story. There is a recording facility on the West Coast that was built by a large Asian software developer that was supposed to do music for CD-ROM projects. It became apparent in rather short order that this company could outsource its music needs to people with ADATs in their bedrooms and get it done as

well—we're talking about 8 bits at 12kHz here—for a lot less money than the notes on the equipment were costing them. I was making some routine enquiries regarding a rumoured sale of the studio to one party or another, and in the course of asking some local facility managers about what might be going on, I had one say to me—verbatim —'Why the heck would anyone want to buy a recording studio?'

That's about as blatant as one can make the sentiment, one that runs deeply in the music recording business over here and one, like a long-embedded splinter, that is working itself towards the surface.

Studio time, like the music products it is used to make, has become a commodity based on volume. Anyone can have their own recording system, and, thanks to the proliferation of independent record labels, anyone can make their own record. And believe me, anyone does.

It says a lot about the character of the people who remain in this end of pro-audio that, while their complaints are louder and less subtle, they nonetheless remain in the game. How they're doing that is sometimes as creative as the musicians, producers and engineers who use those studios. One major Nashville facility has implemented a progressive rate structure that stretches potential discounts based on block time bookings out as several months at a time. In a market like Nashville, one producer often does four or five albums in a year and by block-booking a multi-room facility, the producer is assured availability and tech-

nical consistency while the studio is assured of a client. The rate may have gotten pretty low on a booking of months, but it buys time to figure out the next move while at least covering the leases.

Some music studios have bailed out of music altogether, lured by the rates that audio postproduction commands; rates that are, on average, four times higher than in music recording. But the successful aura surrounding audio post is a chimera in many cases. It might look like it from afar, but you don't get successful in post by buying a DAW and converting a large closet into an audio suite. The largest investment in post is in people, because in postproduction clients follow the talent, not the technology. Besides, proximity to a viable post market is a prerequisite to even thinking that post could pull your rates up, and that's something most markets simply don't have.

Another tack some studios have taken is to reach out to the musician communities in their markets, educating them as to the distinction between a studio and an ADAT in a garage. One studio owner in New York actually goes out to clubs several nights a week and hands out a sampler CD of records that have been done at his studio. To a generation that has not had this distinction made to it before, the result can be very effective indeed. For starters, you're bringing in new business—at the beginning of their careers— on the basis of quality, not price. And you're also building a clientele that in many cases will come back over time as their careers

Europe: licence to license

Unnoticed by the majority of its practitioners, changes in licensing arrangements are quietly redefining quite disparate parts of pro-audio writes **Barry Fox**

BRITAIN'S RADIO microphones and OB audio and video links are now under single management. The object is to create a smoother ride on licensing. But it ain't necessarily so.

Until 1st April, responsibility was split between two industry groups, contracted by the British Government's Radiocommunications Agency to create a working dialogue with the gaggle of disparate users, including independent TV production companies, theatres, PA hire groups and installers. The Association of Service Providers Frequency Management Ltd represented theatres and hire companies, with an 11% share of the available spectrum; the Broadcasters' Joint Frequency Management Group had 89%.

Last year the Radio Authority said it wanted to deal with a single group, and asked for tenders on a 3-year contract with option for a 2-year extension. Both existing bodies applied. JFMG dropped its Broadcasters' tag, but ASPFM was 'confident' that its diverse membership provided a better safeguard against conflicts of interest. Despite the name

change JFMG is jointly owned by the Independent Television Association and Commercial Radio Companies Association.

The IRA bomb in Docklands shut the RA's offices, but the RA pressed on and chose JFMG. Despite ASPFM's objections to the RA, and fears that JFMG would not be ready to start running the system, the winner was mailing out tariff sheets, application forms and guidance notes ahead of the 1st April deadline. I got three separate copies through the mail.

Although ASP now has no authority, the company is remaining in business to act as a voice for the independent operators. John Wykes, formerly Technical Director of ASP FM says he waits to see how a management group with such obvious broadcasting heritage will handle clashing claims by rival broadcasters and non-broadcast independents. 'How will a hire company license equipment', he wonders. 'What will happen about policing? To avoid interference from neighbouring users we had conformance officers who would go round our licensed sites, checking that no-one was exceeding their power limit, or using the

wrong frequencies. Although the subcontractor has no legal rights, it can report breaches to the RA.'

ASP will collate any comments from nonbroadcasters and independents who do not like the way the system is working, and then put them to the RA. If nothing else, this watchdogging should keep JFMG on its toes.

I phoned JFMG to ask why the BBC was no longer represented and how JFMG plans to police its licences. There was no-one who knew, but eight days later I got a call back —JFMG had been too busy with allocations to call earlier. Director Derek Connelly said that conformance policing is up to the RA.

JFMG sees this as a real watershed, coming at the end of the year when the last existing licenses—for instance those held by Sky and the BBC—will need renewing. ASP is at Edgecott House, Lawn Hill, Edgecott, Aylesbury, Bucks HB18 0QW, UK. Telephone: +44 1296 770 458.

The whole point of any licensing scheme is that it must be fair and easy to use. But manufacturers making the new high-density DVD discs and players face a legal minefield, and high claims for royalties.

The ten members of the DVD Consortium cannot agree on a plan to pool their patents and create a one-stop licence shop. Three members, Philips, Pioneer and Sony have broken away and created their own pool, which Philips will run. That leaves seven outside the pool. IBM is outside the Consortium, but claims rights on the idea of storing two recordings on the same side of a DVD disc, at different depths. 3M previously claimed to have

progress. The reach out-educational approach isn't a ready-made solution and it takes time, effort and dedication. But it may be the best long-term approach.

One other approach I've seen lately mirrors a larger trend in the States—one of personal fiscal responsibility. In 1996, the US for the first time broke the million mark in personal bankruptcies (what you call 'administration'), and 1997 is estimated to see it reach 1.4 million. The general population has been living far beyond its means for a long time, and who could have blamed them, lured by 75 channels of commercials hawking everything you'll ever need to make you happy.

Studios have experienced something similar over the last decade, pursuing trendy gear to stay ahead of the curve and give clients what they think they'll want. Trouble is, the technology changes so fast now that the flavour-of-the-month has gone sour by month's end, even though the studio is still paying the note on it. I've seen more studios start to financially retrench themselves, paying off loans on existing gear and facing the issue of eroding rates with a lowered overhead. It means biting the bullet, but it's a workable solution for many studios.

Rates are a problem, but there are ways to deal with the problem. And the true measure of the creativity in this business is being seen as studios find the economic solutions that best work for them. It's certainly as creative an endeavour as making records is.

invented this same technique. DiscoVision Associates holds a large folio of patents on optical recording technology. Although Pioneer owns DVA, DVA will issue its own DVD licences separately.

Literally dozens of companies hold patents on MPEG audio and video compression, and ISO's MPEG committee quickly saw the commercial need for a pool of all these patents, with a one-stop shop which can issue a single licence on all patents. The committee initially entrusted US company CableLabs with the job. But all calls to CableLabs are now referred to another company, in Denver, called MPEG LA. The LA stands for Licensing Authority but the 5-person company seems curiously unhelpful. I was told that the only person who could answer questions was a CEO whose name I finally got spelled out as Baryn Futa, previously with CableLabs. Futa was in the office for the day, but on voicemail. I left a lengthy message expressing interest, but never heard back. It took more calls and pushing to get to talk with the Chief Financial Officer, Jim Kluttz who would not say how many patents are pooled, or what companies are party to the pool. 'We are not yet in business, but plan to be ready with 30 to 60 days' he said in mid-May.

Perhaps if I call back in July, MPEG LA will be more helpful. More to the point, perhaps the Licensing Authority will be more welcoming to manufacturers who want a licence to manufacture DVD. For anyone who wants to give it a whirl, MPEG LA is on 303 331 1880, or fax on 1879.

Artistic licence: the DTT picture

Awarding the licences for British Digital Terrestrial Television could have established a vision of a media empire to rival that of George Orwell's if BSkyB had won through writes **Kevin Hilton**

ONCE IN A WHILE something happens to reassure you that, despite everything, there is a God. However you perceive Him or Her there is a Higher Consciousness, a Supreme Being or at least something or someone who occasionally ensures that the Right Thing happens.

It happened during the UK election when the slimy, xenophobic, Eurosceptic (all of which is strange when you remember he's of Spanish extraction) Michael Portillo lost his seat. I was laughing so much I nearly missed an airplane flight—there is nothing like that warm sense of justification, when you know that someone has just got their comeuppance. But it does not happen very often —or so I thought.

The Independent Television Commission (ITC) announced the winner of the British digital terrestrial television (DTT) multiplex licences B, C and D towards the end of last month, awarding them to British Digital Broadcasting (BDB), a company jointly owned by Carlton Communications and the Granada Group.

Television and radio is now in the grip of the consortium; there are very few (if any) services that are wholly owned by one organisation. This is something that will become more commonplace as the digital age progresses, because it will need huge resources—in both programme production and provision and technical services—to sustain the number of channels made possible by multiplexing.

It's not just TV; there will be capacity for audio channels, Internet services, interactive communication, shopping channels and multimedia. It is a large undertaking but there is the potential for large returns, even if things are split two ways. With BDB the two partners are now looking at both bigger investment and, they hope, bigger returns because these two were originally three. But the third wheel, BSkyB, was forced to withdraw by the ITC.

Let's look at the second clause of that last sentence in isolation: BSkyB was 'forced to withdraw by the ITC. Here comes that wonderful warm feeling again. Ha ha ha ha!

BSkyB, headed by You Know Who, has always been disdainful of any attempts to bring it under UK regulatory restraints, working on the very thin premise that because it is uplinked from Luxembourg it is not a UK broadcaster. Now it has come unstuck—the ITC considered that if it were involved in the winning consortium for DTT too much power

> Like all regulators the ITC has been criticised for being soft, but it has finally found its teeth, depriving News International of an in

and control would be vested with one organisation (or, to be more accurate, with one person). At the time of writing I have not heard what YKW thinks about all this—and, frankly, I don't care. Like all regulators the ITC has been criticised for being soft, but it has finally found its teeth, depriving News International of an in to the terrestrial digital scene.

The order of things has not been shaken completely because BDB is owned by Carlton and Granada, two of the most powerful and ubiquitous groups in UK broadcasting, but at least YKW has had his nose put out of joint. Ha ha ha ha ha ha ha ha ha... Damn, I've missed my flight.

FOR THE PAST DECADE or so, television and radio programmes have been stiff with either documentaries or dramas set in the workplace. Just think about *ER*, *Hill Street Blues* and *Homicide—Life On The Streets*, shows where all is activity and characters speak in a barely comprehensible argot peculiar to that situation.

Naturally, emergency room staff and cops are often critical of such shows because they're not as realistic as they would like their viewers to think they are. This is why there is always a technical consultant credit on such productions, because TV. film and radio people know about TV, film and radio, not hospitals or police stations. So how is it that when media people decide to portray their own world they get it horribly wrong?

I recently came across a new children's series, with the plot line of two teenage drum & bass DJs going to work at an established radio station. From the outset it was clear that the programme makers had not gone anywhere near a radio studio during their research. All the DJs went on air with their headphones slung around their necks. Have these people never heard of howl round? The studios themselves sported turntables and NAB cart machines; just about every station now relies on CD and even the smallest private services are now turning to MD or some disk-based system.

It's always the way. The most naive representation of a radio station I ever saw on TV was about six years ago in a series called *Thin Air*. It was set in a London station that had been on air for ten years, yet it had the cleanest studios and consoles I had ever seen. You know there's a sore lack of reality when a coffee mug is not glued to a desk, giving a home to a completely new lifeform.
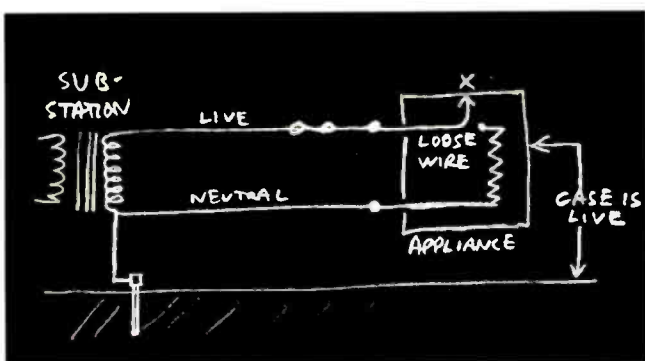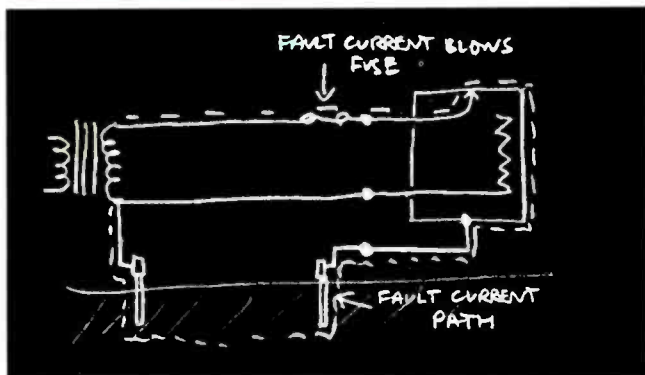
Fig.1a: A dangerous wiring fault

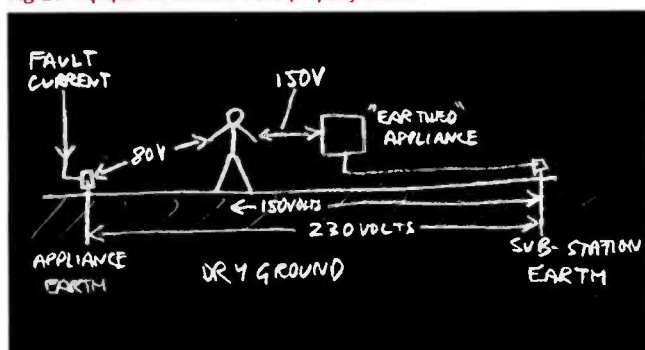Fig.1b: Equipment with its case properly earthed
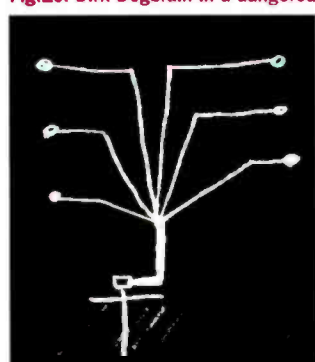

Fig.1c: Dirk Dogbrain in a dangerous spot!

Fig.2a: Radial earthing

Fig.2b: A device that suffers leakage of current
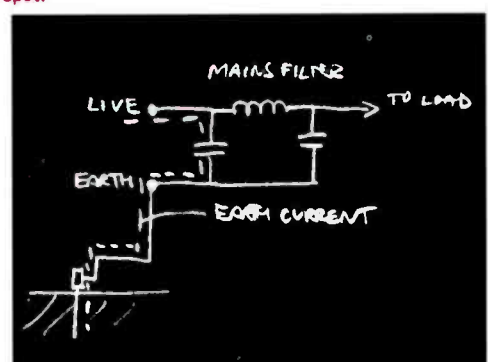

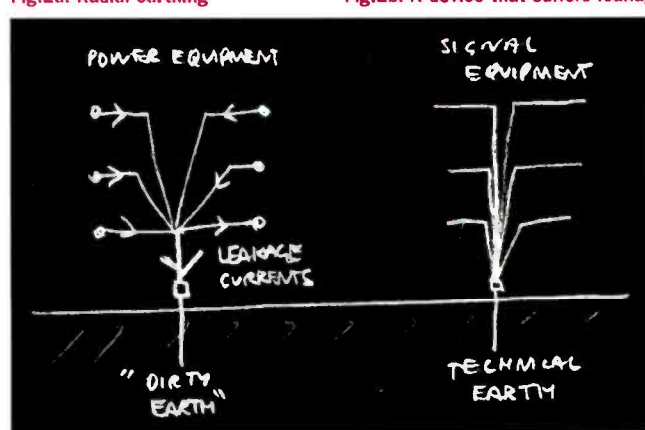Fig.2c: 'Technical ground' earths all signal level gear

Earthly delights

To the electrician earthing is a safety measure;
to an audio engineer poor earthing can lead to
shocking sound quality and poor reliability.
Continuing our tutorial series **John Watkinson**
presents a good grounding in the subject

ELECTRICITY in the wrong place is either a nuisance or a danger depending on the quantity. A few millivolts is not a danger, but causes equipment to hum and time is wasted tracking the problem down. A few hundred volts and your day could be permanently spoiled. A few kilovolts of static in the wrong place and you have bought your ticket to the big house.

The hazard to life caused by electricity is largely a function of the current through the region of the heart and the old trick of keeping one hand in your pocket is soundly based on this, as is wearing dry thick-soled shoes. The voltage is much less relevant. The electrician's phrase 'It's volts what jolts but mills what kills' is too true. Death has occurred on a supply of as low as 38V, but this is exceptional. The current is not just a function of the voltage, but depends on the source impedance and the conductivity of the skin in addition to how well the body is grounded. Standing in water is not very smart, hence the stricter regulations regarding electricity in bathrooms.

Many very high voltage supplies such as EHT in CRTs and the polarisers in electrostatic speakers cannot deliver enough current and are painful rather than lethal. Static build up in dry conditions can easily reach tens of kV, but the high source impedance and the limited charge prevents bodily harm. However the secondaries of the step-up transformers in electrostatics have fairly low output impedance and contact with these is definitely a bad idea.

For mains power delivery, only the live and neutral conductors are necessary. From the standpoint of safety, earthing to an additional conductor should prevent metal parts becoming live in the case of a fault. Fig.1a shows a

dangerous piece of equipment in which a faulty wire has touched the metal case which has become live. Anyone touching the case would complete a circuit to earth and receive a shock.

Fig.1b shows that if the case is properly earthed there is a return path for a fault current which will blow the fuse. The neutral conductor is connected to earth at the substation to complete the fault current path. The substation is normally the only point at which the neutral should be connected to earth. This prevents stray earth currents which could interfere with telephones. It also means that a small voltage difference between neutral and earth exists under load because of the resistance of the neutral conductor.

Clearly, if the fuse is in the neutral lead it will not blow and the fault current could be prolonged and cause a fire. Neutral fuses are banned not just because of this, but also because an open circuit neutral fuse would leave most of the equipment live, but not working.

Sometimes an earth conductor from the substation is provided in the form of a metal sheath around the buried cable. In rural areas with overhead wiring this is not practicable and local earth stakes are needed. In some areas such as sand or chalk the resistivity of the ground can be quite high especially in dry conditions. In this case a fault might not blow the fuse, but simply cause the earth stake to become live. As Fig.1c shows, a voltage gradient is set up around the stake and it is possible to get a shock by touching it, or by standing on 'live' ground and touching a genuine earth. This is particularly dangerous as it would be quite unexpected.

As 'earths' can become live, it is a requirement that all metal, water and gas pipes are bonded to earth, so that no potential difference can exist between them.

The solution in poorly conductive soil is to use an alternative system known as protective multiple earthing. In PME the neutral and ground building wiring is connected together and to a ground stake. As the

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1094A

1031A   1031A
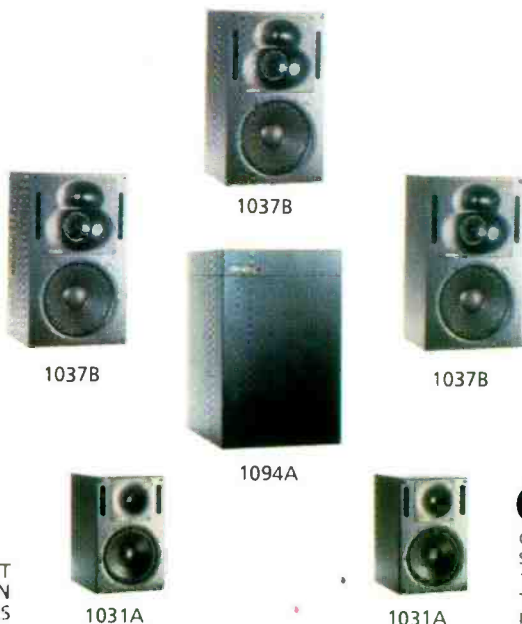
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Fig.2d: At the dirty outlet

Fig.2e: A signal cable connection produces a current loop
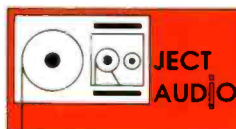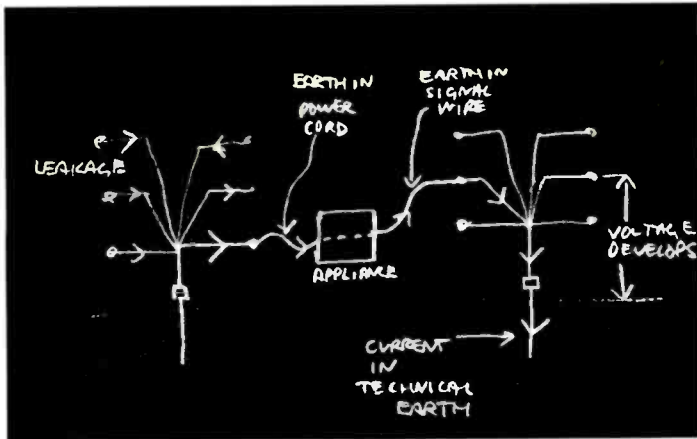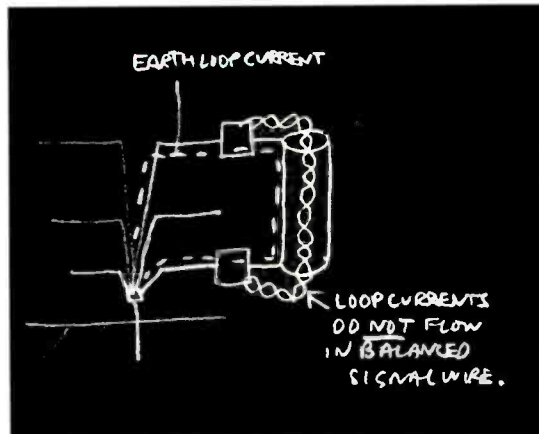
ground stake of every building on the same substation is effectively paralleled the overall resistance to ground is reduced.

Earth leakage trips are valuable where earthing is questionable because they sense any imbalance in the current leaving and returning. This is particularly useful in the case of electric tools such as mowers and saws which could cut their own cable and sever the earth conductor. In service bays the danger of shock can be reduced by running equipment under test from an isolating transformer with a floating secondary.

In audio and video installations earthing is often used to minimise interference picked up by signal wires. The slightly suspect theory holds that if a metal case or conduit is securely earthed its voltage cannot change and signals cannot be induced in the signal wire. Unfortunately this is only true if there is no current flowing in the metal. This can only be achieved if the earthing system is arranged as a perfect radial or star arrangement as shown in Fig.2a.

Fig.2b shows that if a device on the earthing system suffers from leakage or has capacitive AC input filters, current will flow into the ground system. Large motors and transformers often have appreciable leakage current. When current flows in the ground system, voltages are developed and it no longer acts as a signal ground, although safety is not compromised.

Fig.2c shows that one solution is the use of a separate 'technical ground' which is used to ground all signal level equipment, while the ordinary stuff like lights, photocopiers and air conditioners use a 'dirty ground' for safety.

There are a number of flaws in the technical ground approach. Fig.2d shows that Dirk Dogbrain has brought his DAT machine in and plugged the power cord into a dirty outlet. He then connects the audio output from the DAT machine to the input of a device on the technical earth which is now no longer a technical earth. In TV stations and recording studios with constant removal of old equipment and installation of new, the chances of the technical earth being free of such violations is almost nil.

Fig.2e shows that the connection of a signal cable between two units on the same earth system produces a current loop. Alternating magnetic fields from AC powered equipment produce loop currents which cause potential differences between different parts of the ground. In balanced systems the ground

currents do not flow in the signal lines and the common mode rejection eliminates the hum.

In unbalanced systems these add to the signal voltage causing hum and noise. Consequently consumer products are virtually unuseable in professional environments where loop currents are unavoidable. In fact, given the dynamic range of CDs it is questionable whether unbalanced audio is good enough for high-quality domestic installations.

One very sad fact is that there is no such thing as an RF ground, irrespective of the exotic materials which are sometimes suggested. Once the length of a ground lead reaches a significant part of the wavelength of an applied RF signal, potential differences appear along it. Even a straight bus bar has inductance and as frequency rises the impedance rises with it. Is that pipe a length of conduit or a waveguide?

Static electricity is not generally harmful to life, although if you work with explosives or fuel tankers you might not agree. Static can definitely harm electronic components and static awareness is becoming increasingly important as modern high-density logic chips are made with thinner tracks which cannot dissipate so much energy in case of a strike. Analogue equipment is less sensitive and valve equipment is virtually bomb proof.

Equipment is at its most sensitive when it is opened for maintenance, perhaps to put a new plug-in into a PC backplane. Fig.3a shows Dirk Dogbrain holding the new card. He took it out of its static protection in the office next door and walked across the synthetic carpet and charged himself and the card up to zillions of volts. He carefully lowers the card into the PC and discharges himself into the earthed PC. The current pulse flows via the new card and Dirk wonders why it doesn't work.

Fig.3b shows how to do it properly. Bring the card to the machine in its container. Ground yourself to the machine by touching metal. Repeat this every few seconds. While at the same potential as the machine, unpack the card. While holding the metal frame of the machine with one hand, insert the card with the other. Your body forms a safe slow discharge path for any static on the card, ensuring that it has the same potential as the machine. Properly equipped service bays have conductive worktops and technicians wear a wrist strap which connects to earth. The earth lead contains a few megOhms which stops the wearer lighting up if he touches anything live. ■
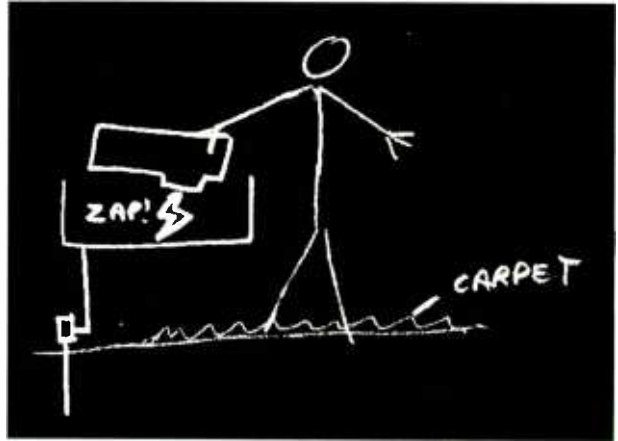

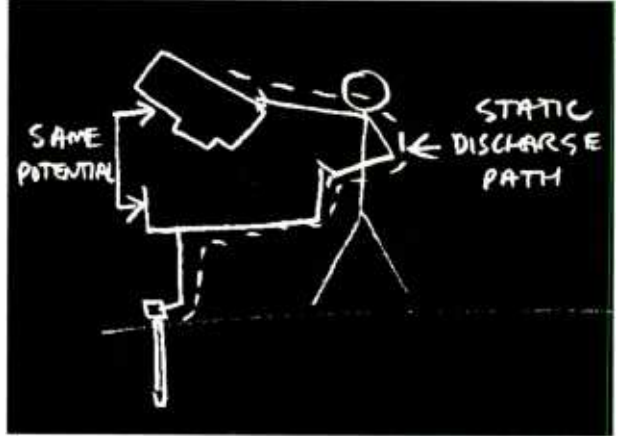Fig.3a: Dirk Dogbrain on the carpet

Fig.3b: Dirk Dogbrain prepares to discharge

Riding the wave

History is likely to cast the present problems with nonlinear digital audio file interchange as an embarrassing oversight —in the meantime, however, it remains a major stumbling block. **Tim Frost** investigates the FilmWav opportunity. It's time to chunk

THE INTRODUCTION of computer-based audio and video systems has readily lent itself to the proliferation of non-compatible storage formats. You only have to look to word processors, databases, and image files for proof: every developer introduces a specific method of storing the same type of information—because it can.

The situation is no more new than necessarily problematic—it is not a problem when files are confined to a single system, but it can be terminal when interchanging files between systems. And it becomes an important issue with workstations and recorders offering a range of storage systems that are incompatible both physically and as file formats.

Plans are now afoot to rationalise the situation for a scalable audio interchange format such that any and every workstation and non-linear recorder manufacturer will be able to understand in making interchange certain. Mark Yonge of SSL sums up the essence of what is at issue: 'It must be a format that is good enough to carry the "G" word—guaranteed interchange. Unless you can guarantee interchange with every possible type of workstation then you haven't reached the level we were at with ¼-inch tape. The test is to create an interchange format such that someone receiving it will be guaranteed to play it.'

The first step towards a workable nonlinear interchange standard was set in November when the EBU agreed on the Broadcast Wave Format. BWF goes some way to solve the more basic requirements for a standardised file. From the principles behind BWF is emerging a further option initially dubbed FilmWav,

but now more likely to emerge as the 'Film Chunk' of BWF. This will be more useful to the film and TV users as it attaches more useful tools and data to the audio file.

'We are looking at something that will start incredibly simple and go up to something that could be object oriented like OMF,' notes Yonge.

Nonlinear audio with its intimate links with video and film editing, requires a different set of interchange solutions to those used for linear recording of complete music tracks. In linear audio, DAT is a formalised standard for digital mono and stereo, and the DA-88 a *de facto* standard for 8-track—these both fit Yonge's 'G' word rule of interchange. Both can, and are, used to transport audio from workstations, but, being real-time formats, they hardly work in the spirit of the nonlinear production world. Workstations encompass video, they are not linear audio-only systems, and here lies the main problem. For a format to be interchangeable in the nonlinear audio world it must be playable on anything upwards from a home multimedia PC, which means absolute simplicity. But that absolute simplicity does not aid the production process further down the line if the file is an element of a film or TV production being created in a room full of Avid systems. Broadcasters need to be able to read the file and also keep track of its origins, the time-code references and check what processing has been applied to the file over time.

The film and TV industry goes further, really needing a file structure that supports full time code implementation, multiple audio channels

and EDL functions. The requirements of simplicity and sophistication could be seen as mutually incompatible, but the circle can be squared. The starting point is the IBM-Microsoft WAV file format—the most minimalist audio file format. There isn't an M-M PC or workstation in the land that can't handle a WAV file—even most Mac systems can handle a WAV file one way or another.

WAV is to computer audio what ASCII text is to word-processing—the one format that will deliver the message intact, even though all the frills are missing.

WAV files are reasonably open-ended as far as audio standards are concerned. It encompasses audio in the form of PCM recordings (any bit and sampling rate you choose) and MPEG audio for the broadcasters—already essential in radio and becoming more important in TV.

WAV meets the 'G' word rule in that it can be imported, played or generated on any computer-based recording system, no matter how

> **Nonlinear audio with its intimate links with video and film editing, requires a different set of interchange solutions to those used for linear recording of complete music tracks**

sophisticated or how lowly. But WAV falls down on what else it offers to the broadcaster or editor—which is virtually nothing.

SA&V's Ted Hayton, another player in BWF and Film chunk's development, notes that the WAV file contains more than audio: 'Everything is arranged in chunks of information. There are essentially two chunks, the audio chunk and a little fact chunk, that tells you the sample rate and so on. The beauty of the WAV file structure is that you can tag on other chunks of data to that file. If your system comes across some chunks that it doesn't need or doesn't understand, then it completely ignores it and just gets on with playing the audio.

'So a file can have the audio, the WAV fact chunk, a BWF chunk and a Film chunk and it will play on any machine that can read the basic WAV file.'

Using this freedom to add information to the fact chunk, the EBU committee has standardised a BWF chunk to add to the start of the file. Coming from the broadcast production world the additional information contains specifically broadcast origination data the sort of things that let the broadcaster track and identify the file.

'Until now, the only interchange format is DAT but WAV is a real blessing,' notes Aaton's Frederic Mangeat, another of BWF's pioneers. 'Even if you have no access to BWF chunks on your computer, you still have access to the data chunk so you can hear the sound. You may not have the additional information, but you do have usable audio.'

He also points out one of the

other key details that must be taken into consideration when developing a universal format—the need to make it easy for the smaller software and hardware developers to generate files in the chosen format, as well as read it.

'With Broadcast Wave it is very easy for semi-pro developers to add the broadcast

> As you go down the less professional systems, it also becomes less and less likely that developers will have the time, knowledge or inclination to add OMF

chunk. So even if you don't use all the facilities, you have time code so you can make notes. Now that is the beauty of this system and that is why we were happy to embark on the WAV rather than OMF.'

OMF users may be bristling at this point, but BWF is not in competition with OMF. Indeed WAV and BWF files can now be read under the OMF shell and building an OMF project using BWF files is relatively straightforward. But for a journalist preparing a mono report on a portable digital unit and phoning it over an ISDN line just minutes before transmission, OMF may seem a rather heavy-handed approach to file interchange.

As you go down the less professional systems, it also becomes less and less likely that developers will have the time, knowledge or inclination to add OMF. Admittedly there are OMF tool-boxes for both the PC and the Mac to make it more accessible to system developers. But the complexity of programming and the file overheads that are involved with a format that can cope with just about any

form of information, puts it out of the league for smaller systems and users with simple needs. 'OMF is a lot of overhead if you only want a simple job done and quite a lot of what we see out there requires just a simple job of work,' notes Yonge.

The BWF chunk was defined by an EBU committee focused on broadcast production and the information is mainly entered when the file is first created. BWF delivers information about who originated it and when, and gives the file a working name and an unambiguous title to uniquely identify it. The only updateable element is the coding history section. This can keep the user updated to what trans-coding has been applied to the file —important in radio where files maybe sent digitally down phone-lines CCITT-coded, stored and edited on MiniDisc ATRAC, and sent to the transmitter over an MPEG link.
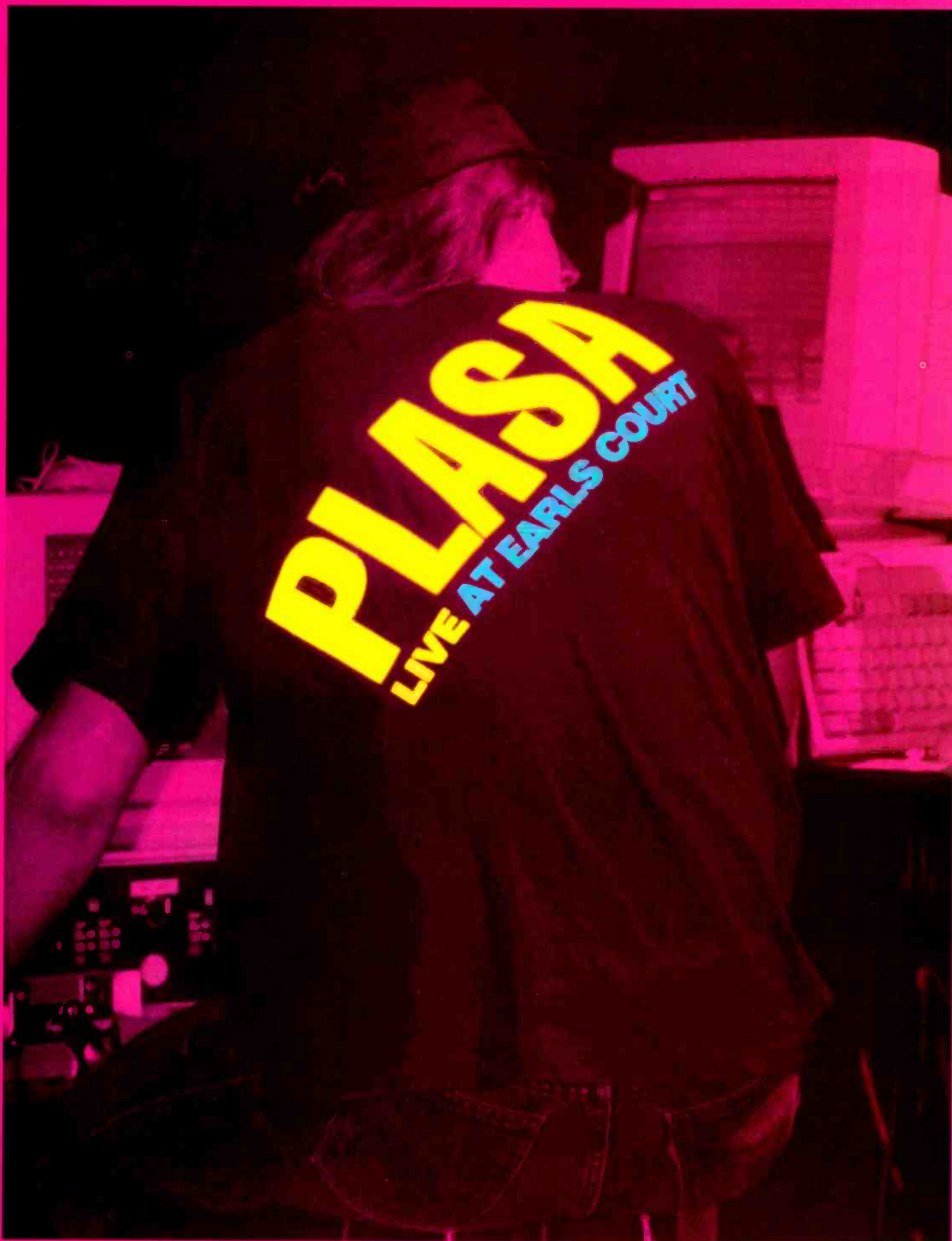
But even basic DAW-picture editing requires more than this—summed up as the EDL function that presents a range of start, stop and fade positions. offsets and track numbering.

The first option is to extend the data chunk in the WAV file to include specific source ID information necessary for the film postproduction—Keykode, location time code and other film specific data.

Then the really useful addition to the idea is to add an entirely separate file that contains EDL-type data. The workstation looks at the EDL file first which tells it which of a collection of BWF files are required, and how to order them within the project.

The edit points are referenced to sample numbers so that its accuracy matches the resolution of the audio file and can be constantly updated while leaving a trail of history behind it.

The main pressure is to get bald BWF accepted first as a real interchange format. When that is sorted, the EDL file system should develop in stages. The first proposal is to add 'simple' EDL file, the sort of information that would appear in scratch autoconforms at the end of the telecine

process. This includes source clip identity, tags to master time-lines and define simple cut points.

The second stage will be to attach most of the other EDL data—crossfades, pans, gains, and so on. This is the key to interchanging finished product for film, TV, radio and music. Also, since it locks all the references to individual audio samples, this form of EDL information becomes independent of specific time code frame-rates.

The logical extension is a move to an object-orientated format, although this may be considered a little more contentious,

> 'The audio industry is much more international than broadcast, as is the film industry, so through these routes we hope to tunnel into the TV world. We are certainly looking at linking with the AES'
> —John Emmett

stepping a fine line between BWF and OMF.

Some involved in the development of BWF view an object-oriented BWF sometime in the future taking the WAV file too far, making the format too complex, especially when OMF offers similar functionality and is already in place as a working 'standard'.

It is important to the EDL file instigators that they hand the idea over to an industry body before it starts suffering from NIH—the not-invented-here syndrome. As soon as one company proposes a standard, others refuse to have anything to do with it for various valid, and not so valid, reasons. The key to BWF's success was that, it was

handed over to the EBU committee before NIH cut in.

The EDL proposal, comes out of contributions, (some focused purely on simple BWF) by people at SA&V, SSL, Pyramix, Digigram, Cedar, Avid, AMS Neve, Lightworks and Aaton among others. Now a general development route has been outlined, it has been handed over to the AES working group that will steer it through to completion as a standard, taking input from anyone and everyone. 'What we need to look at is the anonymous approach where everyone with an interest can contribute, and not let it be considered as the proposal of any one company,' says Hayton, rapidly distancing himself from company-based activity now that the AES is going to further develop the format.

The trouble is when just one manufacturer puts it forward it looks just like a bit of marketing and then it doesn't take off. No-one wants to Shanghai the Film chunk, so now there is some momentum we are handing it over to the AES standards committee and let them drive it forward.' The end result should be a continuum of file extensions. The core part of the file—the audio—can be created and read by any, and every, nonlinear system, while the fact chunks will give increasing amounts of functionality and tightly focus on equipment or activity.

Submitting the plans to the AES is a final interesting move, as it should enable the standard to bridge the Atlantic. BWF is an EBU, therefore a European, standard, but according to John Emmett who represents the UK Independent Broadcasters on the EBU committee there are plans to co-ordinate the work by the EBU and AES on this audio format.

'The audio industry is much more international than broadcast, as is the film industry, so through these routes we hope to tunnel into the TV world. We are certainly looking at linking with the AES,' he says. 'We expect to have some framework in place that will have BWF at the bottom going up to OMF at the top.' ■

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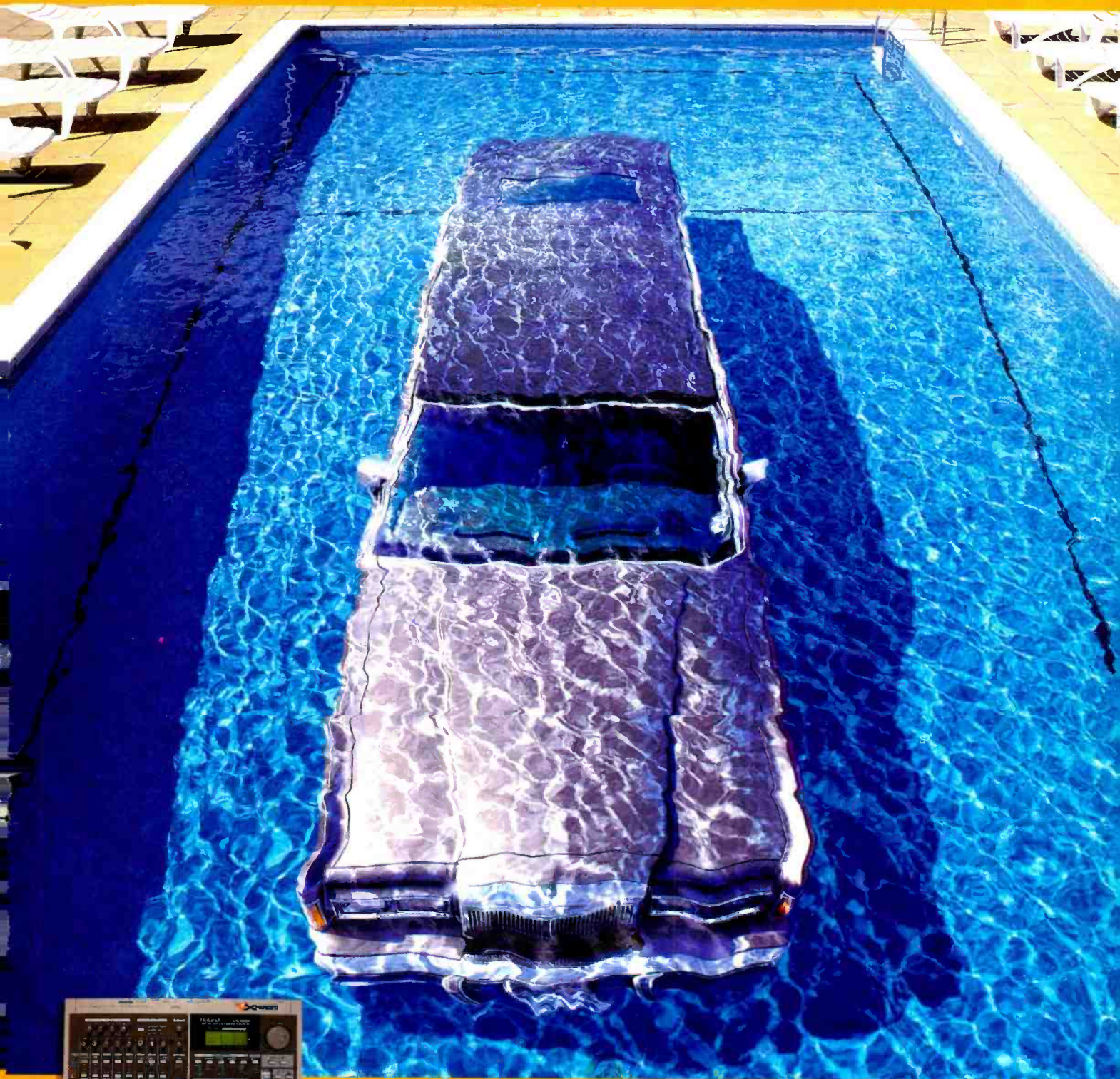
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Don't blame the machine

Studio Sound's critical listening test challenged the folklore surrounding the performance of MDMs. **Philip Newell** highlights one area that defeats many of their users

I greatly enjoyed reading the 'All Ears' article in *Studio Sound* (April 1997). In an industry where fact and myth are now so blurred, carefully constructed presentations such as the above are needed, perhaps more than ever. That issue of the magazine will give me a useful weapon (when rolled up) with which to hit both my clients, and their clients alike, when they spout off, as they all too frequently do, about how DA-88s are vastly superior in sound to ADATs, or vice-versa, or that they are both rubbish when compared to a Sony 3348, or whatever else is in vogue in their own part of the world.

The results of the test really didn't surprise me as I have used most of the systems myself, and have never found them to be sufficiently different in their sonic character for me to unduly worry about. A few years ago, I made a live recording for CD and TV using a pair of ADATs, the mic amp section of a Mackie 1604 (how Mackie achieve the sound quality and output capability for the price, still astounds me), and a set of Schoeps microphones. The subject matter of the recordings was a beautiful pair of Steinway concert grand pianos, and two excellent pianists. I am still very pleased with the sound of the end result and from what I know of the 3324A and 3324S, I would say that there was no lack of quality, *per se*, from the ADATs.

So why are there so many people going around saying that certain digital recording systems are poor—the type of comment that is typically rife in the project-studio end of the industry?

I believe that almost all of these nonsensical comments are because they are blaming the recorders for the sound of the entire audio chain. The reality is that very few people make recordings on ADATs (relative to the size of the ADAT market, that is) using sets of $2,000 microphones, FM Acoustics Class-Amp mic amplifiers, and listen through top quality monitor systems. The truth is that the more professionally-orientated machines (the appearance of the word professional on the front of the machine often indicates that it isn't) are more frequently used with better quality recording chains than the project studio-orientated varieties. As with the old computer adage, rubbish in

equals rubbish out.

Two other aspects of the project world also can blight the less expensive equipment. Firstly, people of lesser levels of training and-

> They are blaming the recorders for the sound of the entire audio chain. The reality is that very few people make recordings on ADATs using sets of $2,000 microphones, FM Acoustics Class-Amp mic amplifiers, and listen through top quality monitor systems

or education about the recording process often tend to believe that balanced, +4dBm inputs and outputs are 'professional' so must sound better than the 'domestic' -10dBv inputs and outputs.

This is not necessarily the case. High quality, sonically neutral, balanced, high-level input and output stages are not cheap things to make. When every possible cent is being cut from the manufacturing price of a piece of project studio equipment, placing too much confidence in the superiority of the high-level connections can be foolhardy. In many cases, the simpler, unbalanced, -10dBv connections are of greater sonic purity.

Secondly—and this point was made a couple of years ago in an article which I wrote with John Watkinson—you will only get optimal quality from a digital recording system if the peak level is at a maximum, just before the onset of 'distortion'. To achieve this via the +4dBv inputs of an ADAT, which registers -18dBv for a +4dBv input, means that a huge +22dBv needs to be sent from (and will return to) the mixing console, or mic amps. This may well be okay if you're using your ADAT with a Neve, SSL, Focusrite, or similar console, but many mixing desks aimed at the project market are beginning to strain their sonic purity (if they ever had any) when working at these levels.

For the above reasons, if I must make recording with ADATs, or similar systems, I invariably use the -10dBv inputs and outputs of the mixing console. This way, the signal enters and leaves the machine via simpler circuitry, and +4dB from the desk will provide a recording level about 4dB below overload. This is certainly healthier for the analogue circuitry of most mixing desks. On playback, the desk metering will also read sensible levels. Why so many manufactures of digital machines do not fit input and output level controls still puzzles me. The machines could then be used as unity gain devices, operating at their optimum digital level, with whatever line level was chosen, and so many of the recording nonsenses could thus be avoided.

Anyhow, the upshot of this is not to blame the machine for the poor quality of sound, which is probably due to associated equipment and-or uneducated interfacing. Perhaps, however, some of the manufacturers could be blamed for leading so many of the unenlightened down some distinctly slippery slopes. ■

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