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Studio sound

THE INTERNATIONAL TECHNICAL MAGAZINE FOR PRO AUDIO, POSTPRODUCTION & BROADCAST

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Post needs to talk

MANKIND TOOK A MAJOR EVOLUTIONARY STEP when it got its head around the concept of currency and money. Freed from the need to take a herd of cattle or daughters along whenever he wanted to do a little shopping, early man soon enjoyed the benefits of dealing with something that could be exchanged directly for something else, that had its own integrity and worth, and that was also strangely portable and convenient. He may have missed the adrenaline rush of the hard barter or swop, but he replaced this with the thrill of the fast cash deal.

The studio community took a major evolutionary step when it got its head around the concept of transferable and interchangeable media. The adoption of 2-inch tape meant an artist could take the reels and travel comfortable in the knowledge that the next studio or mixing room could deal with the stuff. It's a basic approach but it works and is based on the principle of a standard that has been (largely) preserved through the adoption of digital tape-based media.

Things have never been this straightforward in post since the arrival of digital. Despite the now almost universal use of removable media the chances are you'll break something if you remove a drive from one DAW and stuff it in the slot of another. Small clusters of manufacturers have got together to ensure degrees of exchange in this environment and there have even been moves, most notably OMF, to instigate industry-wide interchange. However, most will have noticed the less than complete endorsement of this, and the selective implementation where it has occurred. On the whole, manufacturers remain keen to tie up their users into their own systems.

Clearly this is now as ridiculous as it is outdated. The post community and the manufacturers need to start talking seriously if the situation is not to fragment still further than it already has.

Zenon Schoepe, executive editor

Passion play

IT'S DIFFICULT TO GAUGE PASSION. Until recently, the old line about turning up the 'brilliance' control on the guitarist's amp to elicit the required performance from him was about as scientific as it got. Then came the recession. Here, suddenly, and most unwelcome, was a means by which the passion of studio owners could be reconciled with their business sense. Sadly, those who scored most highly on the pure passion scale were the ones most quickly out of business.

So what is it about music and technology that allows it such command of our feelings? That music it an emotive entity is not in itself the answer to question, since it often plays such a small part in the larger studio scheme of things. There has to be something more. Something that is powerful enough to determine our lifestyles, our 'business' ventures, our spending habits, our conversation and our dreams. Neither is it simply the 'anorak' factor—although it's certainly at work, I don't think there is enough in either the theory or the practice of analogue or digital electronics to cast as powerful a spell as the one we're all apparently under.

I've been pondering this question for some time now, and I don't feel any closer to finding an answer than when it first presented itself to me. Somewhere along the way, however, I realised something, that's not more important, but is certainly more immediately consequential. Just as unravelling the real-world workings of an elegant 'magic' trick can destroy its attraction, the strict imposition of real-world values on the art of recording is certain to leave its mark. Although we have survived the recession with much of our old values intact, much that once was possible is no longer so—denied us by our accountants or our revived fear of failure.

Passion is one of our most essential muses. We must guard it with our lives.

Tim Goodyer, editor

Studio Sound
Incorporating Broadcast Engineering
June 1997 Vol 39. No 6. ISSN 0144 5944

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Total average net circulation of **19,500** copies July 1995–June 1996 **(UK: 6,565. Overseas: 12,935).** (ABC audited)

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Fairlight announces exchange initiative

WORLDWIDE: DAW manufacturer Fairlight has announced co-operative agreements with Lightworks, Studio Audio and Video, TimeLine and Doremi Labs to create a Direct File Exchange capability beyond the scope of OMF. The company's intention is to create a consensus between manufacturers for interchanging file formats and EDLs by a translation process built into the DAW software.

Fairlight is keen to disclose all information required to enable the playback of its files to any manufacturer it enters into an agreement with according to Fairlight director of European operations Nick Cook. 'We have positive indications that several other major manufacturers are interested in joining us in establishing a standards forum to explore the possibility of a common file implementation,' he said.

'Cross platform connectivity and exchange are vital for our industry,' added Doremi Labs' Camille Rizko. 'Customers like to pick the system they feel most comfortable working with. They, however, do not like to be locked into a closed system.'

Fairlight, Europe.
Tel: +44 171 267 3323
USA. Tel: +1 310 287 1400

WORLDWIDE: U2 has bought six channels of Sony wireless mics for its PopMart world tour which was launched in Las Vegas. Including six WRT-860 belt pack transmitters, three WRR-850 diversity receivers and a variety of guitar leads, the wireless mics are used on the guitars of Adam Clayton and The Edge. Bass guitar technician Stuart Morgan said he was drawn to the Sony system's ability to use 64 different frequencies. The PopMart tour will take in more than 40 stadiums in the US on its first leg and is likely to top 100 shows on six continents in the next 14 months.

Europe: Plus XXX studios in Paris has completed the refit of its two main rooms following the installation of an SSL 9000j in Studio 2 last year and the augmentation of its Genelec monitoring systems with stereo subwoofers. Studio 1, with its Neve VR, now runs dual 18-inch subs underneath its triple fronted 1035As in an unprecedented move that is surprisingly civilised given the phenomenal power delivery possible. 'We still like the Genelecs but we've improved them because we wanted extra bass efficiency according to the wide variety of music and film scoring we handle here,' said Plus XXX owner Claude Sahakian.
The smaller Studio 2 employs a similar, although slightly scaled-down, arrangement and Sahakian's objective to extend low end but not to over cook it was placed in the hands of acoustic designer Christian Malcurt. 'The sub bass is not only for the volume, but it's also for the frequency response,' said Malcurt. 'It's important that the room works well at low and high levels.'
Plus XXX, France. Tel: +33 1 42 022102. E-mail: plus30@calva.net

UK: *Beauty and the Beast* which opened recently at the Dominion Theatre, London features two Cadac consoles—a 53-input F-Type with 17 specially designed stereo input modules and a 58-input J-Type. Costing more than £10 million and billed as the capital's most expensive musical ever, the show has a cast of 40 and a 25-piece band. Sound design is by Richard Sharratt with a system from Autograph Sound Recording. The desk's stereo inputs were designed specifically for the production and feature twin input meters, mic inputs and F-Type stereo EQ.

UK: Hard-core dance production house STD has installed a Tascam M1600 console in its studio at 24 Karat Records in London. Made up of the keyboard playing and remixing duo of Tomye Durkin and Ivan Black, the outfit has a *Doctor Who* connection through programmer, arranger and engineer Alistair Lock who has composed soundtracks for *Doctor Who* and *Blake's 7* sci-fi appreciation videos.

IUTC supports HD-CIF

WORLDWIDE: The Inter-Union Technical Committee of the world's eight broadcasting unions has strongly supported the adoption of a unique standard for programme production and exchange of high definition TV following a meeting in Ireland in April. This, it claims, will lead to easier and better exchange of HDTV programmes, lower equipment costs and will accelerate the move to high-definition television.

The IUTC recommends that the new standard should be called HD-CIF (Common Image Format uses the same image sampling matrix irrespective of the field/frame rate) which has a 1080 line by 1920 sample x 50Hz/60Hz scanning system. Manufacturers are being encouraged to provide products to this standard and the Committee will invite them to explain their plans and timescales for making available dual field rate (50/60) HD-CIF equipment.

IUTC. Tel:+6 03 282 3108.

HHB in Toronto

CANADA: HHB Communications has opened a distribution company in Canada based in Toronto with technical and warehouse facilities. HHB Communications Canada is headed by Dave Dysart, formerly of Studer Canada, and will handle the company's entire international product lines including the Genex GX8000 MO recorder which Dysart believes holds much promise in the territory.

HHB Communications MD Ian Jones did not consider the move to be an expansion of the US market as he believes the country has its own identity. 'Canada has a thriving film industry, its music recording facilities rival anything available in the US and its broadcast companies produce programmes of the very highest standard,' he said.

HHB Communications, 260 King Street East, Toronto, Ontario M5A 4LF, Canada. Tel: +1 416 867 9000. Fax: +1 416 867 1080. Email: hhbcan@istar.ca

Sound recordist Martin Wilson is using his Fostex PD-4 portable time-coded DAT machine on the dramatised documentary series *War and Civilisation* for Worldview Pictures with shooting in Turkey, China and Mongolia. 'Despite the dust of central Turkey and the constant joltings in transit, the PD-4 performed flawlessly throughout,' said Wilson who has also used the machine for the location shooting of the *Harry Enfield and Chums* Tiger Aspect/BBC comedy which featured the 'Il Postino Pat' sketch shown—a spoof of the children's Postman Pat series.

■ **Atlanta,** Georgia and Havana, Cuba have seen key placements of Amek's Galileo console. Synchronised Studio B is running the first US Galileo—a 32-input console equipped with 7.1 surround mixing that will handle the facility's larger postpro projects. The Cuban state record company, EGREM, will install a 56-input Galileo for its opening this month. The large Eastlake-designed facility will mainly host tradition salsa recordings requiring the accommodation if large numbers of musicians.
Synchronised Sound, US.
Tel: +1 404 873 4477
Eastlake Audio, UK.
Tel: +44 171 262 3189
Amek, US. Tel: +1 615 662 8939

■ **Spanish** production and post house, Telson Estudio, has ordered two Fairlight MFX3 workstations to pair with two Yamaha 02R digital consoles. The new systems will be used extensively for dialogue replacement in productions for TVE, the Spanish public television company and follows the installation of similar systems at other leading Spanish post facilities.
Fairlight, UK. Tel: +44 171 267 3323
Yamaha-Kemble Music, UK.
Tel: +44 1908 366700

■ **South-east Asia's** biggest post house, Soundfirm In Australia, has installed a Bag End subwoofer system in Theatre 1 of its Melbourne facility. Soundfirm is well established in the international movie post circuit with recent credits including *Romeo and Juliet* and *Shine*.
Soundfirm, Australia.
Tel: +61 613 9645 4522
Bag End, US. Tel: +1 847 382 4550

■ **The Mexican** leg of Enrique Iglasis' American tour finds Clair Brothers mix engineer Dave Morgan surrounded by eight channels of Summit DCL-200 valve comp-limiting from Clair's resource of some 46 Summit units. Further sound hire activity is supplied by Californian-based ATK Audiotek which has recently purchased a further 20 XTA DP200 digital EQ-processors. Recent work for ATK included the Atlanta Olympics.
Summit Audio, US.
Tel: +1 408 464 2448
XTA Electronics, UK.
Tel: +44 1299 879977

■ **London's** Town House has ordered a 72-channel SSL 9000j-series console for Studio 2 to accompany the 9000j already installed at Olympic. The studio is recently undergoing a redesign by Sam Toyashima and will reopen at the end of this month.
The Town House, UK.
Tel: +44 181 932 3200
SSL, UK. Tel: +44 1865 842300

■ **Japan's** hosting of the 1998 Winter Olympics finds Graham Patten Systems proving 52 D-ESAM digital edit suite audio mixers to Panasonic for the international broadcast. The order comprises 820, 400 and 230 D-ESAM models and continues Graham Patten's regular involvement in the Olympics which dates back to 1984.
Graham Patten Systems, US.
Tel: +1 916 273 8412

■ **London's** CTS studios has opened a new digital mastering suite to provide greater production flexibility and reduced turnaround times. Central to the new room will be an SA&V SADiE editing system which can be used in conjunction with the facility's recently installed Pacific Microsonics HDCD processing system—currently the only such system in London. The move increases the number of CTS' mastering suites to three.
CTS, UK. Tel: +44 181 903 4611
Studio Audio & Video, UK.
Tel: +44 1353 648888
Pacific Microsonics, US.
Tel: +1 510 644 2442

■ **Chinese** radio stations, China Radio International and China National Radio, have purchased AMS Neve Capricorn digital consoles. CRI has installed a 48-fader Capricorn for use in high-quality music and drama programming for sale to Chinese-language broadcaster abroad as well as for national transmission. CNR is set to commission a new 48-fader Capricorn imminently, also for quality music and drama broadcast production. In nearby Thailand, the new Fatima Studio setup has commissioned a 48-fader AMS Neve Libra console.
AMS Neve, UK.
Tel: +44 1282 417282

■ **French** studio owner Dan McEnroe has purchased a P&G PP10 audio multiprocessor system in his Paris facility. The PP10 has the Studio Suite from the Pythagoras range installed and will be used for audio restoration work as well as recording and mixing acoustically-orientated music.
Penny & Giles, UK.
Tel: +44 1495 202024

■ **London's** Metropolis studio complex is the home of Planet Audio, a music programming and production suite which has recently installed two Yamaha 02R consoles to run with its three ADATs and 16-track Pro Tools system. The digital desks replace an analogue DDA console where they accompany a selection of high-end outboard including units from Tube-Tech, Helios, GML and Dolby, along with Apogee AD1000 A-D conversion.
Metropolis, UK.
Tel: +44 181 742 4444
Yamaha-Kemble Music, UK.
Tel: +44 1908 366700

■ **Hollywood** post facility, Complete Post, has installed ten pairs of KRK M9000B monitor speakers. The postproduction fraternity's acceptance of KRKs is further underlined by installations of M9000Bs in California at Musicworks and KRoKs at Universal Studios. Seattle's post people have also been installing KRKs, notably at Clatter & Din, Media Partners and Slice Editorial which is entirely operated by female staff.
KRK Monitoring Systems, US.
Tel: +1 714 841 1600

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Orban Audicy

Producing a workstation that intelligently interprets the demands of real working environments has proven remarkably elusive to some manufacturers. Rob James reckons Orban has it

ORBAN'S LATEST workstation the Audicy, is a RAM-based machine designed to be extremely simple to operate and as bulletproof as possible. It is aimed primarily at producing completed productions in the short form radio and TV arena, typically at commercial and promo production. As the maximum record time increases it will find applications on longer projects.

Physically, Audicy consists of a large tower PC case containing... a PC, with an Adaptec SCSI adaptor and a number of proprietary cards, a dedicated hardware control surface with faders, jog wheel, buttons and a pull-out drawer containing a mini PC keyboard used for naming items, and a 17-inch SVGA monitor. There is an optional smart, black designer stand on wheels if the application needs a stand-alone solution.

The keys are chunky in appearance, but some feel rather flimsy and a couple kept bouncing off to hide in the dark recesses beneath my desk, but Orban assures me the production machines will not have this problem.

The digital inputs and outputs all have automatic sample-rate conversion with a wide capture range and good tolerance to less than perfect signals. Internally, the sample rate can be set on a per-project basis to either 44.1kHz or 32kHz.

There is onboard DSP that provides equalisation, reverb and time-domain effects plus two Lexicon reverb algorithms.

System options include Data DAT or Jaz drives for backup, and a network card and software to facilitate uploading 'electronic carts' to a central audio delivery system. Currently the Enco DAD and Broadcast Electronics Audio Vault systems are supported. Under development are links to the Orban Airtime system.

On the front panel the main tower has AES-EBU, SPDIF and analogue I-O for left and right, and outputs from the left and right aux buses. There are also BNCs for wordclock in and out, and video sync in. On the rear there are connections for the control surface, time code, external SCSI devices and the monitor. The tower has the usual noisy cooling fan.

The dedicated hardware controller is open and uncluttered. It is loosely divided into three areas—mixing, editing and transport controls—with menu navigation controls close by the jog wheel. The mixer section has a total of 13 faders.

The jog wheel is chunky with a good feel, and there are three 'gear ratios' to choose from when scrubbing.

There are 24 'tracks' of which any 10 can be played or edited at any one time.

When you switch Audicy on, it first displays the usual PC BIOS boot screen with the customary device loading and checking followed swiftly by the Audicy 'welcome' screen. From this point onwards the PC platform recedes into the background, and the environment is entirely Audicy.

You are initially presented with four pull-down menus which enable you to manage Productions, the Library and System Utilities. The fourth pull-down is the Information Centre that gives an overview of the options fitted to the system and options to check disk and DAT storage. The user-interface is clear and uncluttered both on screen and on the hardware controller. On-screen help appears at virtually every key press. When familiarity increases this can be reduced to alerts or warnings. There is also a HELP key which can be invoked for more in-depth assistance.

The menu system operates in the same way in all screens. The blue cursor buttons above the jog wheel allow you to navigate through the menus and select options which are confirmed with the ENTER button, or rejected with the ESC (escape) button.

If you're in a tearing hurry, selecting Make New, hitting RETURN and then Shift RETURN will get you straight into a new blank Production with the global defaults on sample and frame rate. A better option is to go through the menu choices and name the Production or you can end up with a disk full of Productions called <Untitled>. You can rename Productions later, but it is good discipline to name them from the start. A Production can either be 'shadowed' in which case all audio and edit information is recorded to disk as well as RAM, or 'temporary' which only records to RAM. You get the option to convert a temporary Production to be shadowed either during work or when quitting. The machine saves edit information to disk every 3s so in the event of power failure or similar disaster you loose very little work.

Once you have opened a Production, you're presented with the first mixer screen which gives a graphic representation of the mixing section controls at the top. A bar across the centre of the screen displays icons which indicate the status of various parameters and controls—input type, splice type, output muted, time count, frame rate, what the parameter knobs are currently controlling and whether the machine is locked to time code. Below this is a small track display followed by meters in the bottom left-hand corner, menu options in the middle, and time displays on the right.

Recording audio is simply a matter of turning on the input channels with the buttons above the faders, setting a level, selecting a record channel or channels using the Record buttons above the faders and pressing RECORD and PLAY on the transport controls. Recordings can be mono or stereo, pressing any two of the record arming buttons turns those channels into a stereo pair, so you can have a stereo recording on Tracks 1 and 24, but I wouldn't recommend it because life can get far too confusing.

Record time is limited by available RAM—if there is 64Mb of RAM you get 12m 32s of mono, 6m 16s of stereo at 44.1kHz sampling rate. This is the total amount of audio allowed per Production not per recording. RAM can be expanded up to 256Mb giving a maximum of around 50 track minutes at 44.1kHz.

The transport controls are familiar tape recorder stuff with some additions—Cue modifies the action of the FF, RW and PLAY buttons. In addition to working as normal playing audio with smooth ballistics, FF and RW can move in 20s increments, or skip silence and go to the head of the next segment. The Locate group of buttons has keys that locate the start or end of a segment, the head of the last recording, the head or tail of the Production next or previous locator position and autolocate locations 1 and 2. There are a total of 24 autolocate points accessed by pressing the GOTO key. Locations are incrementally set by hitting SET. Indicators appear in the time line display with the locate point number.

To edit recordings you hit the EDITOR button which—unsurprisingly—changes to the Editor screen. This changes the top section of the screen to a large horizontally scrolling track display with central 'play head'. The smaller track display scales to keep the entire production in view in an overall time line.

Tracks to be edited are selected with the PLAY buttons above the faders. If you are editing stereo recordings, it is necessary to select both tracks. In video-editor-style you select a Source In point for the start of the section you want to remove and a Source Out point for the end. If you are copying or moving audio, you also need to select either a Destination In or Out point and select the same number of tracks for record as you have source tracks. Source and destination AUDIT buttons take you to the In points, Shift AUDIT to the end points, or if AUDIT is pressed twice, the relevant section is played. Edit mode is selected from the pull-down menus. Options available are Leader, which inserts silence, Cut or Erase; Copy In, Over or Self; Move In or Over; and Loop In, Over or Self. In inserts audio at the destination point; Over inserts audio over whatever is already recorded; Self and Loop are interesting—Self performs the selected operation to the same destination tracks as source tracks while Loop allows up to eight copies of the source material to be placed between Destination In and Out points in one operation.

Splice options are described in ¼-inch tape terms with a choice of 90° butt, 45°, ½-inch or 2-inch fade (15ips). These options are obtained by repeatedly pressing the SPLICE button.

In addition to these normal edit functions, there are time-domain tricks on offer under the Studio menu. Time compression and expansion can be performed up to a maximum of 25% with the Time-Fit options. These can be set by specifying a slot to fill or by setting the percentage. The algorithm can be optimised for 'splice quality' or 'maintaining even tempo'. On voice at 10% the >page 12

< page 10 quality is very good if 'splice quality' is selected, even 25% would be acceptable in some circumstances. The pitch and varispeed options operate in similar fashion. There is also a Flip option which allows a segment of audio to be reversed.

The Studio menu also allows access to the Library functions. Sounds can be copied to the library for use in other Productions or library sounds can be Dubbed into the current Production. The library is presented in the form of a single list that can be put in order according to various criteria.

The mixer key takes you back to the mixer section which allows you to mix down a Production to a pair of tracks. This is achieved via the Bounce function which takes the main output post any effects and rerecords the result. An AB button allows switching between groups of channel strips selected for Play which makes Punch In recording simpler. Pressing MIXER again gets you into the expanded Mixer screen. The Aux and Submix outputs can be set pre or post fader. Each channel has aux send pan and gain, submix send pan and gain, and the main pan control. The PARAMETER knob controls each of these with the specific function selected by means of the PAGE UP and PAGE DOWN keys above the jog wheel. Pre and Post selections are made from the Mix Toggle menu.

Effects can be inserted in the channel, the submix or the main output, and where effects are inserted into channels, channel pairs can be linked for stereo. The standard DSP supplied gives 12 'units' of processing. A mono equaliser or compressor uses one unit while Orban has provided its own reverb algorithm that uses three units and is adequate for most purposes. If you need quality reverb, there are two additional Lexicon algorithms of the usual standard that use six units each. An additional DSP card can be fitted that considerably increases the available DSP. An obvious way of husbanding the DSP resource is to use reverbs in the subgroup output rather than individual tracks or alternatively to bounce tracks with processing applied.

When an effect, say an equaliser, is inserted you can choose from a selection of preset settings which may be all you need to solve the problem, or, if you cursor down, the display changes to a graphic representation of a rackmount processor. This allows you to modify the chosen preset. Numbers next to the virtual knobs indicate which channel's parameter knob to use to vary the desired function.

The Patchbay menu allows mapping of any 10 of the 24 virtual tracks to the 10 mixer channels or edit tracks. Tracks can be named or phase inverted and patches can be saved and recalled including effects assignments. The SMPTE time-code setup is also on this menu. A nice touch is that a Production done in one time-code format can be converted to another. At present there is no RS422 (Sony P2 Protocol) control, but the Audicy will chase time code or generate it. It will also record against external time code.

The system appears to fulfil its brief and be thoroughly bullet-proof. I didn't manage to crash it during the test period despite the usual incompetence and attempting illegal acts. In fact, Orban tells me that, of late, they have offered their R&D staff a cash reward if they can make it crash—it remains unclaimed.

As a workstation this has one of the shallowest learning curves I have ever come across. Anyone who understands what a DAW is for, and has at least a nodding acquaintance with the concepts should be able to sit down at an Audicy and perform useful work immediately. I would say this workstation manages the difficult trick of allowing users to operate it at the level appropriate to them. An occasional user can put together a simple production without the technology getting in the way. For more serious users Audicy offers a choice, if you need 'quick and dirty' then the workstation lets you do that, if you want to do a polished, complex production you can do that as well. The Intelligent Digital Interface removes all the bother of worrying about what sampling rate raw material on DAT was recorded at, or fiddling about getting CD music and effects converted when the rest of the source material is 48kHz. After a while you discover practically everything you might want to do in normal recording editing and mixing can be done using the keys on the controller—even the pull-down menus usually have alternatives on keys. In almost all cases there is no necessity to type in numbers. It is usually quicker and more intuitive to use the jog wheel. The neat little QWERTY keyboard really is only used for naming things.

The really clever stuff—which will have some specifiers salivating—is in the networking. There is just a hint of what might eventually be possible. The Holy Grail for major broadcasters is a complete networked system for acquisition, programme production and transmission. External feeds coming in to a central server with workstations taking material appropriate to them individually or simultaneously, editing and packaging it, and returning completed items via the network for transmission. Audicy can already be configured to do the upload bit to the systems mentioned earlier. It will be very interesting to see how much further Orban takes the ideas and how soon. ■

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AMS Neve AudioFile Prolog

The oldest digital audio workstation around is still one of the postproduction industry's most widely used and best respected. **Rob James** catches up with its latest incarnation

I AM REMINDED of a wonderful old British sitcom by the singularly inapt name AMS Neve has chosen for its new AudioFile. I am sure the marketing boys will justify the name because the machine is intended —among other things—to introduce new users to the wonderful world of AudioFile. But how marvellous it is that the company with the best claim to the invention of the digital audio workstation concept should now come up with the name Prolog.

For those of you who remember *Up Pompei*, the whole thing was anchored around Frankie Howerd telling us a story beginning with 'The prologue'. Part of the joke was that at the end of each episode, he had never managed to finish. But the audience had been magnificently entertained by the chance interruptions and tangents along the way which add up to a story in its own right.

Main menu


Recording


Cut and splice

Prolog comes as three 19-inch rackmounting hardware units, although the Spectra control surface will be more commonly used, with its legs attached, as a desktop unit. The largest box is the mainframe; at 8U and 40kg, this is a serious piece of kit and contains the majority of the processing, the mass storage and a plethora of interfaces (some of these have no function in Prolog, but are there to facilitate future expansion). All the audio, sync and machine control connections are made to this box.

A 'tranlink' cable connects the 2U-high Spectra interface to the mainframe. The cable can be up to 60m long which makes for very flexible siting. The Spectra interface has a diskette drive for loading and saving event lists, EDLs and software updates. There are connectors for GPIs, printer, remote monitor (VGA) video editor, PC keyboard and key remotes. The Interface connects to the control surface via a D-connector and 4 BNCs. The control surface has a colour TFT screen, to the right and underneath of which are softkey Trigger keys. These all control a variety of functions depending on which screen you are using.

Transport controls are bottom right with two jog wheels and associated nudge buttons above. Moving upwards there is a small status display then the numeric keypad and a group of six dedicated Function keys. At the top is a small QWERTY keyboard used for naming things. Time displays can be in seconds, time code or film footages.

To understand Prolog you really need to know a little of the history of AudioFile.

When I first saw a prototype in the early 1980s, there was no culture of digital nonlinear recording. The sampler had been with us a while and was beginning to find applications in areas other than music—for looping atmospheres and playing sound effects, for example, but no-one really knew what to do with the thing. Virtually everybody could see possibilities and applications if only it could do... Here you could substitute wish lists from people with considerable experience in a plethora of audio disciplines. The research done by AMS resulted in certain features which may appear strange to those brought up on the current generation of DAW. The Trigger keys are a case in point—these are a legacy from one of the earliest applications to be identified, replacing tape cartridge machines for spot effects. They are still there partly because the original application is still valid, but also because they have had other uses superimposed on them.

If you compare the 'green-eyed monster' (original AudioFiles had 9-inch green screens) with a current Spectra you will find marked similarities.

AMS Neve seems to believe changing the look and feel of AudioFile too dramatically would bring howls of protest from their existing user-base; however the improvements are dramatic. The colour TFT screen is excellent—you'll never want to look at a CRT monitor again. The speed of operation has been dramatically improved and the programmers are beginning to take advantage of the inherently multitasking parallel processing transputer architecture.

From the start, AMS Neve imposed certain constraints on AudioFile. All crossfading and play out would be done in real time, no rendered crossfades and so on. This remains the case today.

Tracks, outputs and data streams may need a little elucidation—a Track is simply a graphic representation of a data stream or streams. If a crossfade is taking place, track two data streams from disk are in use simultaneously for the duration of the crossfade. Thus if there is a crossfade edit across 16 tracks, 32 data streams are in use. Some DAWs do not allow crossfades of more than a few milliseconds within tracks or require them to be rendered. This means that in sound-for-picture work where most edits are crossfades of a few frames or so, you have to use two tracks or render (rerecord) to disk. This uses up disk space and takes a little time.

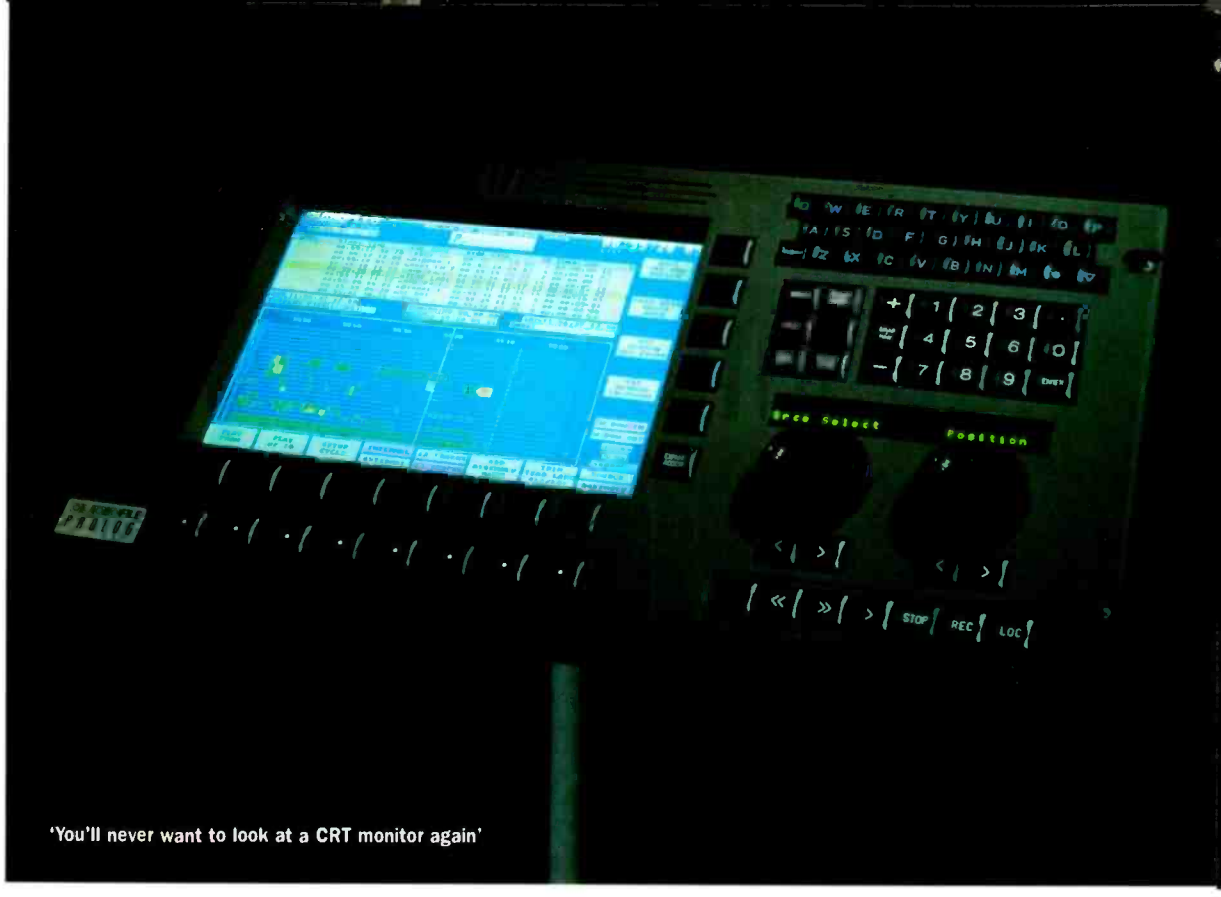
One of the hardware limitations which necessitates these approaches is the number of data streams the device is capable of playing simultaneously, which in turn is dependent on factors such as disk bandwidth. In some other machines large FIFO buffers are used to get around the bandwidth restrictions. The disadvantage with this approach being the time it takes to load the buffers before pressing play has any effect.

Prolog (and its siblings) works on a Cues and Events model. Each section of raw audio recorded into the machine becomes a Cue that can be of any length up to the available storage capacity of the destination drive. Events are instructions to the machine telling it which Cue to play, how much of it to play, where to play it, what output to play it out of, what level and balance to play it at, and whether to fade in or out. Thus a complete project will consist of a number of Cues and an Events list that defines how the Cues will be played. Cues may be used once or many times in one or several projects.

There are usually several ways of carrying out any given operation with Prolog. Which you use depends on what process you are doing at the time.

Recording can take place in a variety of ways and in different screens depending on circumstance. The simplest way to record Cues into the system is in the Source Record page. Recordings can be mono or stereo, and each is automatically assigned an assignation that can be changed to something meaningful either while recording or later. There are other ways of recording against time code.

Cues can be topped and tailed (nondestructively) and be given sync marks to expedite use in projects—if the sound of an aeroplane landing has been recorded as a sin-



'You'll never want to look at a CRT monitor again'

gle cue and there is silence at the start and finish, and a squawk where the tyres hit the runway which is the likely sync point, the head and tail can be trimmed to remove the silence and the squawk marked.

Cues are placed into an Event List in the Assembly page. Cues are initially placed on a default track or tracks and then moved to other tracks as desired. There are numerous options which define how a Cue is placed and what happens to other events on the same track—the aeroplane effect could be placed with the marked 'squawk' as the sync point. Cues can also be looped to produce, say, long atmospheres from short sections.

If required, unused portions of Cues can be destructively erased by using the Discard Ends function. This recovers disk space and can be very useful where a long Cue is recorded on spec and the required audio is very short—when auditioning sound effects you can save time by recording as you listen then discarding the unwanted 'ends' later.

Editing can be carried out in a number of ways, but the recommended method for most purposes is to use the Cut & Splice and subsidiary Trim page. For the experienced user, the Trim page offers one of the most versatile and quick ways of refining a crossfade edit available on any DAW. The edit points can be slipped individually or together, and the fades can be adjusted individually or together. Fades up to 200ms duration are equal energy.

There is also a powerful, but potentially dangerous, Macro function. Sequences of key presses can be mapped to keys to perform complex repetitive tasks with a single keystroke. The danger lies in invoking a macro in a different screen to the one in which it was recorded. This can have unexpected and destructive consequences.

The new V2 software also contains comprehensive ADR functions. These allow the creation of 'loops' for revoicing, and there are programmable countdown bleeps and relay outputs for cueing.

The TimeFlex function, accessible from the Event Edit screen, allows the time compression or expansion of a piece of mono or stereo audio without affecting the pitch. The maximum 'squeeze factors' are 50% and 200%. As with most time-domain effects, artefacts will be audible on some material and these will be more obvious on the higher percentages of squeeze. However, since you can audition the TimeFlex immediately, if there is a problem a small change in squeeze factor may cure it. Audio can be stretched to 100th frame accuracy. The required length can be specified or you can shrink or expand to fit a given space.

Options divide into three groups: hardware interfacing, hardware storage and software.

Prolog comes as standard with four analogue inputs and 16 outputs which may be supplemented or replaced by AES-EBU cards. The limit is a total of 8 in and 16 out of whichever flavour at any one time. Further interfacing options cover more comprehensive machine control with the MCS Multi-Machine Control System which allows direct control of up to six external machines and-or links to external synchroniser systems. The latest software allows remote machines to record under the control of the AudioFile.

Standard storage is six track-hours of Winchester disk. Options include Exabyte for backup, Jaz, Maxoptic Tahiti 3 1Gb magneto-optical and removable Winchesters.

SOFTWARE OPTIONS ARE available for autoconforming, the EDL package, OMF which confers the ability to read disks from Mac-based nonlinear picture editing systems such as Avid Media Composer. Transfer from Lightworks editing systems is part of the standard software. There is also a module available that allows the Prolog to 'hang off' a VT edit controller as an additional VTR.

The EDL software is worthy of note as it is a very powerful tool for the postproduction of film and TV projects. It includes such luxuries as the ability to map source and destination tracks and to read multitrack lists. It also performs intelligent conforms. Where it cannot find exact matches between Cues and the EDL it compares in-out times to find missing events.

Prolog is a relatively cost-effective way of joining the AudioFile club. It may be limited to 16-bit working, but for many applications this is all that is required. What you do get is a full 16-output device with the ability to tracklay over a total of 24 tracks for later replay on larger systems.

Prolog and other AudioFiles perform highly specific tasks with speed and precision; Trim is one good example. An experienced operator can go through an entire dialogue tracklay, sorting out edits, using Turbo Trim, at a speed that has to be seen to be believed. The main strengths of AudioFile lie in scalability and workflow. They are well suited to facilities where processes require work to be moved from, say, autoconform in one room, tracklaying in one or more other rooms and mixing in a third. Their connectivity and machine control functions are impressive—not to mention integration with the estimable Logic series of digital mixing consoles.

AudioFiles are not immediately intuitive to the generation brought up on PCs and Macs. They are not 'I can make whole programmes five minutes after opening the box' machines. If you invest the time to learn to drive them instinctively they offer a range of comprehensive editing tools appropriate for people who are regularly going to do very serious work in a big hurry.

Over the many years AudioFile has been in production, there have been many tangential developments to the original concept which have enhanced it out of all recognition. The story is by no means complete—AudioFile will continue to develop, perhaps in ways we would not expect. So it only remains for me to say, 'The prologue...' ■

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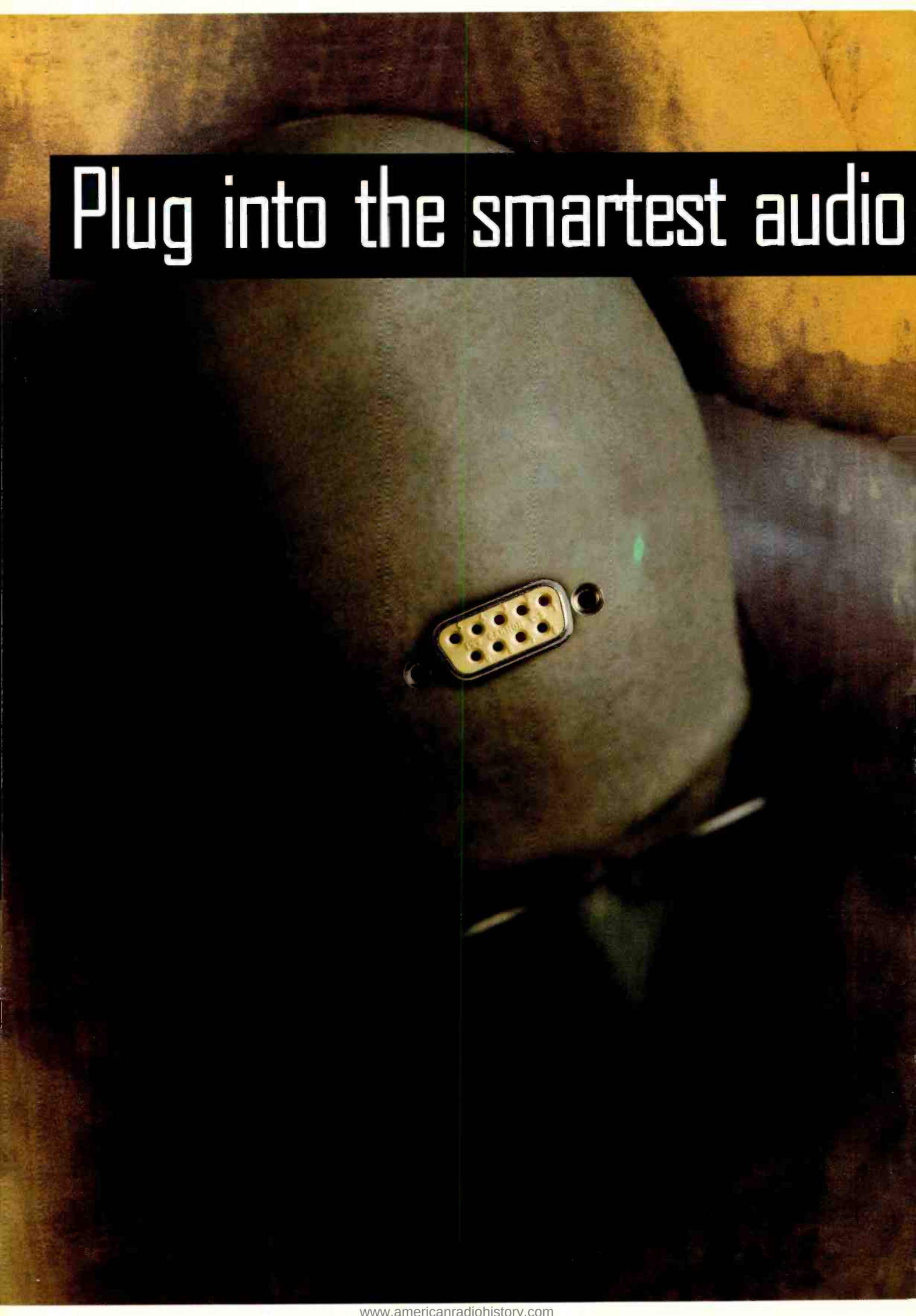

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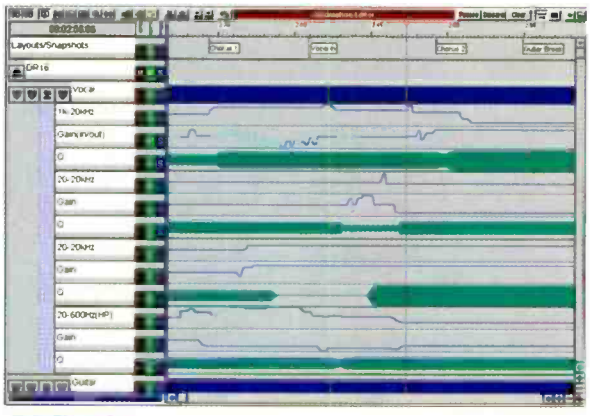
Soundtracs Virtua V2

Although digital desks represent the latest technology in a modernist pro audio industry, to maintain high performance standards regular software upgrades are becoming a necessary part of studio life. **Zenon Schoepe** investigates the refinements of a major software upgrade for the Virtua

AFTER the confusion concerning the name of Soundtracs' first digital console—which was launched as the Virtua, then, recently and briefly became the Virtual only to revert back to good old title of Virtua—and despite the attention that was grabbed by its new DPC11 large scale digital desk, the Virtua continues to fulfil a price and functionality niche in a sector that is threatening to become more closely populated. Soundtracs has recognised the importance of its core digital product with the release of an upgrade to the system's software control. It's a sign of these software-driven times that a 'mere' software upgrade should warrant close examination, but the Version 2 development is an important one for the product because it responds to many of the few shortcomings that the desk had on first release (*Studio Sound*, November 1996).

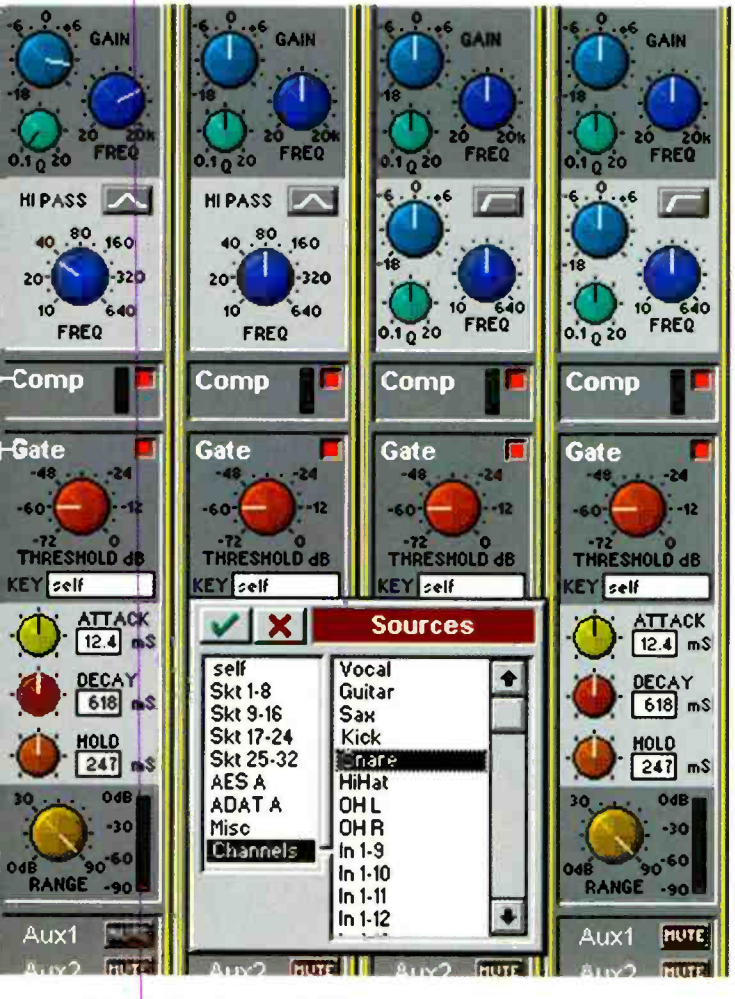
The result of the most desired requests on a wish-list compiled from users feedback, Version 2 adds a number of significant enhancements and modifications, and is a pure software upgrade with no modifications to the hardware of the system. It is therefore relevant to the considerable number of existing Virtuas in the field and is now the state of the desk's abilities. However, you won't be able to spot a V2 Virtua by looking at the control surface as the majority of tweaks have been performed in the on-screen displays which have been altered to reflect the release of more of the system's power.



EQ Edit screen

While the company originally marketed the console as having 8 auxes and 8 group buses supplemented by a large number of direct outs that could be addressed, V2 has placed the selection of the busing format into the hands of the user —something it alluded to at the launch of the console. Whereas before the desk would ask at setup whether you wanted to mix in stereo or surround, it now gives a list of group buses and permits the selection of how many mono groups, stereo groups, and surround stems are required, and similar options now also exist for the aux buses. A tracking setup can thus reduce the number of aux buses and employ more group buses with the balance reversed at mixdown.

This amounts to more flexibility and is very welcome when mixing in surround where break-off groups can be created for things like dialogue. It's something of a philosophical difference in how Soundtracs approaches the business of allocating the Virtua's resources —rather than allowing the processing to be assigned it's chosen instead to make all the processing available on every channel and concentrated instead on allowing the free assignment of buses. It's a subtle distinction, but arguably significant.



Key HPF screen

V2 coincides with the arrival of extra control surface pods, essentially channel control sections of the master control surface, but without its master facilities, which not only increase the amount of physical control available, but also allow operators at each pod to alter values and parameters simultaneously on their associated signal paths. Flexibility has also been added to the work surface as channels can be moved around and reorganised on-screen in addition to the touch-fader surface method of locking channels onto the surface.

Talkback can now be routed to external sockets in addition to the buses and auxes with latching and nonlatching switch response while a decidedly top-end style auto cut function cuts the talkback when the desk receives time code and re-opens it when code stops. Solo buttons can now latch (and interlock) or be momentary.

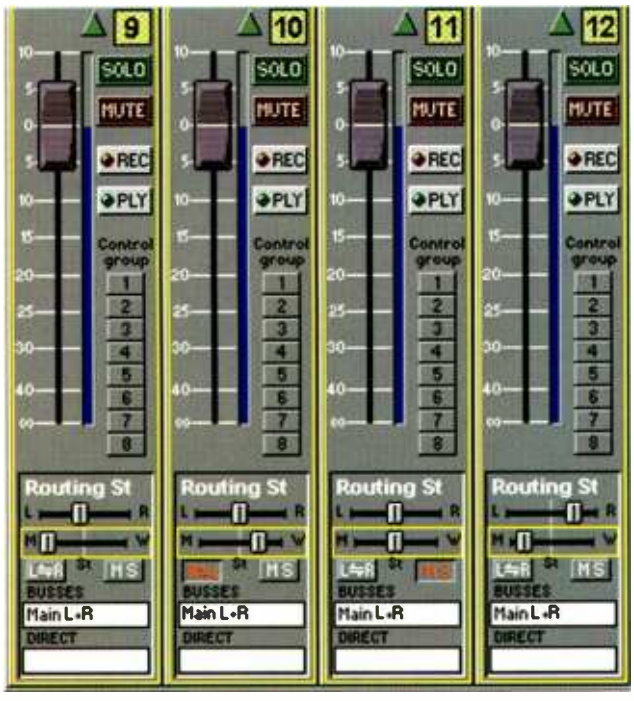
MS input decoding on any stereo input is now possible with new width controls that are dynamically automated as are MS and AB leg reverse switching. Gates can be keyed from specified sources either before they are processed, or from fully processed named channels.

The rather excellent EQ has been augmented with a 12dB/octave sweepable high-pass filter that can be substituted for a channel's LF EQ band, and the switching of this function is automated. The desk now also supports multiple format mix outputs in a variety of analogue and digital simultaneously.

Perhaps the greatest advances in V2 are presented in the desk's automation which always looked promising, but suffered from some early blind-spots. These concern themselves with what is automatable on Virtua —now pretty much everything you could want—and how this extra data is **page 20 >**

< page 19 presented for the purposes of editing. How Soundtracs has tackled the difficult task of illustrating dynamic automation data is clever given that its attempting to display surround-sound panning among other things. It's chosen to use vertically scrolling bars against a timeline for each parameter and in the case of surround panning employs squiggly lines for the left and right quotient, and a second similar bar for the front to back information.

This is not an immediately clear representation and requires something of a mind set, but it's important to remember that this display of automation data is not there for hard value editing—it is far easier to pick up and adjust a controller from the desk surface—but rather it allows this data to be manipulated within the powerful off-line mix editing functions and its cut and paste-style capabilities.



Busing screen

The business of showing the large number of channel parameters on endless vertical bars is facilitated by the breaking down of the automation screen display into blocks of inputs as they are arranged on the desk controller surface. This display will additionally track and keep up with any rearrangement to the physical controller surface. Access to these blocks of channels is aided by drop-down buttons which are assisted by yet more drop-down buttons corresponding to the automatable sections of the channel paths, such as auxes, EQ and dynamics.

Auxes are shown with levels and mutes plus panning information, where relevant, which is simple enough, but the 4-band EQ, and its switches, is more of a challenge. Squiggly lines show frequency and gain and a novel 'width' display depicts Q value. Those paying attention will have spotted the potential for copying tempo-based filter sweeps along a track in off-line.

Compressor and gate threshold level displays are provided, but substantial improvements have also occurred in the dynamic automation system. This now has the facility to put a single control into overwrite record and has two versions of Safe mode: Safe from the dynamic automation data, which still leaves the controller susceptible to snapshot automation, and Safe from everything. The benefits are apparent.

Some of the functions badly missing from the first incarnation of the automation that have now been added include touch record on the faders, plus autonull release and a programmable autonull time that was previously fixed. Autonulling is also available on the pot controllers, which while not touch sensitive (they require a turn to instigate an action) can pick up the automation and when Play is activated the last value cross-fades back at a programmed rate to the previously recorded data. These are major enhancements to the usability of the automation.

The system now supports the recording of snapshots into the dynamic automation together with a label in the timeline, and these events can, just like everything else that is automated in this system, be edited, and dragged, and dropped. There are also separate record and play statuses for snapshot and dynamic automation.

Mix pass storage has been greatly improved. Every time you come to the end of a pass the system will store it while also retaining the previous pass. You can't perform seamless and butted comparisons of the current and previous passes, but because they can be stored without committing yourself, you can audition 99 passes at leisure, each tagged by date and time. Hooray! It's what they should have done with Tracmix years ago.

The system still can't work to bars and beats, and none of the enhancements in V2 can address Virtua's lack of visual pot position feedback from the desk surface in the way that a logicator-equipped console does, for example. If you want to see values then you really have to get your information from the screen and this remains splendidly detailed. Most will be content with this, those that aren't should remember the price. You still get a phenomenal amount of hard control.

Whether you want to admit it or not there is no escaping the fact that Virtua has evolved one of the most powerful automation systems around. Despite threatened competition, Virtua still stands alone in its class. A good desk has just improved markedly. ■

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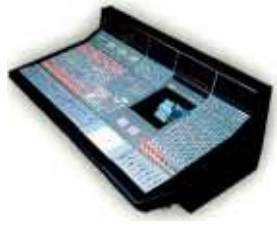

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Audio Toys Inc Pro^6

The potential of the 1U-high outboard continues to amaze—with ATI's 8MX2 and Pro^6 commanding increasing attention. **Dave Foister** checks out this highly-specified American channel strip

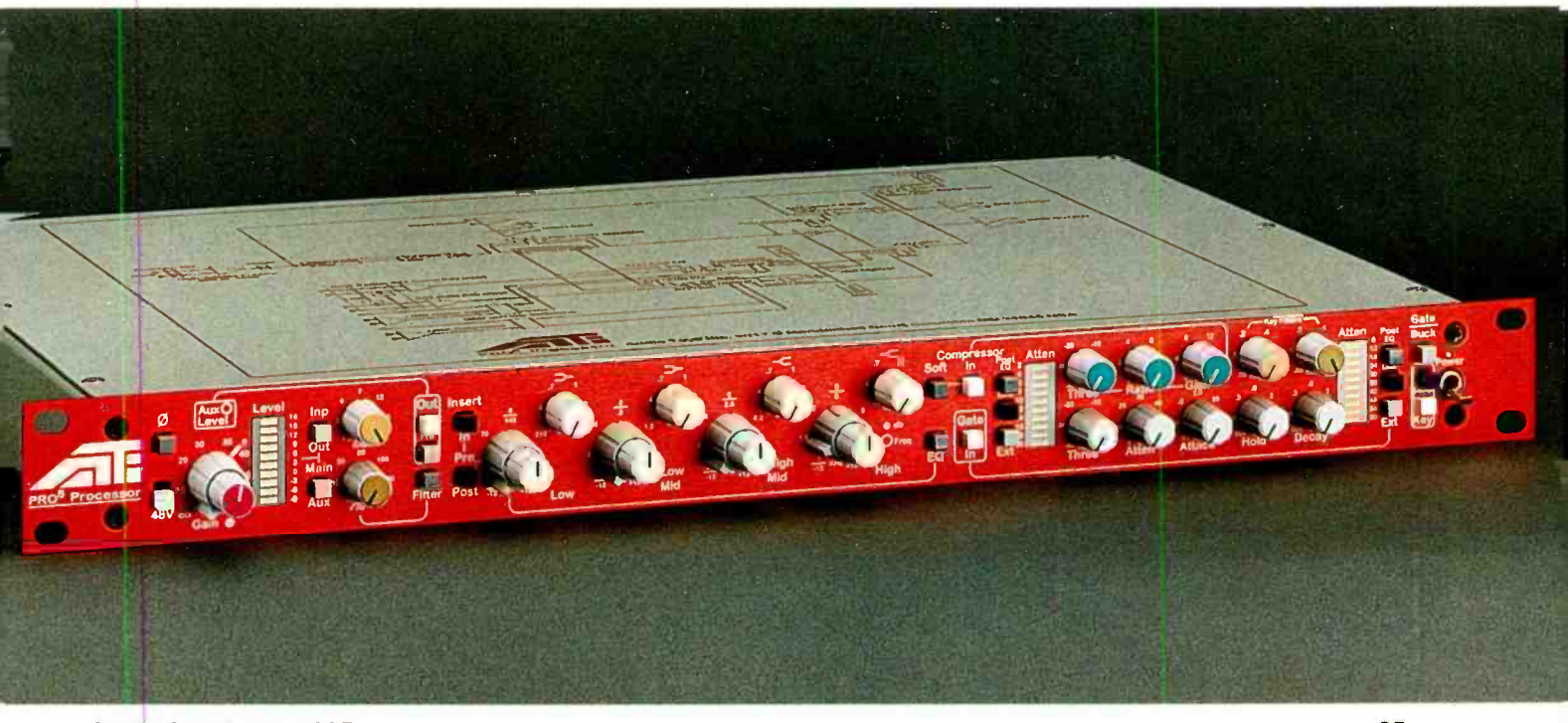
THE SPIN-OFFS from Audio Toys' modestly named Paragon live consoles seem to have been attracting a lot of attention—more than would be justified simply by their knob-encrusted metallic red panels. ATI's stated reason for producing them was engineers' wishes to see elements of the consoles available for use elsewhere, and nothing could fulfil that requirement better than the Pro^6, a complete signal path with no less than six circuit blocks within a 1U-high racknount box. Like the sister 8MX2 8-channel mixer (reviewed *Studio Sound*, March 1997), there's so much squeezed into the unit that the front panel is inevitably crowded; both make a remarkable job of keeping it manageable, despite having even more facilities than one would at first assume.

The Pro^6 is effectively a complete mixer channel strip, and quite a comprehensive one at that. The single-preamp stage claims to have the headroom to deal with both line and microphone levels without a pad, adjusting gain with a solitary detented control. It has switchable phantom power (with the only indicator LED on the panel) and phase reverse, and it looks more complex than it is thanks to the presence of various controls for the unit's auxiliary output (covered later) placed oddly among its own. Powerfully wide-ranging swept high-pass and low-pass filters follow, with steep 24dB per octave slopes, and frequency spans that almost meet in the middle. At the other end of the scale, their ranges effectively leave the audio spectrum, at 10Hz and 40kHz. Like most other blocks, the filter section can be switched in and out, but in practice this is hardly necessary given the quality of the filters and the wide window between their outer frequency extremes.

Where some such preamp/channel strips have simple EQ that is sometimes no more than a token gesture, the Pro^6 goes the whole hog with a 4-band fully parametric equaliser. All the bands can be used in peaking mode with variable Q, and unusually all four can also be switched to shelving, two at the upper end, and two at the lower. Frequencies are fully variable in either mode, with sensible overlapping ranges, and the available gain of each band is a healthy 15dB either way. You'll look in vain for switches to select peak or shelf operation; to save precious real estate the Q controls have push-button mechanisms under them, which when in the out position put the band in shelf mode, disabling the Q control. This adds up to a really powerful equaliser, with at least as much scope as many dedicated units, and it also sounds remarkably good, musical and unstrained.

The compressor section, which uses ATI's patented circuitry, is perhaps the simplest on the unit. It has only three main controls, for Threshold, Ratio and Gain, as its time constants are programme-dependent. Flexibility is provided by a SOFT KNEE switch, and gain reduction is shown on a vertical LED ladder. ATI has spotted a useful corollary to the combination of EQ and compression in the same box; the control signal for the compressor can be taken before or after the equaliser, even though the EQ always comes first in the signal chain. Besides the obvious result that the raw signal can control the compression regardless of the extremity of the applied EQ, this means that if the EQ is not required in the main signal path it can be used to tailor the compressor response for frequency-conscious action. The obvious application is de-essing, and ATI reports that some users reserve their Pro^{6}s for this purpose, preferring the results to those from dedicated de-essers. Certainly, the control offered by the four extremely flexible bands makes setting up for this kind of job very straightforward, helped still further by the compressor's key listen facility which monitors the side chain. This can also be used to check an external control signal, for which there is a balanced TRS input on the rear and a front panel select switch.

The same facilities are available for the gate, which, surprisingly, is more comprehensively equipped than the compressor in terms of user adjustment. Wide-ranging ATTACK and DECAY (Release) time controls are joined by a variable hold time, and adjustable attenuation when closed complements the THRESHOLD control. The resulting gain reduction is shown on another LED meter. A small point is that both these gain reduction meters only show anything when the associated process is switched in; I find it useful to see what the page 27 >



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< page 25 control signal is doing before the process is added, just to check that the control settings are in the right ball park.

The same option to derive the control signal pre or post the EQ is available as it is on the compressor, but here dedicated filtering is also provided, with 24dB per octave high-pass and low-pass variable filters, both spanning the range 50Hz–8kHz—drastic enough for almost anything. Again there is a Key Listen facility as well as a front-panel selectable external key input, which can be linked to the compressor's external control input with a rear-panel switch. Besides this, both dynamics sections have DC control links in and out available on the back, so that two Pro[e] units can be set up as master and slave. Note that both processes work on the same VCA by summing their control voltages: the signal path remains the same whether one or both are in use. To complete the picture, the gate has a switch to invert its operation to become a ducker.

IT WILL BE apparent by now that most of the facilities on the Pro[e] are far more sophisticated than they at first seem, and that they have certainly not been pared down or compromised in order to squeeze them into the box. Typical of the unexpected extra detail is an insert point, fully balanced, which can be placed before or after the EQ by a front-panel button. Connections for this and several other ins and outs are on TRS jacks to save space, but all remain balanced; the insert has separate send and return jacks rather than the unbalanced single TRS often found elsewhere. There is even a ground-lift switch on the back for the insert and key input jacks, separate from the ground lift for the input XLR.

Further possibilities still are offered by a secondary or auxiliary output. This is not an aux in the conventional mixer sense, but an additional output which can be tapped off at various points in the signal path. Since one of those points is immediately after the microphone preamp this can provide the shortest signal path for instances where the processing is not required. In this context it should be noted that the VCA which provides the compressor and gate functions is always in circuit to the main output; the associated function switches simply connect the relevant control circuit to the VCA. The aux output can also be fed immediately after the EQ and before the VCA.

These selection switches are found among the input controls, and there is also an output level control for the aux signal. Also nestling at this end of the box is the main level meter, which can be switched to show the signal after the preamp, the main output or the aux signal. This slightly odd layout is the only exception to the general logic and simplicity of the front panel, which as a whole represents a remarkable achievement; to pack so much functionality into 1U is a feat in itself, but to lay it out so that operation is as intuitive as this takes some doing. There's enough space between the controls to get your fingers in, and the zoning and colour coding of the knobs makes perfect sense, so that you soon forget how compact it is. Some of the colours ATI has chosen are a little idiosyncratic—the input level knob in particular is a livid shade of mauve—but the resulting clarity, especially set against the red panel, makes any aesthetic affronts tolerable.

The appeal of a unit like this depends on two things, assuming of course that the overall audio quality is up to the mark, a factor never in doubt here. The first is how well the individual circuit blocks operate, and the second is the selection of processes the manufacturer has chosen to include. Here approaches differ, with some apparently feeling that their products need an unusual twist in order to get noticed while others believing that going too far beyond the conventional signal path will put purchasers off as they don't want to pay for something they will rarely use. ATI has adopted the latter stance, with a comprehensive collection of standard facilities which most engineers will need every day. As to the first factor, the individual blocks within the Pro[e] are without exception worthy of consideration as processors in their own right, and to have them all provided in such a convenient format makes for a winning combination. ■

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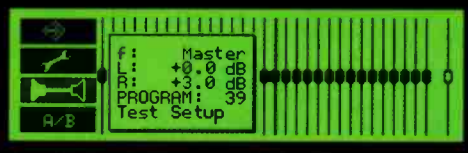

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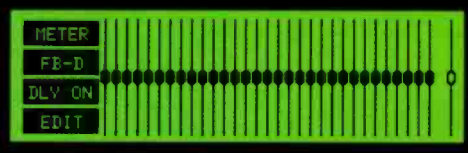

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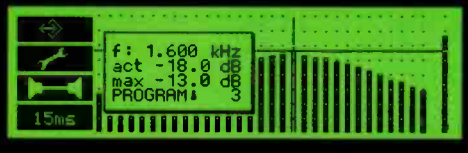

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vides a direct interface to the ADR-Foley console's playback monitor as well as track arming switches. Martinsound's AIP Pro Section automatically handles the switching of various monitoring and record functions.
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JBL monitors

JBL's DMS-1H studio monitor is now shipping with a choice of DSC280 digital system controller or the SMC24 analogue system controller.

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AKG

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AKG has introduced XLR phantom powered versions of the C921 and C947. They come in black and white, and have a dedicated suspension with non-twisting cable.
AKG. Tel: +43 1 866540

Meyer self-powers

Meyer Sound has introduced several new self-powered loudspeakers. The CQ-1 and CQ-2 are phase corrected, sound reinforcement boxes that are configured as high Q (the CQ-2) or low Q (the CQ-1) units. Both can be flown and arrayed, and offer consistent coverage for all frequencies in the horizontal and vertical planes, hence the name CQ ('Constant Q'). Both feature a 15-inch low frequency driver.

The PSM-2 is a self-powered monitor wedge with a medium Q horn, with even coverage in both planes and phase page 30 >

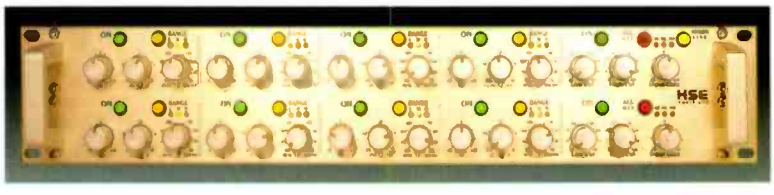
HSE EQ1

Traditional Swiss sophistication finds itself at home in this new high-quality studio parametric equaliser. **Terry Nelson** puts it to the test

THE MOMENT the HSE EQ-1 parametric equaliser comes out of its box you get the feeling that it means business. The stereotype of Swiss excellence in hand-crafted equipment made by small companies is borne out here, as the EQ-1 oozes the necessary indications of quality and functionality. But does the world need another parametric equaliser? HSE seems to think it does and after some fairly intensive testing, I think I would agree with them.

The EQ-1 comes in a standard high quality 2U-high rack chassis with a suitably thick 'champagne' front panel and custom-turned aluminium knobs (you won't get replacements for these off the shelf). It features four fully-parametric EQ sections per channel, plus variable high-pass and low-pass filters and input gain controls.

The EQ sections are identical and with cut-boost control (±15dB) and dented 0dB position, variable Q (0.5–10) and sweep frequency control. In addition, there is an illuminated EQ IN-OUT push-button and a RANGE switch with three associated LEDs indicating L (low), M (medium) and H (high). Repeatedly pressing RANGE cycles through the LEDs and selects one of the three frequency ranges: 10Hz–300Hz, 300Hz–1kHz, 1kHz–30kHz.



The master section features low-cut and high-cut filters plus illuminated IN-OUT push-buttons, input gain controls (±20dB with 0dB detent), green signal present LED (threshold at -40dB), red overload LED, SW LED (more later), master EQ defeat for each channel and STEREO LINK switch.

The overall impression made by the EQ-1 is very sober and workman-like. Admittedly, HSE says it prefers to invest in hand-picked components and advanced circuitry rather than frivolous cosmetics, but it's not the way I'd have styled it.

Connection is via balanced XLR connectors and there are two trim controls—factory set to unity gain—should any fine tweaking be required to balance the channels or suit studio operating levels. The IEC mains connector features an integral on-off switch so once the unit is in the rack, you turn it on and off and leave it. The final connector is a mono 6.3mm jack for a footswitch.

I used a variety of material to test the equaliser, from individual raw tracks through to master DAT's needing a magic touch. My initial impression was one of extra depth and body to the sound and this is no doubt thanks to the EQ-1's class-A circuitry. In fact, the danger when first using the EQ-1 is to over EQ as the sound tends to remain very smooth and it is only when you compare the unequalised signal with the processed signal that you realise just what a difference it makes.

Rather than employ a hard bypass, the unit functions as if all gain pots are set flat—or as an extremely neutral line amplifier. It does, however, employ 'Fix Q Technology' which means that the Q factor stays constant whatever the gain setting, unlike equalisers that reduce Q with gain.

I found the equaliser very capable of picking out instruments that have been buried in a mix—or not been mixed high enough—and bringing them out without affecting the overall balance adversely. Some careful adjustment of the Q and gain produced some very interesting results. Setting two of the EQ-1's sections close together in the same frequency band and boosting one while cutting the other did not have any detrimental effects either—and proved useful with a female vocal track where I could take out some lower-mid while adding warmth just a bit higher.

Initially, I must admit to having cast the equaliser in the mastering category, but while it fits perfectly here, it has much more to offer. The EQ-1 works perfectly as both problem-solver for difficult tracks as well as enhancing good ones and manages to instil body and an overall 'shine' where it might otherwise be lacking. And as a sound shaper it is great —I was able to fine tweak sounds without any of the harshness or loss of depth that can be characteristic of other units.

The mono jack mentioned earlier gives access to the Switch function and this allows two layers of switch status to be programmed and changed with a footswitch (or any other closed contact).

Programming is simplicity itself: set up Layer 1 and then Layer 2 (indicated by the SW LED). You can then toggle between the two layers for different EQ settings as required. All switch operations—with the exception of the LP-HP filters—are silent so you can change settings during recording, mixing, transmission or whatever.

The HSE EQ-1 represents what is virtually a custom-made parametric equaliser at an 'affordable' price—expensive but not ridiculously so. It looks good, sounds great and can be used in any application where creative or corrective equalisation is required—from single sources to finished product. Top of the class. ■

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Brauner VM1

The choice of microphones available has never been wider; so the issue is now one of quality. **Dave Foister** tests a German valve mic

THE REBIRTH of the microphone has had far-reaching consequences. It has spurred on the top manufacturers to rediscover their past; it has encouraged smaller players to re-invent themselves with much higher aspirations; it has brought us hitherto unknown delights from the former Soviet territories; and it has enabled a selection of very small companies to offer us their visions of the ideal microphone. One such is Dirk Brauner's VM1.

If you didn't know that the VM1 was German, you'd soon guess. Every detail smacks of precision and quality, and plenty of unusual details there are. It is (naturally) a valve microphone, with a clearly-visible dual diaphragm capsule reminiscent of a well-known compatriot, all the way down to the big screw in the middle. The capsule, like the rest of the microphone, is hand-built, and sits in an open-mesh cage atop a substantial cylindrical body, all with a familiar satin finish. Being a valve design, there are no controls on the microphone itself, any adjustment being made from the power supply.

Stand attachment is by means of a large suspension mount, supplied as standard, with a novel means of holding the microphone. The body slides into a metal sleeve, and plastic collars top and bottom are turned to tighten internal circular clamps, making for a very secure arrangement. The mount's angle is adjusted with a lever-operated clamp, which the manual admits is so far over the top as to risk breaking something if it's over-tightened. The result is that even such a heavy microphone is supported as reassuringly as it possibly could be, although the danger of a badly-arranged stand toppling over is still present.

A substantial white cable terminated in Tuchel multiways links the microphone to its power supply, which is itself quite eye-catching. It appears to be intended to form part of a modular rack, with all the connectors on the back and the necessary controls on the front. Connectors comprise IEC mains in with an illuminated rocker switch, microphone in and XLR out, the last having an associated 3-position ground lift switch. This provides hard grounding from signal screen to chassis, complete disconnection, and soft grounding via a capacitor.

The front is dominated by a single knurled silver rotary control for polar pattern adjustment. This offers the unusual luxury of being infinitely variable from omni to fig-8 rather than a multi-position switch, a feature I can only remember seeing before on a very few top-end microphones. While the nine standard settings often provided elsewhere may be more than some people feel they need, the fine in-between adjustment offered by the VM1's arrangement can sometimes make all the difference. The only other control is a toggle switch for a 10dB pad, which when on turns the power indicator LED from red to green. No low-end filtering is provided.

The manual is a joy. It looks to have been individually printed on Brauner's ink-jet printer, and all the illustrations are pencil drawings. This is not to suggest that they are crude: in fact they are almost works of art, with a texture and detail that comes across better than a poorly-reproduced photograph would. Not that there's much to say in the way of instructions, other than to explain how to get the microphone into its mount, which without the manual's explanation of which screws and marks to line up is something of a challenge.

What the manual does do is go out on a limb with claims about accuracy, transparency and all-round quality which could leave it looking pretty silly if the microphone's performance was anything other than excellent. Happily it lives up to the claims; the moment I pushed the faders up I knew this was something special. Forget preconceived ideas about valves being warm at the expense of top, or present at the expense of clarity: the VM1 is as open and natural a microphone as you could wish to find. The top seems to go on forever, placing nothing in the way of a cymbal's sparkle when used over drums and delivering all the subtleties and air of the upper end of a good piano. The bottom never suffers; the same piano retained all its depth alongside the transparent top.

With a pair of VM1s to try, the piano and kit had spaced omnis, and I was also able to rig 90° fig-8s for an *a capella* vocal group. This worked immaculately, showing a superb stereo image, and the full clean vocal sound continued with a close vocal on cardioid. One or two of the voices I recorded would have benefited from a bit more flattery than the VM1 provides, but that's not the microphone's fault; this is a truthful, delightful microphone that deserves close inspection by serious microphone users of every kind. ■

Contact
Brauner, Van de Wallstr 64, DE-46499, Hamminkeln, Germany.
Tel: +49 2856 9270.
Fax: +49 2856 9271.

NEW TECHNOLOGIES

< page 29 corrected frequency response.

On the subwoofer front the PSW-2 has a dual amplified, fourth order bass reflex construction and can be arrayed and flown.
Meyer. Tel: +1 510 486 1166

Spendor

Spendor has introduced the power QT100 nearfield which uses the company's polymer cone, low distortion 190mm elliptical

driver, and 28mm soft dome tweeter crossing over at 5kHz in an asymmetrical reflex ported cabinet. The integral amps are capable of SPLs of 103dB at 1m with a cabinet response of 68Hz to 22kHz (±2.5dB).
Spendor. Tel: +44 1323 843474

Double debut

The AD144 mixer is designed for studio and outside broadcast use. Featuring four outputs and four auxiliaries, and extensive monitor features, it is now available in standard 12-24 channel frames. An optional communications module will be available from the middle of the year.

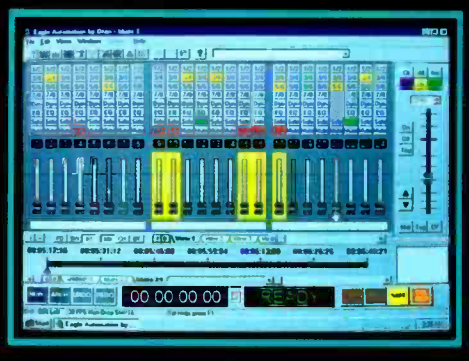
The AD149 console for film and drama applications features two outputs and two auxiliaries, continuously variable gain, comprehensive switched EQ, channel inserts and limiters. This console replaces the AD062, offering more features in a smaller footprint. There are two levels of multipin options for fast setup and break down. 6–12 channel versions are available.
Audio Developments. Tel: +44 1962 868830

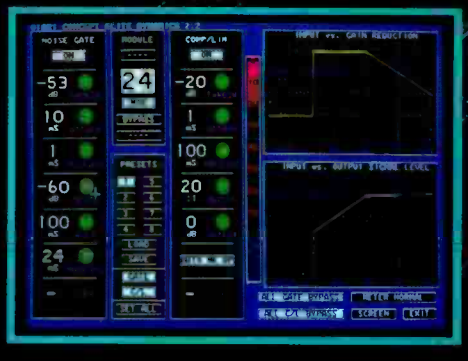
See more at C4

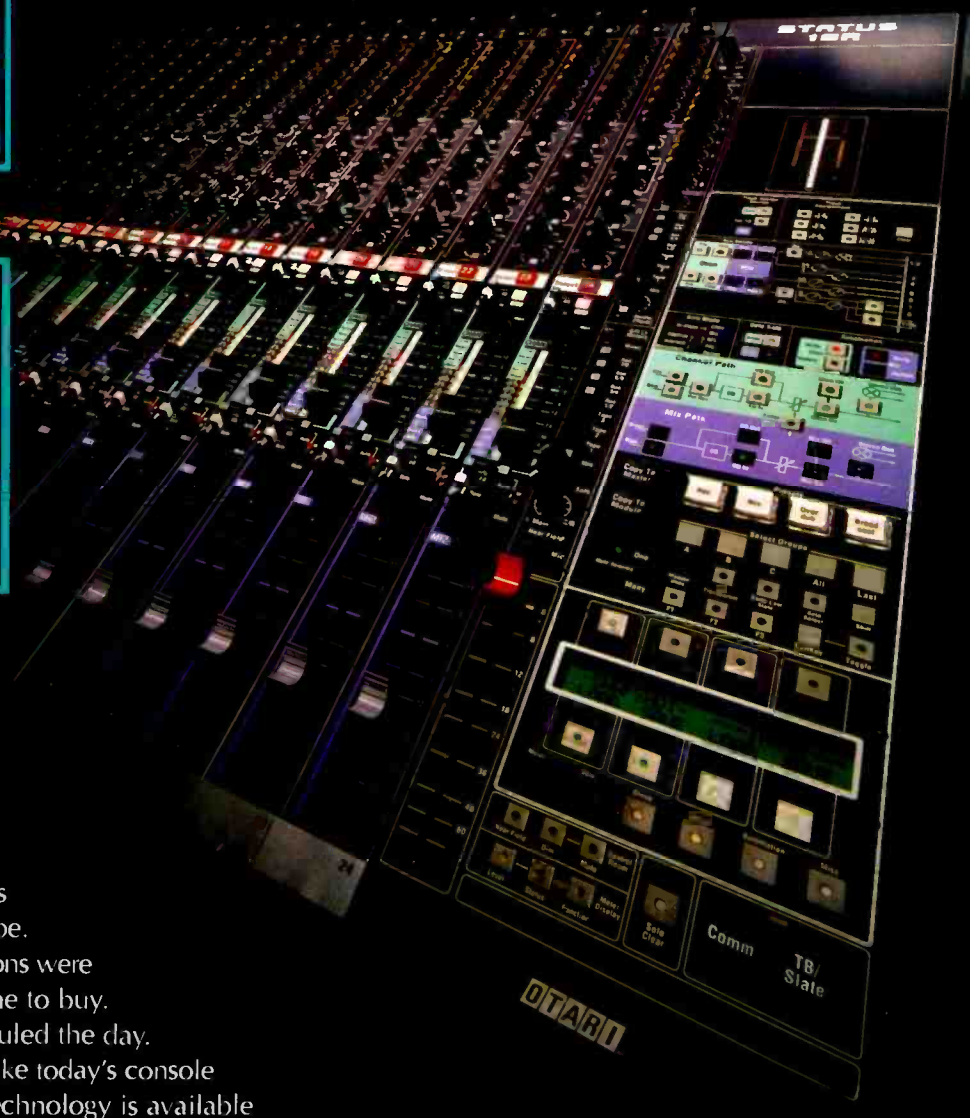
The SPDA1 RCA to XLR adaptor connector with a built-in format convertor from C4 enables low-cost and easy adaption of digital consumer units to professional standard. It converts the 16-bit SPDIF to AES-EBU including level, impedance and data structure matching. The adaptor includes a PLL jitter reduction stage and can be phantom powered with 9V DC (8.5-12V) from the balanced XLR output in addition to unbalanced powering.
C4. Tel: +49 5331 76985

KT speaker processor

The DN8000 loudspeaker processor combines a 5-way crossover with digital multiprocessing applicable to FOH systems with delayed stacks or multi-zoned installations. The device has 32 user memories, page 32 >







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Zefiro Acoustics ZA2

So you need to get audio into your PC, but you're confused by the selection? **Rob James** does some of the leg work for you

THERE HAS BEEN something of a deluge of interface cards to enable the PC literate to get sound into and out of IBM clone PCs of late. These range from simple(!), soundcards that offer analogue input and output in consumer formats through to those with General MIDI and maybe Wavetable synthesis, to cards dedicated to I-O with analogue and-or digital interfacing. Simple cards offer record or playback, while more complex cards allow simultaneous record and playback, sample-rate conversion, and so on.

One newcomer to the field, the ZA2 from the American Zefiro Acoustics, is interesting for a number of reasons. For a start, it offers true AES-EBU interfacing in addition to the SPDIF and Toslink (optical) consumer digital interfaces commonly found on more affordable cards. There is also an analogue output for monitoring purposes. To digress for a moment, I have yet to find an internal interface card on which I would be happy to use the onboard convertors for anything other than monitoring. A PC is a very noisy environment not designed with analogue audio in mind. Introducing analogue signal and expecting it to survive untainted is asking a lot.

The Zefiro hardware consists of a half-length ISA bus PC card with phono sockets for SPDIF digital I-O, two Toslink connectors for consumer optical I-O and a 15-pin D-connector, and a breakout cable that terminates in two XLRs for AES-EBU I-O and two phonos for analogue monitoring. Future hardware developments slated are breakout cables with BNC connector for wordclock and LTC SMPTE time code I-O. Drivers to enable 20-bit recording under DOS are already available.

Basic software is included that allows recording and replay of .WAV files, and real-time playback of MPEG1 Layer 2 encoded files. The Crystal Semiconductor DSP engine (CS4290) that powers the ZA2 was originally designed as an MPEG1 Layer 2 real-time decoder, hence you get this function thrown in. The UK package includes the manufacturers disk of drivers and an update disk from the distributor with any later versions since the card left the factory. Future software updates are obtainable via download or on a diskette.

There are Windows drivers to record or playback from inside a variety of popular editing packages such as Soundforge, the shareware Cool Edit, Cakewalk Pro Audio, Cubase, Wavelab, and so on. Although not currently endorsed by the manufacturers the ZA2 is reputed to be the card of choice for Soundforge programmers.

Also included is DOS software allowing data backup to an audio DAT machine. A 120-minute tape will store 1.2Gigabytes of data. Additional Reed-Solomon error correction and a 32-bit checksum on each block should assure data integrity with anything less than calamitous tape problems.

I wish I could say installation is a doddle —it can be, if you're lucky otherwise, unless you are familiar with DMA and IRQ; and the other horrors that lurk 'under the hood' of a PC, then leave it to the company that supplies the card either to install or advise you. This criticism is not aimed exclusively at the ZA2 as, unfortunately, it applies to all the audio I-O cards I have come across. A further lurking nasty, which Greg Hanssen at Zefiro has researched, is the propensity of some manufacturers of graphics cards to use software drivers that 'illegally' lock up the PCI bus to enhance the performance of their products. For most 'normal' applications this has no discernible effects, but for audio interface cards it can be terminal, resulting in glitching or channel swapping. In practice, although my PC has a graphics card from one of the offending manufacturers, I had no problems.

Also applicable are the usual caveats when using PCs for audio: you need the fastest CPU you can afford, at least 32Mb of RAM and a big, fast, hard-disk drive to avoid disappointment.

I managed to install the ZA-2 in about half an hour after a few dramas over IRQ (Interrupt ReQuest) conflicts. I first used the built-in test tone function to check channel assignment. Recording audio using the freebie wave Recorder that comes with Windows is a doddle (I used a Yamaha 03D as a convenient digital source). The ZA2 immediately recognises the sampling rate and whether there is copy protection. If the destination file sampling rate is different to the source, the ZA2 automatically converts. When conversion is taking place, the card can only operate in simplex mode —either record or replay. Where rate conversion is not required it can operate full duplex. If you have sources connected to all three inputs you can switch between them from the Windows driver. The driver also allows AES-EBU subcodes, and the SCMS and pre-emphasis bits to be set on the output. I also tried out the card with Cool Edit. The card performed without drama or fuss.

If you need a simple no nonsense way of getting AES-EBU signals into and out of a PC at reasonable cost take a look at the ZA-2. ■

Contact

Zefiro Acoustics, US.
Tel: +1 714 551 5833.
UK: RKMS, Freepost (NG 6175), Nottingham NG4 1BR.
Tel: +44 115 9611398.
Fax: +44 115 95 33802.

< page 30 two inputs and five fully configurable outputs.

Outputs have a choice of Linkwitz-Riley, Butterworth or Bessel crossover with slopes selectable between 6dB to 48dB/octave. Channels have high and low pass filters, HF and LF shelving, two parametric bands,

phase alignment, limiting, compression and noise reduction. Delay is programmable in 21 microsecond increments to one second on the inputs and 300ms on outputs.
Klark Teknik. Tel: +44 1562 741515

Rane on parade

Rane has introduced Engineered Conference Systems (ECS), a line of electronics for engineered and audio teleconferencing, distance learning, boardroom and courtroom applications, plus a new DSP multiprocessor, the RPM 26.

ECS is RaneWare RW232 software-controlled and features automixers, echo cancellers, telephone access (two-four wire), remote setup and diagnostics with password protection. It is a system rather than a selection of products, and was conceived as such from a field perspective.

The system is predominantly software programmed and controlled, using a serial port on a PC or Mac Power PC. There are currently five distinct products and others are slated for introduction in 1997. At the heart of the system is the Base ECB-6 unit which interfaces with all other ECS products and external devices. A dedicated automixer, the Mixer ECM-8, was designed specifically for teleconferencing and distance learning. Each can accommodate up to eight sources, and six automixers can be used with a Base ECB-6 to give a total of 48 inputs. The mixers interface with each other and the Base unit via a proprietary protocol. An optional echo canceller, the ECA-1, can form an internal part of the mixer. The ECS-6 stereo option allows the Base unit to process stereo programme material and mono signal sources while the ECT-1 telephone interface provides access to the telephone service and serves as an interface with the control system (such as AMX or Crestron), the Base unit and the programming computer. ECS integrates with other Rane products such as the RPE 228, equaliser and MA 6S multichannel amplifier.

The RPM 26 DSP multiprocessor contains all signal processing required between the mixer and amps in small to medium fixed installations. Using preprogrammed configurations and RaneWare software, the RPM 26 provides signal-processing functions including high and low cut filters, compression, delay, crossover, parametric EQ, splitting, input summing, limiting, level control and sine wave or pink noise generation. It contains two analogue inputs and six analogue out- page 34 >

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THE BENCHMARK IN DIGITAL AUDIO

Fairman Tube Master Compressor

For the times when your work demands the 'Fairchild' sound **Zenon Schoepe** examines a Danish solution to a long-standing problem

IT'S AMAZING what a bit of weight does to add substance to a piece of kit. Somehow, a box that is slight and featherweight doesn't encourage confidence no matter how clever the chips and solid-state circuitry may be. But Fairman's latest creation, the TMC tube master compressor, weighs in as a super heavyweight.

This mass is one of its most surprising revelations particularly as the company's original unit, the TRC Tube Recording Channel, while large felt a little small-boned, and almost fragile. The TRC combines a Fairchild 670-650 compressor clone with Pultec EQ-PM mid and EQ-P1A high-low clone EQs in a single channel for mic, line and instrument inputs. The TMC is a far simpler proposition presenting as it does a clone of the Fairchild 660-670—it's a stereo program compressor, and, for all intents and purposes, it's a copy. Inside there is an army of 16 valves awaiting your command—four per path, and four for the detector circuitry on the VCA (claimed to be identical to the Fairchild 670's) per channel, plus a monster rectifier for the high voltages. The valves are ECC81s as the original 8668s are now extremely scarce. Fairman is taking this seriously.

The presentation is a substantial improvement over the look of the TRC, and the TMC benefits from flush-mounted mechanical VU meters, monster long-throw switches and whopping great knobs for the rotary switches that would rate well in a 1950's sci-fi B-movie's idea of high tech. The whole lot sits in no less than 6U of rack space and needs industrial-grade rear support if it's not to rip the rack strips from the cabinet. It's heavy because the chassis is heavy and it employs in and out transformers that are MU metal-cored Lundahls.

The weight of the unit is all the more remarkable given that the device employs a separate power supply connecting via multipin umbilical. Power on and the box takes a few seconds to come up to temperature with the VU meter needles sweeping up to zero to tell you that it's about ready for you.

Back-panel connectors are on XLRs and these are accompanied by four fuse slots, an earthing bar and an earth-lift toggle switch.

Each channel is identically featured, but the two can be strapped together on a switch. Metering can be switched for -10dB operation and display gain reduction or output level and has zero adjustment screw access. Each channel has three rotary control switches under the control of a bypass switch which inexplicably is labelled BYPAS. You're presented with a 20-position gain make-up control, 20-position threshold control (both are 0.5dB steps) and six preset attack and release constant combinations. The last of these refers to the attack and release time settings with two letters giving an indication of their settings. Thus you get FF, FM, FS, MF, MM and SS with F standing for fast, M for medium and S for slow—not exactly specific, but then this is the sort of vagueness that builds the mystique.

This unit does wonderful and beautiful things to music. Musicality is an expression that all manufacturers feel they have a right to use when describing their gear, but it's the word I would use to describe what the TMC is all about. There is enough variability available through the six presets to cover most eventualities and in those instances where they straddle the type of envelope you want, it still sounds so good.

The FF setting is a fine starting point although MM works great on vocals and FS is the setting of choice for bass. It's all tremendous stuff, and so beautifully flattering on stereo programme material because the impression it gives is that the whole signal, and not just the bits you are most susceptible to, is being processed. It sounds the way you'd imagine compression ought to sound, but often doesn't in practice. Full and rounded, more of everything, and closer to you in the room.

About the only thing you won't get from the TMC is the smell of old valve gear and that horrible feeling of inadequacy and underachievement you get when you ask the price of something that you really like, only to learn that it is so outside your means or capabilities as to make it academic.

At a UK price of around £4,500 (plus VAT) it will still make many wince, but you have to bear in mind what you would be expected to pay for an original in sound and working order. Bargain it may not seem, but it could be construed as value for money and it is the real thing.

This is a wonderful and effective box, with a sound that is worth every penny. ■

Contact

Fairman, Musikhuset Aage Jensen, Landemœrket 27–29, DK-1119 København K, Denmark.
Tel: +45 33 14 8996.
Fax: +45 33 33 8271.

NEW TECHNOLOGIES

< page 32 puts and configurations and settings are stored in 16 internal, nonvolatile memories, eight of which are contact closure accessible from the rear panel Remote Switch Interface (RSI) port. All 16 memories are accessible from RaneWare's Site Control Panel.
Rane. Tel: +1 206 355 6000

Small mic cuts a dash

Designed for TV and broadcast applications, the compact Schoeps CMXY 4V stereo microphone can be used for single speakers, table discussions, on a boom or as a stereo spot microphone..

The microphone has three sections, the first of which consists of two rotatable cardioid microphones with lateral sound pickup which are closely adjacent to one another. The second section consists of a swivel in which the two microphones are held, allowing the microphone section to be tilted into the desired direction. The third section is the base which also contains the output connector.

CCM-L

The capsule section is only 41mm wide this corresponds to the diameter of conventional coincident stereo microphones. Another feature is that the angle between the microphones can be altered without affecting the stereo main axis. This is possible because the CMXY 4V microphones always rotate equally in an opposite direction to the main axis using toothed wheels at their bottom ends.

The CCM range has been expanded with the CCM-L fitted with a detachable Lemo connector.
Schoeps. Tel: +49 721 943200

Octavia goes 96kHz with dCS

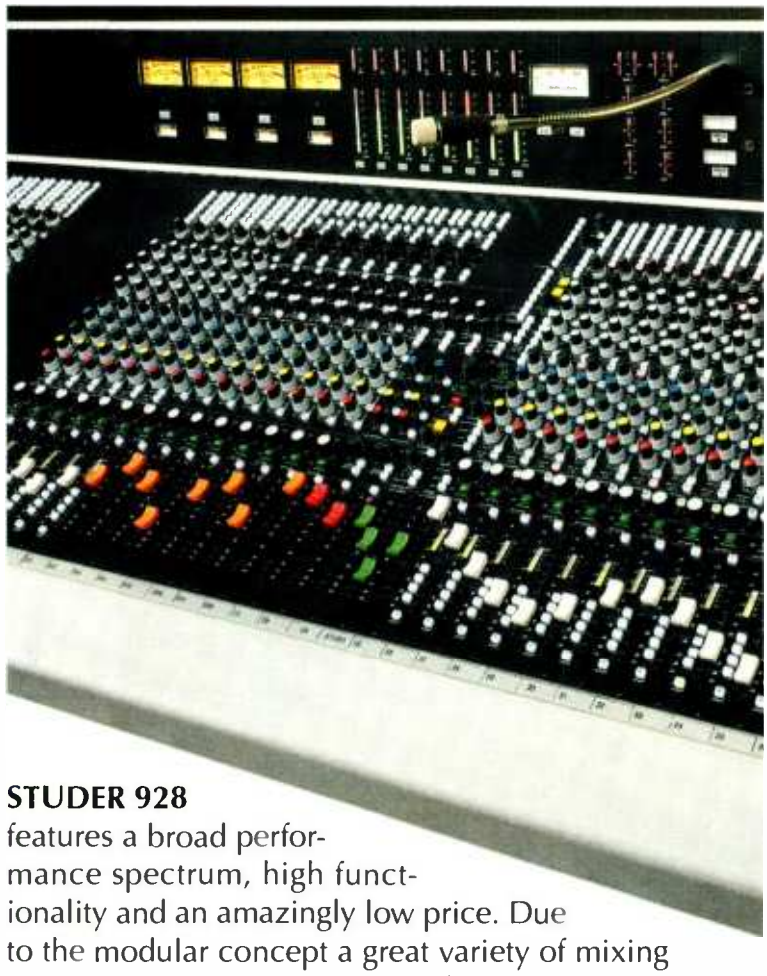
Studio Audio & Video has chosen Data Conversion System's 96kHz 24-bit convertors to complement the new high resolution editing features of its SADiE3 software. The dCS 902 and 952 ADCs and DACs have been designed for high speed operation, while a set of standard inputs and outputs enable them to be used in a wide range of digital audio applications. The units use a proprietary, multibit oversampling technique that achieves low differential nonlinearity and gain ranging is not used.
SAV. Tel: +44 1353 648888
dCS: Tel: +44 1223 423299

Dynaudio surround

The C4A is the culmination of page 36 >

Compact functionality – STUDER 928

Analog Mixing Console for various applications in broadcast, sound recording and theater



STUDER 928 features a broad performance spectrum, high functionality and an amazingly low price. Due to the modular concept a great variety of mixing console configurations is possible which means that it can be accurately tailored to your specific needs. Upgrading to surround sound is possible. **But there is more:** The STUDER 928 fulfills our uncomprising quality and safety standards. This is ensured by STUDER transformers at all inputs and outputs and high-quality VCAs.

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Focusrite Green Channel Strip

The latest item in this affordable coloured range, **Zenon Schoepe** investigates the direct to tape route

AFTER the initial surprise generated by the cosmetics of Focusrite's Green series processors I have to admit that the look of these units has grown on me. There's actually something quite chunky and workmanlike about the high density of controls that this Channel Strip presents; although the centre detented pots still don't quite manage to line up the caps' markers with the 12 noon position.

It would be tempting to think that Focusrite has simply repackaged and recombined existing processing sections from other Green units in this single channel device, but closer inspection reveals that it's not quite the case. While it's highly likely that the preamp section is extremely close to that found in the rest of the range, the EQ and dynamics sections are subtly different. For example, EQ is 4-band ±18dB with swept peaking mids, high and low shelves plus high and low pass filters, and

while this arrangement is similar to that found on the Focus EQ, the frequency ranges are certainly not the same. Low and high pass filters are 12dB/octave affairs covering 65Hz–25kHz and 15Hz–10kHz while the HF and LF cover 30–480Hz and 2.5–18kHz, respectively. The mids cover roughly the same range as the Focus EQ (40Hz–1.2kHz and 600Hz–18kHz), but are available on the single travel of the frequency pots rather than employing the multiplier-divider switches of the Focus EQ.

Q is switchable between 0.7 and 2.5 on the low mid and 0.5, and 1.1 on the high mid. The filters can be bypassed on a switch and inserted into the side chain of the dynamics section.

Dynamics are not a clone of the Voicebox, nor, for that matter, a direct lift from the dual channel Green Compressor-Limiter.

Again the approach is unique to the Channel Strip. You have the option of a gate or an expander with variable threshold and release (100ms-4s) plus a Fast attack switch that activates 70 microseconds response rather than the default 6ms. This section is bypassable separately as is the compressor which has Fast (1.5ms as opposed to the default 8ms) attack and Auto Release switches, variable threshold, variable ratio (1.5-10:1), variable release (100ms-4s) and a Gain makeup pot. Key input is also selected from this area. The input section can handle mic, line and instrument inputs with the selection of line or instrument instigated by an anticlockwise click on the mic gain pot. The instrument input is selected inconveniently on the back panel where you'll also find the high impedance jack socket and its selection is indicated by an LED on the front panel. Switches are also provided for phantom and phase reverse along with a centre-detented fine gain pot and an overall output level control.

Metering continues Focusrite's novel and useful implementation of combining bar graph VU display of the input in one mode with the simultaneous display of gate or expander activity and gain reduction in a dynamics mode. It sounds as if it would be confusing, in practice it really works well. Clever.

I have to rate the Channel Strip as a very rounded and complete single channel that is ultimately a little more flexible than the Voicebox, with the qualification that the Voicebox is more specific in what it aims to achieve. If you were to stereo link two Channel Strips together, which you can do with rear panel jacks, then this would amount to some powerful and slick processing.

The EQ really is excellent, less resolution admittedly than the Focus EQ offers, perhaps predictably, but still a good deal better than you'd likely get on anything less than a middle market board. Four good strong bands with the rare luxury of superb high and low pass filters. And it's a true all-rounder, and you can track and mix through it.

The dynamics are exceptional with so much variability and fine control afforded by the options of gate or expander selection and the various switches on the compressor plus internal or external key selection with or without the filters.

There are so many little twists and conveniences provided on this 1U box. Focusrite should develop a Green console based around this unit.

Mic and line performance is predictably sound, but, as with the Focus EQ, you have to wind back heavily on the gain when using a guitar input as this stage runs unusually close to the limits of the overload LED indicator.

What I like about the Channel Strip is that Focusrite has not simply transplanted sections from other Greens, but has instead paid attention to the arrangement and capabilities of the unit to give it its own personality. It's incredibly versatile, and as such is the outstanding processor in an outstanding range. ■

Contact
Focusrite Audio Engineering
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Fax: +44 1494 459920
Net: www.focusrite.com

NEW TECHNOLOGIES

< page 34 12 months research and development resulting in a 3 or 4-way system designed to work in any control room without the need for special flush mounting or acoustic redesign.

The system uses an acoustically optimised asymmetric enclosure and modular Active Bass Extension System (ABES) units which are controlled by an analogue, or optional digital, active system controller. This technique means that a fully modular system can be created to suit almost any size control room or SPL requirement.

Each system can be customised for individual studios with three amplifier and crossover options giving expandable power configurations from 1kW to 5kW. Maximum SPLs of up to 126dB can be achieved at the mixing position in typical rooms of up to 100m².
Dynaudio. Tel: +45 86 523411

Tape saturation

Complementing its digital red products, SPL's Machinehead digital processor is claimed to simulate the tape-saturation effects of analogue tape machines. The sonic effect is said to increase loudness, power and warmth. Intended for use with

solo instruments or complete mixes, controls are provided for input and output gain, drive and HF damping. I-Os are AES-EBU and SPDIF with wordclock, MIDI control and a computer interface.
Beyerdynamic. Tel: +44 1444 258258

Mandozzi Electronics

The ME-DART portable digital-audio recorder and editor has built-in codecs and ISDN terminal adaptor and is designed for use by broadcast journalists. It records 7, 10 or 15kHz, mono or stereo audio in linear mode without audio compression. Recording is done on laptop computer PCMCIA hard disks and the machine features built-in CCITT G.722 and ISO/MPEG 1 Layer II codecs as well as an ISDN terminal adaptor and transmits the edited contributions in real time via ISDN. It is battery operated and designed to be light, robust, splash-proof and easy to operate.
Mandozzi. Tel: +41 91 9452351

New cards for Akais

Akai has a number of card options for its DR8, DR16 and DD8 family of hard disk recorder-editors starting with the IB809E ethernet interface which allows control of the aforementioned units by the DL1500 system controller. The facility was previously only available on the flagship DD1500.

The DL1500 adds remote control of multitrack recording, enhanced editing, a precision jog wheel, multilevel Undo, a QWERTY keyboard input and control of on-board mixing functions. An LCD and page 38 >

AMEK RNCL

The Compressor

The tradition · The pedigree

System 9098
Dual Compressor/Limiter
by Rupert Neve the Designer

The addition of a new Compressor/Limiter to my SYSTEM 9098 product family is justified by the continuing popularity of the famous old 2254 devices I designed in the late 1960s. More than 25 years later, their performance undeniably still brings benefits to engineers and producers seeking inconspicuous control over the dynamic range of microphone signals. Just as importantly, they are used today in digital recording to manage critical levels, to preclude the effects of hard, unforgiving clipping and to impart warmth.

In those days, the Compressor/Limiter had to be almost all things to all men. Controls had to be accurately calibrated for the broadcaster and have the right subjective 'feel' for the music engineer. Attack and decay times, the rate of change of slope, the order of harmonics generated by the non-linear transfer characteristic etc. were arrived at empirically after a lot of listening with golden-eared people. The result was a Compressor/Limiter, the 2254 and its later derivatives, which sounded right and over the years achieved an amazing reputation.

The same principles have been applied to the new SYSTEM 9098 Compressor-Limiter. Considerable advances have been made in technology and I am now able to provide a much more flexible device which retains all of the character and musicality of the original design while incorporating some exciting new features.

Ratio, Threshold, Attack and Release are familiar controls with recognisable ancestry but an important new feature called Ambience has been introduced.

Operating the Ambience switch does not affect signals above the threshold but reduces or mutes signals below the threshold level. The effect is rather like a Gate but is much more subtle. Not only steady background noise but fluctuating ambience and apparent reverberation time can be reduced at will with the Gain control. For example unwanted environmental sound can be re-balanced, or even eliminated, from speech recorded out of doors. The Ambience control will also regulate reverberation - for example, a large reverberant studio can be made to sound like a small speech booth.

The 9098 Compressor-Limiter has a totally analogue signal path which employs transformers at both the input and the output. For the highest possible performance, input and output interfaces must be insensitive to anything other than the signal we want to receive - or there is little point in striving for excellence in the unit itself.

The heart of a Limiter or Compressor is the gain controlling device. The original 2254 used a diode bridge in a classic balanced ring modulator configuration. A very similar technique is used in the 9098 Compressor/Limiter except that semiconductor devices and amplifiers have greatly improved in the last 30 years. For example the original 2254 design had a noise floor of about -55 dBu. Noise performance of the 9098 unit is 35 to 40 dB better.

I believe that the new SYSTEM 9098 Compressor-Limiter continues the rich heritage of earlier designs and its flexibility and extremely high standard of performance will find many satisfied owners in all areas of audio production, whether recording, post-production, mastering or live performance.

Rupert

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TL Audio VP-2051

The growing use of MDMs has created a demand for well-specified outboard processors. **Tim Goodyer** evaluates a valve contender

YOU CANNOT BE SERIOUS about valve audio equipment and avoid TL Audio. You can be elitist, certainly, if your pockets are deep enough, but if you're looking to cash in on the cachet of the vacuum tube without busting the budget, you're going to have to weigh up the Indigo series sooner or later. And if you take into account the growing popularity of 'channel strip' processors —those that combine just about all the elements of voice processing in a single box—the VP-2051, although it's been out for a while, makes a timely study.

Having established its colour, its purpose, the nature of its circuitry and its general pricing, it should come as no surprise to learn that the 2051 is a single-channel unit incorporating a mic-line preamp, compressor and equaliser. It employs six valve stages: one in the preamp, two in the compressor, and three shared between the four EQ bands. The panel layout is uncluttered and holds no surprises, more a series of reassurances—running left to right, the sections follow the signal path and place the compressor before the EQ, but the inclusion of an EQ PRE button makes it possible to reverse these stages. The EQ and compressor can be switched in and out and, like all significant operational switching, their status is indicated by LEDs. Metering is via a peak indicator at the input and an 8-segment LED ladder which can be switched to display either the output level or the amount of compression. Compression is metered regardless of whether the compressor is switched in or out of circuit and reads in the same direction as output level, not with the scale reversed. To make this function a little easier to recognise, an orange LED indicates gain reduction metering.

The presence of an instrument input (on unbalanced jack) on the front panel is an indication of the thoroughness of the 2051's design. The rear panel reveals the expected balanced mic input and output XLR connections, an additional XLR to accept a balanced line-level input, side-chain insert on 2-pole jack, a link jack for stereo operation of two 2051s, and two further jacks for unbalanced input and output connection. Signal levels are pro +4dBu on the XLRs, and semipro -10dBu on the unbalanced jacks, with the output metering set to correspond to 0dB on both. Between these audio connections and metering alignment, the unit can be readily run in just about any conceivable setup. Compare this with, for example, the more costly Focusrite Green Voicebox where you'll find just two XLRs for audio: a balanced mic in and balanced line out.

Returning to the front of the 2051, we find that as well as level trim, the input section includes 48V phantom power, phase reverse and a 90Hz high-pass filter. The compressor is a friendly affair operating a soft knee with the threshold variable between +20dB and -20dB, ratio variable between 1:1.5 and 1:30, and gain make-up of up to +20dB. Compressor dynamics are limited to slow and fast switching of the attack and release—0.5ms or 20ms attack and 40ms or 2s release. While the declared purpose of the unit is for vocal use, I found this up to everything I threw at it without wishing for manual controls.

The EQ, meanwhile, offers 12dB of cut or boost in four bands—an LF shelf switchable between 80Hz and 120Hz; a lower mid band switched between 250Hz, 500Hz, 1kHz and 2.2kHz; an upper mid band switched between 1.5kHz, 2.2kHz, 3.6kHz and 5kHz; and an HF shelf switchable between 8kHz and 12kHz. Note the overlap and duplication of the 2.2k frequency. In operation, the EQ is smooth and ideally suited to enhancing a vocal, thanks largely due to the low (0.5) Q of the mid bands. You could find it wanting in more desperate situations—salvage jobs or difficult live situations, for example. The Green Voicebox EQ might have the edge in such cases as, although it has just one swept band, its frequency settings are not stepped and its Q is certainly narrower. Indeed, it's easy to make a strong case for the Focusrite unit in the live situation as it is smaller and lighter than the 2051 and isn't full of glass.

The TL unit signs off with its output section containing output attenuator and the meter. It also contains the STEREO LINK button which, with the rear-panel jack, allows the control voltages of two 2051s to be commoned for stereo operation.

I have heard it said that the valve owes its current popularity to Tony Larking—the 'TL' in TL Audio. I don't think I'd go that far, but it's hard not to recognise the appeal of the Indigo-series units. So the 2051 is probably better suited to life in the studio than life on the road, but then I tried it live and liked it, and it's now a regular part of Alanis Morisette's touring rig. Ultimately, everything it has is classy, up to, and including, the valves —which definitely give it a character over and above many competing units including its own Crimson solid-state stablemate. The 2051 comes out well featured, well voiced, well priced and well received. ■

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NEW TECHNOLOGIES

< page 36 large colour-coded keys are provided in addition to S-VGA display connection in a package that amounts to all the facilities offered by the DD1500. Software permits networking of six DD or DR units for a maximum of 256 tracks.

The IB808G general purpose I-O allows the custom application of any remote control system using GPIs. A total of 16 GPI inputs are provided for track arming and transport functions while 12 GPI outputs can be used for such things as reading record tallies.

The company has developed an 8-in/16-out ADAT interface for the DR16, which has applications for users of Yamaha's 02R desk, an 8-channel AES-EBU I-O board and an 8-channel TDIF IO board.
Akai UK: +44 181 897 6388

A&H 8-bus

Allen & Heath's WZ20:8:2 8-track recording console has stereo cue and effects routeing plus a mixdown switch that reconfigures the multitrack signals to the channel facilities.

It has 8 mic-line inputs on balanced XLRs or jacks with inserts. Each of the six auxes are on individual controls and EQ is 4-band with two sweeps. Six stereo line inputs have 2-band EQ and full aux access and 100mm faders are used throughout. Additional features include an oscillator and talkback.
Allen & Heath. Tel: +44 1326 372070
E-mail: 106030.1426@compuserve.com

Libra Live

The Libra Live digital console has been specifically designed for broadcast production, but is based on the Libra music recording desk architecture. Using the same processor and I-O hardware as Libra, dynamic automation and music recording controls have been replaced with live production functions such as the ability to switch in a backup microphone. Input channels can route to console-wide output, auxiliary, mix-minus and subgroup buses. VCA-style groups can also be created and each input channel, group or output can be configured with filters, EQ, dynamics and inserts in almost any order.

New features for the 55 Series console include VCA faders and input preselectors.

The VCA fader system now provides 8 group master faders to supplement the 8 audio groups which the 55 desk may already be fitted with. The input page 40 >

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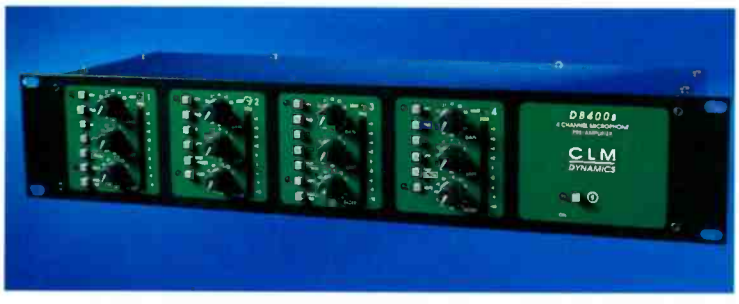
CLM Dynamics DB400s

Championing M-S has given a new name to mic preamps, a welcome 'in'. **Dave Foister** discovers tricks up its sleeve

YOU WOULDN'T THINK there could be many new twists left for microphone preamp designers, unless someone's planning a hybrid clockwork/steam-powered one. CLM Dynamics has managed to spot a gap, however, and has produced a box of preamps with one particular feature aimed at the revitalised interest in M-S techniques.

The DB400s starts as a neat halfway house between the fully-featured mono or stereo units and the 8-knobs-in-1U tweezer-operated multipacks. It has four complete preamps in 2U, all with a surprising amount of adjustment capability and metering. Three knobs and six switches on each provide everything expected of a preamp and a little bit more, while a seventh button on two of them takes the extra step into M-S.

In conventional operational mode, all four preamps are completely independent. Each has switchable phantom and phase, and a 20dB pad before the continuous gain pot.



There is an 80Hz 24dB per octave high-pass filter, and a rotary output 'fader'. Every channel has a limiter, with variable threshold and a switch to select fast attack. Limiter action is shown on an LED, and I found it to be reliable and unobtrusive.

Every channel has its own meter, a vertical LED strip, that shows levels after the limiter but before the output fader. This is particularly bright and easy to read; although it could do with going a little higher than +9dB; driving it straight into the input of a digital recorder meant that levels were constantly at, or near, the top of the scale even though the signal was perfectly clean. The mute switches on each preamp are helpful in this kind of application, where the preamps take the place of a mixer between microphones and multitrack. A final switch, hidden on the rear panel, pads the output down by 14dB to allow for connection to semi-pro equipment (apparently overlooking the fact that the difference between +4dBm and -10dBu is not 14dB). The inputs and outputs are balanced on XLRs, and there is also an unbalanced insert point on a TRS jack. A single overall signal ground lift switch is provided, which is not as useful as individual switches. In fact I had problems in my notoriously hum-loop prone studio with a valve microphone on the far end; this, of course, was plugged locally into the mains and hummed merrily whichever way the DB400s' switch was set.

Mid and Side microphone techniques have always had their adherents, and the advantages for mono compatibility, particularly important in broadcast, are readily acknowledged. Despite this, enthusiasm for M-S has generally been confined to a small minority of devotees—until, that is, the advent of stereo location sound for pictures. Suddenly the need to control a stereo image to match a picture you couldn't see using a monitoring system you couldn't hear meant that a technique that allowed such decisions to be made in post-production was a godsend. This was the hidden bonus of M-S: if the middle and side signals—a front-facing microphone of suitable pattern and a side-facing figure-of-eight—are recorded raw on a 2-channel recorder, the width of the resulting stereo signal can be determined at a later date.

The DB400s caters more for those who wish to employ an M-S configuration and convert to stereo on the spot, and allows two M-S microphone arrays to be set up independently of each other at the same time. Each pair of channels can be switched to M-S decode mode, where an incoming M-S signal is converted to left-right stereo with the image width controlled by the S channel's gain control. For reasons that aren't clear, the resulting left signal appears at the even numbered output with the right channel at the odd one; there would appear to be no reason for this illogicality, as a simple phase reversal of the S signal would correct this.

Still, it works, and provides a simple means of doing the job. CLM doesn't miss a trick, however, and has spotted that the presence of two sum and difference matrices means the two pairs can be chained together to form a general purpose stereo width control, for which detailed instructions are provided. Since the same process converts L-R to M-S and vice versa, the first two channels are used to derive sum and difference signals from a stereo source, which the second pair then converts back to stereo with the width controlled again by the S level. M-S fan Mike Skeet is credited with help in figuring this out, although the principle is pretty straightforward given the right connecting leads.

All this is icing on an already worthwhile cake; the DB400s' package of four high quality versatile preamps is attractive in its own right, with many potential applications in and out of the studio. ■

Contact

CLM Dynamics, 26 Tay Street, Monifieth, Dundee DD5 4BG, Scotland.
Tel-fax: +44 1382 534 868.

NEW TECHNOLOGIES

< page 38 preselectors constitute a snapshot automation system which controls the mic and line level inputs to the console.

A new addition to its family of digital audio editing systems, the AudioFile Prolog is an 'attractively priced' 16-bit system that includes integral machine control plus a range of options, including future upgrade to 24-bit.

Features include up to 24 'virtual' tracks, user-defined graphic presentation of 16 or 24 tracks, and 16-track audio outputs
AMS Neve. Tel: +44 1282 457011

Taking the mic

Since the launch of the MCD 100 digital condenser microphone further changes have resulted from discussions with end users and digital product manufacturers.

The operating principle remains the same, using the cardioid MC 834 microphone system, an impedance convertor, an ultra-low-noise op-amp and an internal 22-bit AD convertor, but a new output format is being employed featuring conventional XLR connectors for AES-EBU output.

New from the manufacturer is the MCE 7 omnidirectional miniature clip-on mic, which is aimed at the theatre and broadcast market, and is claimed to be one of the smallest transducers in the world.

The MCE 82 is a stereo condenser hand-

MCE82

held mic designed to operate in XY stereo via phantom power or internal 1.5V battery. Targeted at ENG-IFB reporting it has a built-in shockmount and rumble filter.
Beyerdynamic. Tel: +49 7131 6170

SCSI for DMT-8 VL

The Fostex DMT-8 VI has added the 8339 SCSI interface card which is available fitted on new machines or as a retrofit. The expansion card offers fast backup and restore of recorded data to external SCSI devices such as removable cartridge systems and external hard drives.
SCV, London.
Tel: +44 171 923 1892

page 42 >

When *"good enough"* isn't good enough look to Harrison

Find out why the world's premier producers, engineers, film re-recording engineers, broadcasters and post-production professionals are flocking *en masse* to Harrison. For some, it's the incomparable sound quality behind history's biggest records and film sound tracks. For some, it's the rock-solid reliability, being able to run day after day, week after week, month after month, year after year without worry. Still others have come to rely upon the world's most comprehensive dynamic total automation system, another Harrison invention. They own Harrison SeriesTwelves - the most sophisticated fully automated mixing console in the world.

Keep in mind that before they all switched to Harrison, they used to own something else.

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Amek 9098 Dual Mic Amplifier

The 9098 range now has an MS-capable dual-channel preamp. **Zenon Schoepe** reports on Amek's new front end

THOSE WHO HAVE explored the original Amek Rupert Neve-designed Amek outboard unit, the 9098 EQ, will remember that one of its strongest points was a rather fine mic preamp. Although this could be used in isolation from the device's elaborate equaliser, the unit could handle just one channel at a time. Recognising the opportunity, Amek has released the Dual Mic Amplifier that combines two preamp sections drawn, like the 9098 EQ, from the lineage established by that mother of all analogue boards the 9098. The DMA also includes a couple of extras, most notably in how it approaches stereo mic processing and also includes, for the first time on an Amek unit to my knowledge, front panel DI inputs.

Rear panel connections are on balanced XLRs while the DIs enter on jacks which, incidentally, can also be run balanced. Each channel has a switched gain pot and a detented ±6dB trim pot running a range of -6dB to +72dB. All switches have associated LEDs and these cover phase invert, phantom power, a 120Hz high-pass filter plus a MUTE switch which, while it kills the channel's output, does not affect the action of an 8-segment bar graph meter which follows output level.

Not content with two channels of quality preamplification, Amek has gone a little further with the inclusion of MS processing. This offers input and output MS matrix switches and a switchable and centre-detented WIDTH pot which ranges from monoing the stereo signal to widening the stereo image—although the latter does introduce extra out of phase information. These sorts of stereo facilities are convenient, but frequently left off multichannel mic preamps.

You feel you want to like this box as soon as it powers up. Build-wise the DMA is finished in 9098 livery and stinks of quality with its expensive-feeling pots and distinctive long-travel push-buttons. There are LEDs for pretty much all the switches including the MUTE switches which are especially useful for those times when you want to change a mic's pattern, or do anything else that could cause a thump.

These are two premium quality mic preamps—silent, open, airy and fast—and the channels will push out a ton of level given an obliging transducer and a snare drum hit with a vengeance. They're good all-rounders, as comfortable with delicate and room-orientated sources as they are with vocals and general miking duties.

The fact that there are two of them means you can take in stereo and even play around with MS as an exercise. However, the lack of an output level control necessitates a little more caution when recording directly to digital media.

I have to admit to feeling a certain scepticism towards the inclusion of DI sockets on this unit—Rupert Neve, the grandpapa of the mixing console, and the electric guitar did not sit together comfortably in my audio universe. But I was surprised with the reality; plug in your guitar and it's instant new strings. This can only add to the versatility and worth of the DMA which does considerably more than just bringing a mic level up to usable.

If you are convinced you could do with some outboard preamps—because you don't think you have a desk of great distinction, don't have a desk at all or because you just fancy some—then investigate these. This is particularly the case if stereo is a dimension you're interested in and if you want the prestige of a 'name' brand unit.

The DMA is a well-featured, high-quality unit offering outstanding performance. ■

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NEW TECHNOLOGIES

Dialog4 hails its new miniature MPEG recorder

Dialog4 has extended the choice of location reporting recorders with the C(Centronics)-TAXI which measures 140 x 200 x 40mm. Using Flash-EPROM, the machine can record 40 minutes of stereo with ISO-MPEG. Material is stored in Broadcast Interchange Format (BIF).

Offering mic, line and headphone connections, record parameters like level and limiter settings can be locked in during operation, while an index key allows access to different takes. Finished material can either be transferred as a file or in real-time using the Digital Audio Library interactive program.

Dialog4. Tel: +49 7141 22660
E-mail: dialog4@proaudio.de

Crown adds SLM

The IQ PIP SLM module is designed to provide full audio system control and load monitoring for IQ PIP2-compatible amps. With a built-in sine wave generator and data acquisition system for the measurement of

load impedance and test voltage levels for real-time testing, the unit fits onto the back of amps without extra cabling and connects via the Crown Bus port.

K series amps are now available in 13 different colour finishes.

Meanwhile, Crown has introduced the PCC130SW phase coherent cardioid mic which is a miniature version of the PCC170SW. It can be programmed for touch on-off and a high intensity LED signals status. Capable of withstanding 120dB without distortion, the mic's electret-condenser capsule has a frequency response of 50Hz to 20kHz.

Crown. Tel: +1 219 294 8000

newt add ons A–D preamp

Audio and Design's MIDA System 4 dual-channel mic preamp with 20-bit A-D convertors and optional remote control is designed as an accessory to the company's DMM1 digital mixer. The RS484 control permits gain and phantom power alterations to be made remotely.

Analogue inputs are electronically balanced and mic input levels can be calibrated into the A-DC while aux analogue outputs are available for local foldback or headphones monitoring. The **page 44 >**

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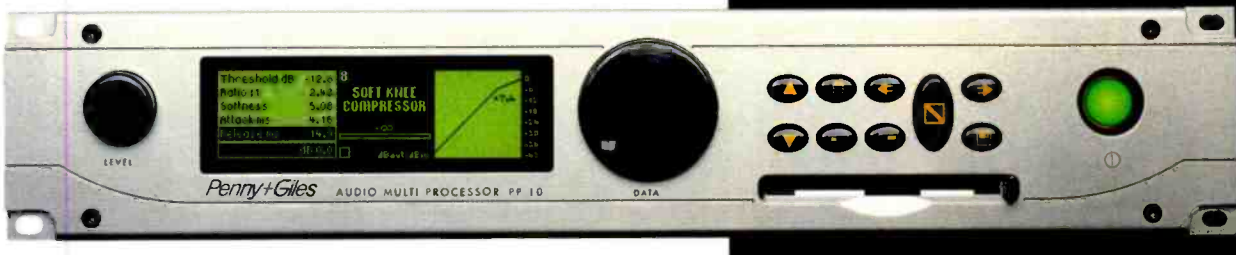

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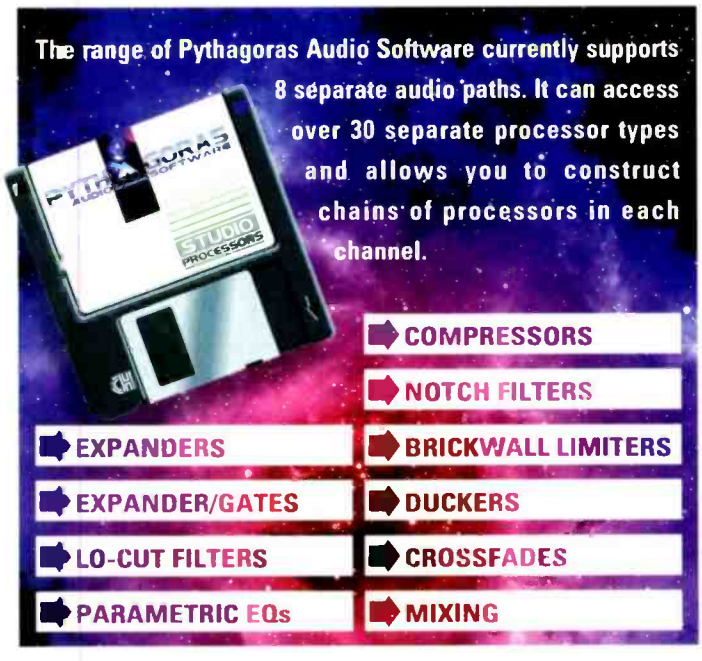

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Martech MSS-10

Martech's new mic preamp has just three things going for it: class, class and class. **Dave Foister** proclaims it a winner

FOR A MICROPHONE preamp to be worth having, it must offer something extra compared with a typical console input stage. Some add a specific character: some additional facilities, building towards the channel strip in a box concept. For some, the USP remains the sheer quality of the audio path, trading on the assumption that most console preamps are less than ideal and therefore compromise your signal before you've even started. New to this camp is Martech's MSS-10.

Its intentions are clear from its outward design, which is simple, uncluttered and stylish. And it looks expensive, which it is. The shape suggests that it constitutes a module for a racking system, and indeed it is descended from Martech's Mic Pre 1.0d, which with the Equaliser 1.0d formed the basis of just such a range. Here the presence of a big carrying handle on the top and the absence of the original modules' rear-panel linking facilities suggests that the MSS-10 is intended for stand-alone use, although a rack for four such modules is available (free with four MSS-10s, and I should think so too).

A huge vu meter dominates, viewed through a shaped and contoured window in a chunky front panel. Two rather elegant knobs control levels, one a switched preamp gain setting, and the other an output fader. These are flanked by six push-buttons in what looks like frosted acrylic, all of which illuminate from within when engaged, some green and some red. The usual preamp facilities are provided by the left-hand three—phantom, 20dB pad and phase reverse. The others show that all is not quite as simple as it seems: the unit also has a line input, which can be connected directly to the line driver and controlled with the output fader. Unusually, the mic preamp output is still available on the back when the unit is being used in this mode, tapped off before the line driver. A CAL switch bypasses the output fader, and there is a mute switch on the output, which I found particularly useful when using the preamp to feed a multitrack direct.

That's it for facilities: a little more than the bare minimum, but not much. Clearly then, the appeal of the MSS-10 must lie in its quality, and when such a simple unit carries such a substantial price tag it's more tempting than ever to lift the lid and find out what you get for your money. What is revealed is a joy to behold, a superbly designed piece of equipment built to the highest audiophile standards. The earlier Mic Pre 1.0d was available in both solid-state and valve versions, and although Martech stated a preference for the valve it also recommended its solid-state circuitry to the valve enthusiasts for its musicality.

The MSS-10 here is strictly solid-state. The basic building block is an all-discrete 14-transistor amplifier board notable for the presence of a yellow LED, invisible from the outside. This is in common with some other high-end designs I have seen, where LEDs are used in preference to conventional Zener diodes simply because their noise performance is better. Four of these boards are used as gain blocks, two in the mic pre and two in the line driver. Other signs of the audiophile approach are electrolytics with Martech printed all over them and a huge can for the input transformer, which again is a custom Martech component produced after careful comparisons of classic transformers and based on a design which had not been made for 25 years. Other notable touches include the obvious high quality of the PCBs and the meter illumination, which instead of filament bulbs uses 15 yellow LEDs arranged around its edge. The visual effect is the same, but presumably it will never pack up.

Indestructible meter lights apart, the thrust of all this careful and expensive design is the sound. The specs themselves are impressive, with figures little worse than a piece of wire, but Martech clearly intends the performance to be judged by the ear rather than the meter. Nonetheless, the line driver's claimed frequency response from 10Hz to 200kHz (sic) within a quarter of a dB is worth mentioning.

But the sound alone is what Martech expects to justify the asking price, and it would certainly seem to do so. Rarely will you hear such a quiet preamp, or one with such a complete frequency response and low distortion that it seems to open a window on the microphone. The MSS-10 is therefore doing exactly what any preamp should attempt to do; it is delivering the maximum that the microphone is capable of producing to whatever follows. This is certainly what I want from a preamp, and the Martech comes as close as any I have heard to achieving it. ■

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NEW TECHNOLOGIES

< page 42 front panel has phantom power switching for each mic and up/down gain adjustment plus a ganging mode for stereo mics which allows relative gain to track on each channel. A 9-pin D-sub GPI is provided to activate cough muting.
Audio and Design: +44 118 984 4545
Email: sales@adr.tcom.co.uk

02R surround

Danish company SoundWare has developed the SP02 surround joystick for the Yamaha 02R which with V2 software for the desk allows the group outputs to be used instead of the auxes. Formats supported include 4-channel LCRS, 5.1 channel, 6-channel, 7.1 and 8-channel formats.

For theatre and live applications, custom software can handle 14 surround channels with constant power correction.
SoundWare: +45 8613 2400
Email: soundware@post1.tele.dk

Soundcraft Series Five

Following numerous clandestine appearances at various trade shows, Soundcraft has formerly announced its new FOH desk, the Series Five, which the company claims is its finest to date.

Available in sizes from 24 to 48 mono channels with an additional 4 mic-line stereo inputs, the desk has 8 groups, 12 auxes, sweepable high and low pass filters, LCR panning, 256 MIDI snapshot memories, 10 matrix outputs and 10 VCA groups. It also has a new power supply with a 3-year warranty.
Soundcraft. Tel: +44 1707 665000
Net: www.soundcraft.com

TLM50 returns

The Neumann TLM50 omnidirectional small diaphragm condenser is back in production and replacement capsules for it and its valve predecessor the M50 are once again available.

The mic's pickup pattern provides a smooth rise in frequency response with a related increased degree of directivity in the upper frequency range, akin to the performance of a pressure gradient mic, while in the lower frequency spectrum it performs more as a pressure transducer with a linear response down to low frequencies.

The TLM50 is well suited to providing a principle stereo signal when used in AB stereo configurations. A -10dB pad and high pass filter are located at the base of the microphone, the latter allowing the attenuation of frequencies below 100Hz or 30Hz.
Sennheiser: +49 5130 6000.
Net: www.sennheiser.com ■

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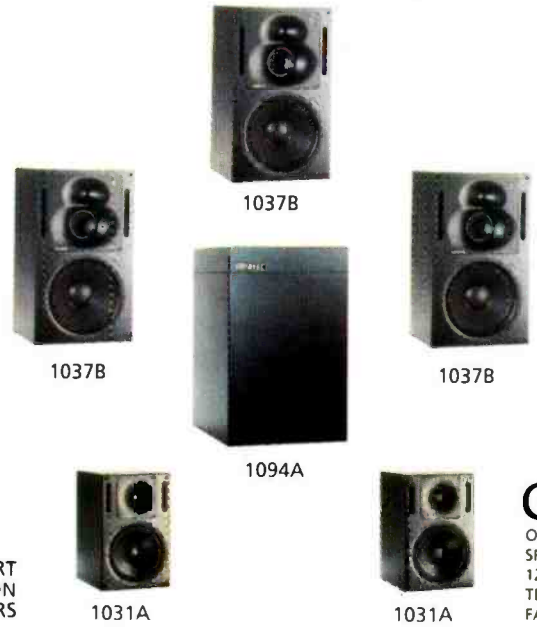

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JECT AUDIO

Mic Preamps

The microphone and it's means of amplification is enjoying renewed and worthy interest. **Studio Sound** polls its reviewers for their favourite mic preamps looks at the selection on offer

DESPITE an alarming period where it seemed that mixing consoles were doomed to become line input only devices, microphones have bounced back and enjoyed a resurgence in interest. Indeed mic awareness it at something of high. With this has come a commensurate rise in interest in the specific means of their amplification and this has produced a wealth of outboard preamps that attempt to prick the conscience with matters of quality. These units would like to suggest that their performance is better than that of the average desk's amps and users clearly like the freedom to adopt the mix and match approach of employing valve or other specific circuitry to impart a variety of flavours at the mic stage.

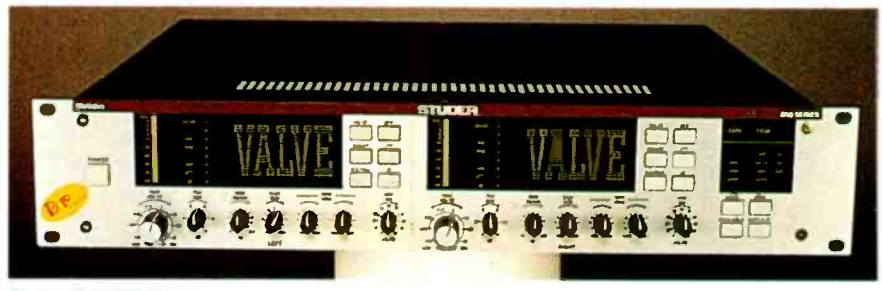
Owning a wide selection of outboard pres is an expensive proposition particularly as the manufacturers have almost invariably targeted the deluxe sector of the market. However, owning at least a couple of channels of outboard preamp for special occasions is now commonplace in recording, post and broadcast facilities.

One of the most interesting developments is the advent of large numbers of dedicated "voice channel" processors that combine mic pres with EQ and dynamics sections. Devised originally for users running hard disk workstations who had a need for a single channel of processing with which to input audio in to their systems, these boxes have now been recognised as good outboard supplements for all environments.

Here are the comments of the *Studio Sound* reviewers' jury

George Shilling: The GML Microphone Preamplifier comes as a 4-channel unit, with a simple and straightforward switched gain control for each channel. George Massenburg equipment is always very pure sounding, and I use these for vocals and other instrument overdubs, especially when working on SSL E-series or G-series consoles. I recently recorded The Wannadies' vocals and some other instruments (such as melodica) with GMLs-they gave a superb smoothness when combined with the right mic, (a U67 in this case) and a good old Urei 1176. A natural sound was achieved without EQing: just a little HF was added to the vocals at mixdown.

The Focusrite ISA 85110 is the original, superb Focusrite EQ and mic-line amplifier. I recently used one when recording vocals with Teenage Fanclub's Norman Blake using an AKG C12VR reissue, and Raymond McGinley, using an SM57. The difference between the Focusrite and the very high-quality, clean-sounding SSL 8000 G Plus was page 48 >



Studer D-19 MicValve

Summit MPC-100A

Millennia Media

TL Audio Dual Valve Mic preamp/DI


Drawmer 1962

FM Acoustics ClassAmp M1

< page 47 surprising. There is a definite character to this preamp, a certain dryness with a diamond sparkling top end. The EQ is neat and precise-sounding, with nice big smooth boost-cut controls, although I have to say I am not a fan of x3 multiplier buttons for frequency select.'

Dave Foister: 'I've been fortunate to use a lot of different microphone preamps in the course of reviewing them for *Studio Sound*, and strangely it's often the simplest ones that have left the strongest impression. It surprises me how many people are prepared to dismiss quality outboard preamps, despite having no hesitation in using special equalisers or dynamics in preference to those in their consoles. There can be little doubt that a well-designed high-end preamp is just as likely to be an improvement over the couple of op-amps in the desk as the fancy EQ is. For me the experience of hearing the right microphone fed to a tape machine via a grown-up quality-conscious preamp is the kind of thing that makes my job worthwhile and the hairs on the back of my neck stand up.

'And I have to say that one of this month's review items had just that effect. The Martech MSS-10 is exactly the kind of preamp I like, an audiophile design with neither frills or compromises. Using a good microphone with a preamp like this is like fitting the best possible lens to a camera, and reveals just how much of the raw material we work with gets lost through anything less.

Preamps in a nutshell

John Watkinson outlines what is and what is not important in a mic preamp.

IN A SENSE, the quality of the audio signal is determined by the mic preamp because it is the first electronic stage in the chain. Getting things wrong at this point means there is no hope of putting them right later. While consoles almost always contain preamps and appropriate routing, this often results in a lot of circuitry and switching between mic and recorder. An alternative approach is to achieve the simplest and cleanest path from mic to tape and this requires an outboard preamp. If the recording is digital then a preamp with integral ADCs, or an ADC with mic inputs, should be considered.

Mic preamps exist because microphones sacrifice efficiency in the interests of quality. Microphones can never be efficient because the diaphragm will always be heavier than the air trying to move it. A microphone that has to produce a large signal will need a large diaphragm and this will result in directivity problems at high frequencies. The ideal microphone will have a directivity pattern which is independent of frequency and this requires a very small diaphragm and results in a small output signal. Consequently one the main issues in a mic preamp is the noise level.

Microphone signals potentially have the greatest dynamic range in the audio chain because the microphone is exposed to real life sounds prior to any gain control or compression stage. Thus in addition to low noise a microphone input must accept high levels without distorting. The problem is compounded by the wide range of sensitivities that different microphone models display. A preamp optimised for a certain microphone would give an ideal balance between noise and overload. Such a preamp would appear noisy with a lower sensitivity mic, but would appear to distort prematurely with a high sensitivity mic.

The practical solution is to incorporate a sensitivity control into the mic preamp so that dynamic range of the preamp can track that of the microphone. For a given microphone, the control would have one optimum setting.

A simple potentiometer between stages is not a solution as this leaves the input stage unoptimised. A measure of the engineering quality of a preamp is the way in which the dynamic range is preserved over the full range of the sensitivity control.

Device noise is not the only issue. The preamp and the mic are often remote from one another and connected by a balanced cable. The ability of the input to reject interference has a significant effect on the in-service noise performance.

The user's expectations of dynamic range continue to increase with the development of ADCs with longer wordlength, yet the electromagnetic environment continues to deteriorate. A quality unit will have addressed electromagnetic compatibility issues as part of the design process. It will have metal XLR connectors with pin 1 and the body firmly bonded to the metal case so that screen currents are shunted harmlessly away.

Measuring the common mode rejection ratio within the audio band is only part of the story as this tells nothing about the sensitivity to RF. Unless the two input pins have absolutely identical impedance at all radio frequencies RF rejection will suffer. If the impedance is different the interference undergoes mode conversion and appears as a differential signal which is amplified. The balanced transformer input has a lot going for it as it can offer extremely good RF rejection. With the very low signal levels involved hysteresis distortion is unlikely to be significant.

Generally poor RF immunity is due to poor layout and poor layout is also one of the dominant causes of distortion. Consequently a unit which has poor RF rejection will probably not sound very good even in the absence of interference. Incidentally with the incessant growth of RFI the use of star-quad cable is a useful extra weapon. In theory and in practice the geometry of star-quad causes the interference picked up on the two signal legs to be more similar than is the case with ordinary twisted pairs. This gives better cancellation at the differential input.

Interference can also enter via the AC supply and it is important that supply is suitably filtered. Safety is also paramount because nearly all microphones will be used hand held at some time. With a metal cased preamp it's easy to prove that the case and the cable screens leaving it are securely grounded so that any fault won't use the performer as a fuse.

The modular mic amp has the advantage over a console-based amp that it can be placed near the mic so that the longest cable run is at line level. The integrated mic preamp and ADC has a noise advantage because the digital signals it outputs are much more noise immune than analogue signals can be.

In my view a good mic amplifier should not have a 'sound'. It should simply make the differential input bigger. The frequency response should be flat and the transfer function should be linear and phase linear. Such a unit can then be used with absolutely any kind of subject matter. If a particular 'sound' is needed it can be added with an effects unit. If the preamp causes an effect it's impossible to switch it off and it puts a footprint on everything that is recorded with it.

The mechanical aspects of a mic amp are worth worrying about. For economic reasons, most units use printed circuit-mounted connectors to eliminate hand wiring. This is fine provided the case is rigid enough to prevent movement between the PCB and the case. If the case is flimsy the connector solder joints are stressed and eventually become intermittent. There is a law that says they finally go open circuit in the middle of a session. □

The same is true of the enormous DW Fearn VT-1, another single-channel straightforward preamp, this time using valves. Despite appearances, I'm no placard-waving valve fanatic, but it does sometimes seem easier to do something special with valves than with solid-state circuits, particularly around the mid-price range—look at TL Audio. The Fearn sits considerably higher up the budgetary ladder than that, and is a shining example of what can be achieved in terms of musicality and transparency. Having said that, the Martech shows that if the same kind of care and dedication is lavished on a box of transistors the result can be equally special.

I very much enjoyed Amek's RCMA system, partly because of the huge versatility and control, and partly because the sonic page 50 >

Demeter

< page 49 performance remains the number one priority—it is, after all, the reason for the remote control facilities in the first place. The combination of the eminently practical remote control concept, the solid engineering and Rupert Neve sonic standards made the Amek system something I was more than usually sorry to part with.

'Among all the preamps with extra processing in them, a format becoming ever more popular, the Studer D19 MicValve stands out. As ever, all depends on the quality of the initial preamp, which in the case of the Studer is never in doubt. The surprise inclusion of a valve-based enhancement section could have swung it either way for the D19, but I found it hugely enjoyable and capable of very distinctive yet subtle results.

'Last but not least, I've always found John Oram's designs compulsive listening, and the Oram Sonics Microphone Work Station warrants inclusion here. Its microphone preamps have an extra something, in common with all the others on my list, and the addition of the simple, beautifully sweet EQ makes it a joy to use.'

Zenon Schoepe: 'Although I've never been able to get on with its peak LED indication, I'd have to mention TL Audio's Classic range dual mic preamp/DI for solid performance at an attainable price. Combining valves with solid state circuitry, the company's range of boxes has given tube character to more people than any other brand. It's the same preamp as that used in the other units in the range and while it won't make a lousy mic sound like a piece of vintage exotica, it will make a decent mic sound special by balancing thickness on the gain pot.

'Demeter's VTMP-2b, the version of the old standard which received a pad, low cut, phase reverse and 10dB extra gain, employs fancy transformers, capacitors and resistors and adds up to a dual-channel 2U with a decidedly high-quality feel and enormous rotary faders for fine gain. It's also got one of the nicest clean electric guitar DIs I've heard. Mic performance isn't as soft and muddy on the front end as some tube gear can be. Strangely I like this because it sounds quite contemporary.

'A bit of an odd one this, but Night Technologies' PreQ3 is worthy of mention for being different. It's a modular 1U rackmount that can house up to four channels and each has the company's AirBand five-frequency boost-only shelving EQ which the incoming mic signal hits before it enters the unit's gain circuitry. This is unusual and reveals that processing at this stage allows mic characters to be altered in a way that is distinct and legitimate.

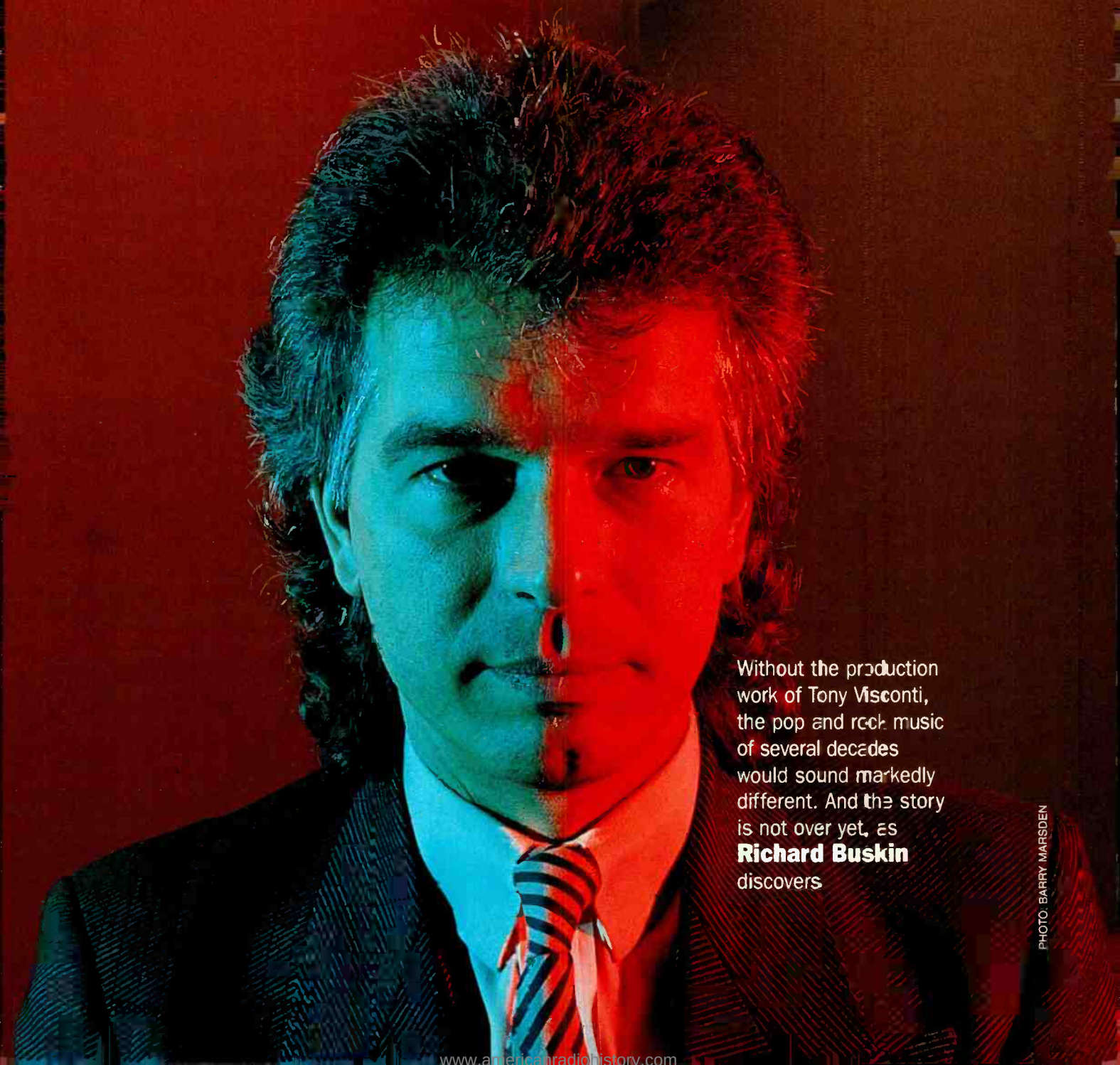
'If I was restricted to choosing only one preamp then the Studer Micvalve would be the one I'd go for. For starters it has valve circuitry that can be varied with subtlety but quite fundamentally from front panel pots. It produces a wonderful sound on mic and line signals plus you can additionally bypass the valves completely to give the best of both worlds. Best of all you can fit MDM digital interfaces and route the two channels individually to the eight buses. It's so self-contained. Ingenious.' ■

Alice
Mic-Amp-Pak Single and dual-channel mic preamps
Amek
9098 DMA Dual-channel mic preamp
9098 RCMA Remote-controlled dual-channel mic preamp
AMS Neve
1081 Modular mic preamp
Anthony DeMaria Labs
C/L1500 Dual-channel valve mic preamp
AT!
8MX2 8-channel mic preamp with stereo mixer
Aphex
107 Dual-channel Tubessence mic preamp
Avalon
AD U5 Single-channel line amp-DI
beyerdynamic
MV100 Dual-channel battery powered mic preamp
Canford
Mic Preamplifier Dual-channel mic preamp
MS Mic Preamplifier Dual-channel M-S mic preamp
CLM Dynamics
DB400s 4-channel M-S mic preamp
Crookwood
Paintpot Dual-channel remote-controlled mic preamp
DACS
MicAmp Dual-channel mic preamp
dbx
dbx 786 Dual-channel mic preamp
Drawmer
1962 Dual-channel valve preamp with A-D convertor
Demeter
VTMP 2B Dual-channel valve mic preamp-DI
HM1 Dual-channel valve mic preamp-DI
DW Fearn
VT-1 Single-channel mic preamp
VT-2 Dual-channel valve preamp
Event
EMP-1 Single-channel mic preamp
FM Acoustics
Class Amp M1 Single-channel class-A mic preamp
Focusrite
Red 1 4-channel preamp
Red 8 Dual-channel preamp
Green 1 Dual-channel mic-line preamp
LA Audio
MLX2 Dual-channel preamp-DI
Manley
MP40/1 Single-channel valve mic preamp
MP40/2 Dual-channel valve mic preamp
MP60/1 Single-channel valve mic preamp
Martech
MSS-10 Single-channel mic preamp
Millennia Media
HV-3 Dual-channel mic preamp
MTA
Intermix 16-channel mic-line preamp
Signature Series 4-channel mic-line preamp
NTI
PreQ3 Multichannel mic preamp
SPL
MikeMan Dual-channel mic preamp
Studer
Studer D19 MicValve Dual-channel valve preamp with A-D convertor
Summit
TPA200B Dual-channel valve preamp
Symetrix
SX202 Dual-channel mic preamp
Tascam
MX4 4-channel M-S mic preamp
TL Audio
2001 4-channel valve preamp
Classic Dual-channel valve preamp-DI
Tube-Tech
MP1A Dual-channel mic preamp
Yamaha
HA8 8-channel mic preamp

Focusrite Red 8

Interview

Product Perfect



Without the production work of Tony Visconti, the pop and rock music of several decades would sound markedly different. And the story is not over yet, as **Richard Buskin** discovers

PHOTO: BARRY MARSDEN

IT'S EASY to listen to one of Tony Visconti's productions and not hear him. On this basis you can, if you wish, choose to discount him as a producer of note. Alternatively, you can regard it as an example of the most seamless production work around.

'People say that working with me is effortless, and I think that's an important factor in record production,' is how Visconti himself explains it. 'I am transparent in the studio and everything about the artist is fully realised on tape.' And he's right, of course—listen to T Rex, David Bowie, Thin Lizzy, Iggy Pop, Joe Cocker, Paul McCartney, U2, The Stranglers and The Moody Blues for confirmation.

During the 1970s and 1980s Tony Visconti produced and engineered for an eclectic array of rock icons, and to each project he brought his technical know-how, sharp ears for arranging, and ability to incite the best performance out of the available talent.

'There's nothing standard about my approach,' he says. 'Every time I arrange for an artist it's a unique situation.'

As indicated by his distinctive mid-Atlantic accent, Visconti is originally a native of New York but spent several highly successful years living in London—23 years, in fact. He made his musical entrance here playing with a band by the name of Mike Dee and the Dukes, made his first recording at the age of 15 and, on leaving school, worked in A&R at RCA. His first ventures behind the console occurred in the UK during the late-1960s, when he worked for Essex Music and produced artists such as Cocker, The Move and Procol Harum. Then, in 1974, he formed his own company, Good Earth Productions, and bought a 16-track facility in West London, before upgrading to Zodiac Studios in Soho a couple of years later and also renaming that Good Earth. He would remain there until 1989.

'I got really pissed off with being in the studio business,' Visconti now admits. 'When I started off with the SSL board there were only about two or three SSL facilities in the Greater London area, but when I sold Good Earth there were loads. That was a sign of the times, and some SSL facilities were dropping their prices to under £500 a day. I thought, "This is ridiculous; if I'm not producing all year long I have to be a studio owner."'

Visconti sold Good Earth to a jingles company, but still uses the facility today. Now, however, he has to travel by plane to get there, because about six months after off-loading the Soho operation he moved back to New York. Initially he was busy working on both sides of the pond, but then the assignments started to thin out and Visconti took advantage of the hiatus to study the Alexander Technique, a mind-and-body method of focusing on the precise coordination of head, neck and spine in order to attain economy of movement, physical comfort and peak artistic performance. Sting and Paul McCartney are two of its more notable advocates.

Now a qualified teacher in the AT, Visconti asserts that he can not only 'sit at a mixing console all day long without developing stiffness and backache' but also help the artists who he is producing to improve their singing and playing. Of late, there have been plenty of them for Tony Visconti to deal with.

'Things have really picked up incredibly well,' he says, referring to recent projects with Sparks, Alex Forbes, Air (aka Arvingarna in their native Sweden) and The Sea Horses, the new outfit of former Stone Roses lead guitarist, John Squire.

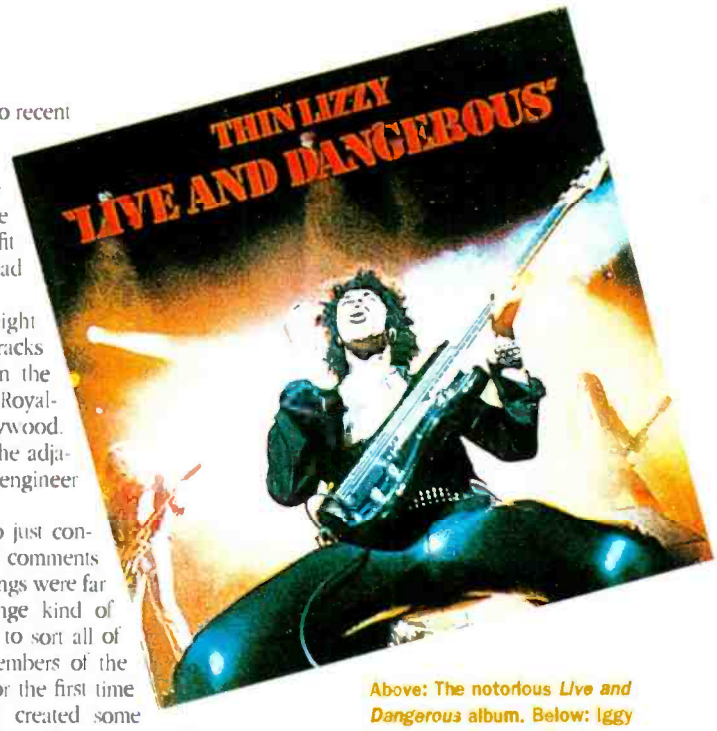
Visconti spent eight weeks recording 15 tracks with The Sea Horses in the vintage Neve room at Royaltone Studios in Hollywood. The mix took place in the adjacent SSL room and the engineer was Rob Jacob.

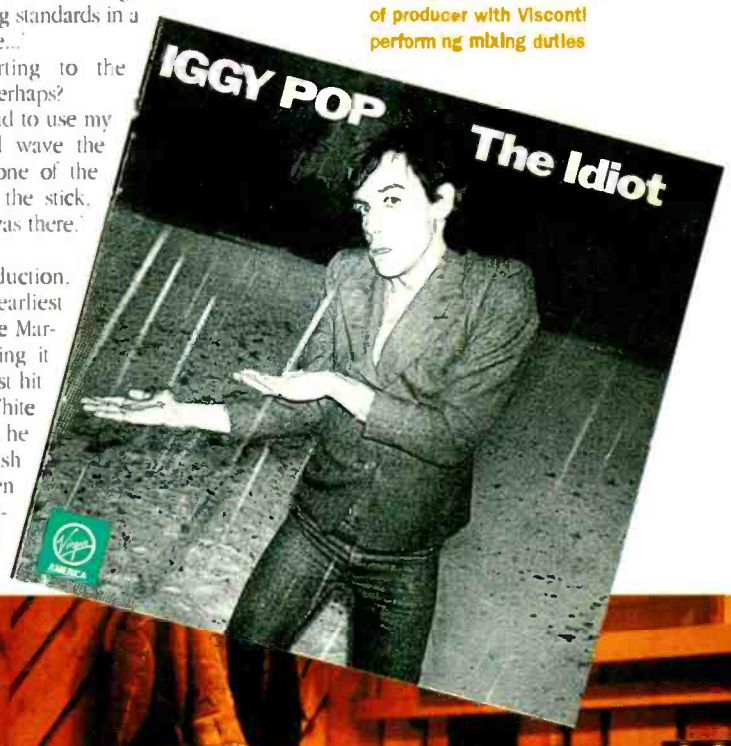
'I loved being able to just concentrate on the music,' comments Visconti. 'Some of the songs were far too long, or in a strange kind of sequence, and so I had to sort all of that out. Also, three members of the group were recording for the first time in their lives, and that created some problems which I don't usually encounter. It was quite a task to coach them and get them up to studio-playing standards in a very short period of time...'

Which meant resorting to the Alexander Technique, perhaps?

'Actually, no. I just had to use my Gestapo technique and wave the stick a few times. As one of the guys said, I didn't use the stick, I just let them know it was there.'

IN TERMS of production, Tony Visconti's earliest influence was George Martin. In terms of arranging it was Phil Spector. His first hit was T Rex' 'Ride a White Swan', and looking back he now can hardly distinguish the difference between the quality of, and working approach towards, that record **page 55 >**





Above: The notorious *Live and Dangerous* album. Below: Iggy Pop's The Idiot saw David Bowie graduate to the role of producer with Visconti perform ng mixing duties

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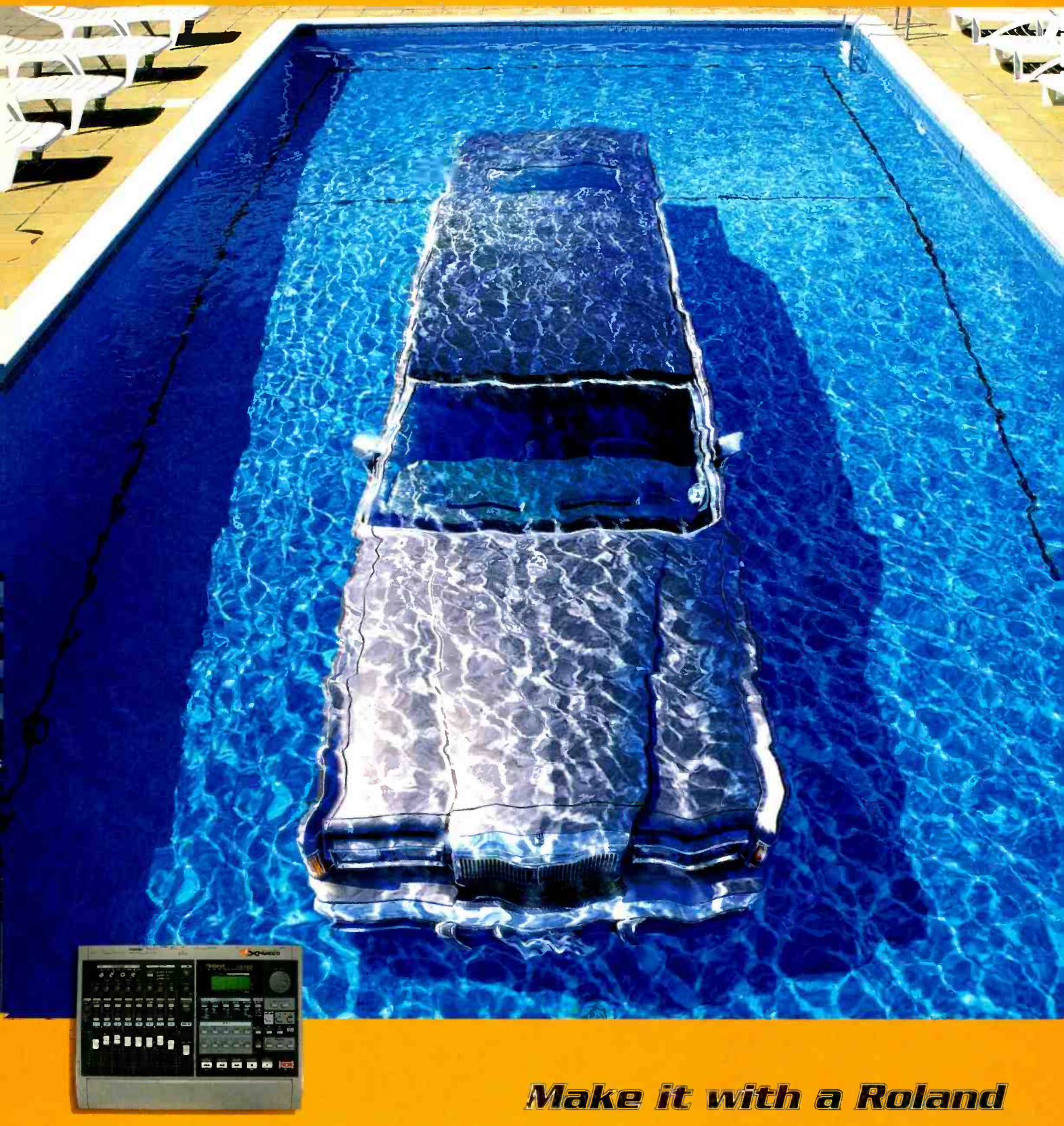

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<page 55 and those that preceded it.

'We did everything the same way,' he explains, 'except that we added a little string section, and perhaps the song was a bit simpler and easier to understand. Whatever it was, that single went up to No.2 in the charts, and suddenly that encouraged a shift of consciousness for me and Marc Bolan.

'Everything that we did after that seemed to work, and practically all of my projects were successful. So, I'm not sure if I was consciously aware of what I was doing, but on a sub-conscious level I suddenly had a knack for making hit records.

'The T Rex recordings brought together a lot of elements. Denny Cordell taught me how to get that bass and drum sound. He really drilled me; he was my mentor. It was a case of tuning the drums, positioning the mics, using the room, damping the snare in a certain way—the very live snare drum sound that people are getting these days is not the same as that thud of the 1960s and '70s.

'Marc's songwriting was a bit retro, but the sound pretty much defined the early 1970s. Before that the guitars always sounded clean. Now they were distorted. Then there was the use of a string section, which was an old concept, except that in our case we used larger sections and the way that I wrote for them took things a stage further. One of our big inspirations was the string section to 'Will You Still Love Me Tomorrow' by The Shirelles —I think that's the coolest writing. It sounds like three violins and one cello, and initially the strings of T Rex were based on that. On the later albums, however, the string writing got completely insane. Like that track, 'Whatever Happened to the Teenage Dream' on the *Zinc Alloy* album; that was the most bizarre string part that I had ever written. So we kept breaking ground on a lot of levels.'

Visconti was formally trained in orchestration at high school and by way of private lessons. So it was that, with classical guitar on lap and score paper in front of him, he would write the arrangements for Bolan and co.

'By the time that I got around to T Rex I was quite adept at strange voicings and especially the string orchestra,' he says. 'I know that well because I'm also a double bass player. So, I'd sit down with the T Rex track cued up on a Revox in front of me and a remote control next to my right hand, and then I'd play the intro, stop the machine, scribble out a few lines, check on my guitar, or on paper, that the voicings were good, and so on. A couple of hours later I would have the whole string arrangement written, and then I'd send the score to a copyist and a week or two later we would be in front of a string section that was playing the part.

'There was only one string chart that I actually improvised in the studio, and that was for 'Get It On' [US title: 'Bang a Gong']. I pointed out to Mark that 'Ride a White Swan' and 'Hot Love' both had strings, whereas this new single didn't, and he suddenly got very superstitious. I already was a little superstitious and so I was just transferring my fear onto him. He said, "Can you write something right now?". We were in the studio and the strings were sitting down—they had already played on one or two other songs—and so I said, "Well, the quickest thing I can do is come up with a string line over the chorus". The chords were G, A, and E, and so that's what I had everyone play, and it worked beautifully! It was the missing element and I'm just so happy we did that. Anyway, that was the only time that I actually arranged on the spot—apart from correcting bad notes, which is something that you always have to do when you use a copyist.'

Caught in the studio: a young Tony Visconti and Marc Bolan

Talking of bad notes, Bolan himself may have hit the right ones on his guitar, but he often did so while clearly out of tune.

'That was something that I couldn't even address on the first three albums,' muses the producer. 'Eventually, as he began to trust me, I would adjust the tuning more and more, but I still had to do it in private. I couldn't tell him his guitar was out of tune on the talkback. He would get very obstreperous. So, I'd just take a break, walk downstairs—say this was at Trident—and tell him politely in his ear that his guitar needed tuning.

'I mean, once or twice when I did say it in front of other people he would tell me I was mad or tell me to f-off. He was a hard guy to produce on that level but once I knew the secret, I could get him to do anything.'

By the time of *Zinc Alloy and the Hidden Riders of Tomorrow*, criticisms were being levelled at Bolan that he was repeating himself. Visconti confronted him on this issue and proposed a year's sabbatical, during which time > page 57

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<page 57 he could compose his own *Sgt. Pepper-Tommy*-type opus.

'He agreed with me,' Visconti recalls, 'but he said, "I think we still need to do one more pop album for the fans". The fans, the fans, the fans. Unfortunately, he was using copious amounts of cocaine and Cognac during that period, his wife had left him, and he had a new girlfriend in the form of Gloria Jones. You'd look at it all and say, "What's wrong with this picture?" Gloria came in at such a tail-end of the T Rex phenomenon, and Marc was trying to make her fit. Also, there was great acrimony between him and the members of the band who, after all those years of success, were still on a basic wage of between £75 and £100 a week. Here he was, giving Gloria a big break and making her a member of T Rex, yet he wasn't really acknowledging the guys who had stood by him for many, many years.

'The whole thing was off-kilter. The songs were not very good, his attitude towards making records was old-fashioned, he still thought he was current when everyone was telling him he wasn't.'

Which is somewhat surprising considering that the artist was still in his mid-20s.

'Well, that was the drugs. I couldn't get him out of that mind-set, even though he did agree with me that we would take a break after this next album and do something different. I also had a demo of this little rock opera that he had written, called *The Children of Rarn*, but he said that we couldn't do that until far off in the future. At the same time, during the making of T Rex, the way that he treated me and the way he treated others got to such a point that I thought the situation was irretrievable. We couldn't find that original spark or friendship ever again. I felt very put out, and I called it a day. I quit, and the rest of the guys followed.

'It was a very sad way to end things. A few years later Bowie brought him to see Good Earth Studios, and he said that he would love to make a record there, but then he died a few days later. So, there could have been a Visconti-T Rex reunion, and it would have been the right time too. He had picked himself up, he was no longer using drugs, and he looked like he was in fine shape, but unfortunately he had the car accident.'

THEN there was the aforementioned Bowie, with whom Visconti ended up working on 13 albums, starting with *Space Oddity* and ending with the music to a 1982 BBC TV drama named *Baal*. In between there were some seminal works—including *Young Americans* and *Scary Monsters*—and a fair few in-studio innovations.

'The harmonised snare drum sound made its debut on *Low*,' cites Visconti. 'We then kept that up on *Heroes* and used the harmoniser for a few other little things too. The secret, however, was that the harmoniser was played live. Dennis Davis heard what he was doing in his headphones as he played, so that's why no one else can get that drum sound.'

The harmoniser also came into play when Bowie was shooting the 1983 film, *The Hunger*, during the course of which his voice was required to age. The complication arose with Bowie's wish to record live dialogue without picking up the voice of his leading lady. Visconti's solution was a personal mic and a harmoniser.

'I strapped a C-Ducer to his neck and he wore a scarf over it,' Visconti recalls. 'I then put it through a harmoniser and made his voice go up in pitch, and it worked really well, although I had to filter out the pulse that was coming through the arteries in his neck. He also had to shave to prevent the sound of his bristles scratching.'

Given the constant metamorphosing of Bowie's professional persona and the wildly eclectic nature of his songwriting, Visconti's major contribution was to impart a sense of direction to his albums, in terms of the sound and arrangements. Another was his occasional playing of bass and various other instruments.

'I think I helped David by getting him a little more focused that he would have normally been,' he says. 'Certainly, with *The Man Who Sold The World* we paved the way for the Ziggy Stardust sound. That identical sound was used afterwards without me, and I helped to forge that. Also, with regard to his own writing, I would definitely help him to organise his thoughts, because a lot of his songs would just be backing tracks with a melody that he'd sing la-la to. After weeks of overdubs, a track would suddenly develop a personality and then he would write lyrics and a different melody, and I would help him with that. It was very much a cooperative thing and he was a joy to work with.'

Here again, Visconti also helped cure sibilance problems caused by Bowie's teeth.

'I sometimes used a Beyer M160 dual-ribbon mic for David's voice,' he says. >page 59

The Project Studio

EVEN THOUGH he no longer owns his own commercial studio, Tony Visconti is still a compulsive shopper when it comes to some of the latest gear. To help economise on certain projects he will often record in his largely digital facility, which is equipped with a 32-track ADAT machine, Macintosh hard-disk recording, and Notator and Sound Designer for editing and pitch-correction. There is also a MIDI-addressable 32-channel Soundtracs board, and Alesis 1, KRK 6000 and Genelec 1030 monitors, as well as a good collection of microphones, including a pair of Neumann U87s that were retained from Good Earth.

Having performed on many of the records that he has produced, Visconti also has a collection of vintage guitars, including a '59 Fender Strat, a couple of Tony Zemaitis electrics and the original Fender Precision bass that he played on Bowie's 'The Man Who Sold The World' album. Then there are the keyboards, 'too numerous to mention', of which TV's favourites are his old MIDI-retrofitted Jupiter 8, a new Jupiter 1080, a Wave SR, a Roland D50, an Akai S1000 and a Roland digital piano.

As for the toys; 'I've got loads of reverb units and so on, but I rely more and more on my Macintosh for special effects than outboard gear now. Compression, gating, you name it. I'd much rather do those things on a Mac than in an automated mix.'

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<page 59 'It's not as responsive to high frequencies as it is to low ones, and so it made him sound clear and warm. Basically, however, he's a phenomenal singer. I've never worked with anyone better. He is the best. His mind is mercurial and he's very quick to see unique angles, yet one of his marked talents is to hire a studio full of talented people and to let them do their thing. He'd sit back for hours while I was working with, say, Carlos Alomar or Robert Fripp or Brian Eno, and he'd observe and maybe make a suggestion now and then. His openness is the best thing about him, and he's willing to try anything —as long as it's quick!'

Which is what Tony Visconti and Thin Lizzy hoped would be the case when it came to do the 'mix' of their classic *Live and Dangerous* album. However, a 3-week 'quickie' project soon turned into a 2-month affair, as *Live and Dangerous* assumed all the hallmarks of a studio recording—save for the fact that the drums and the applause were salvaged from the original concert performances.

'Everyone was doing it,' Visconti is quick to point out. 'We just did it blatantly!'

Surely, though, if the band could really play their instruments there should be no need for extensive studio overdubs.

'As Phil [Lynott] sang and played bass at the same time, he made a lot of bass mistakes,' Visconti explains, 'so he wanted to fix them in the studio. But then, when we tried to do that, it was very, very difficult to match the sound. I told him it would be easier for him to play the whole bass part again rather than fixing just a few notes, and he said, "Fine". Then he said that he had to replace the vocals—"I was off-mic, I was out of breath"—so we replaced the vocals.

'These things were well recorded. We were able to do a new vocal and you couldn't hear the old vocal on the audience mics. He was so close to the old vocal anyway. That was as far as we were going to go, but then the guitarists heard that and noticed that we were able to do it without any leakage problems. So we started doing the guitars and it worked—it kept sounding better and better. The audience reaction was always in the background, and so we ended up with what I call a "live-plus" album. In fact we'd given ourselves three weeks to finish it and ended up spending two months on it—a very well-crafted studio album.'

PHOTO: BARRY MARSDEN

'I intend to keep adding to the list of achievements before I kick off,' concludes Tony Visconti, 'but I suppose if I have to choose three tracks right now that I would like to represent me...'

And he does, because I've asked him to: 'First off I'll go for any track off of the new album by Sparks. It's a remake of some of their hits and also some of their more obscure tracks. Not a symphonic album, but a sort of alternative album, using a small string section, electronic effects and a madrigal choir, to redo all of their songs in an avant-garde, classical way. I'm very, very proud of it. So I'll go for any one of the Sparks tracks, together with 'Ashes To Ashes' and 'Get It On'... Anyway, I'm sure the list will change in a year from now.'

But while Tony Visconti is sure that his view of his own work may change, it is certain that nothing will replace what has gone before. Rather, the legacy will keep on growing. ■

Terry Wogan and Katrina. With a smile and a song, Terry Wogan brings his own brand of humour to the proceedings as he provides the UK commentary to the 42nd Eurovision Song Contest, held for the seventh time in Ireland

Inner vision

The world's largest satellite transmission takes Eurovision around the globe. **Kevin Hilton** discovers the value of organised production, the real meaning of technical redundancy, and the relief of a bad song

THE PUBLIC FACE of countries in friendly competition is one thing, but backstage at the Eurovision Song Contest is something else. Good luck that all this is taking place in Ireland, where the production team and technical crew exude typical Irish laid-back-ness. The only thing causing concern is the thought that the country could win again and have to do it all next year—again. Four times in five years is surely enough.

It works in the favour of host broadcaster RTE (Radio Telefis Eireann), though. Eurovision cuts into RTE's equipment budget—and sponsorship from manufacturers is always welcome—but at least it has honed skills and techniques, all of which are necessary to approach the biggest television satellite transmission in the world. There's an audience in the Point Theatre in Dublin, but the real concern is making sure that the 300 million plus viewers watching see the show and that when the all important votes come in, everybody knows who got 'nil point' (it was Norway, again).

RTE's love-hate relationship with the Eurovision Finals began in 1971, following Dana's win the previous year in the Netherlands. It was renewed in 1981, but 1988 was the significant year, the one in which many of the current technical team became involved and devised the methodology that has served it so well over a long, unprecedented run. Linda Martin's winning song in 1992, prophetically titled 'Why Me?', kicked off Ireland's domination of the competition, with further wins in 1993 and 1994. Norway broke the chain in 1995 but it was still a spiritual win for Ireland, as the Norwegian band Secret Garden was fronted by an Irish singer performing a decidedly Celtic song. (Then again, Celine Dion won for Switzerland in 1988 and American-born Katrina Leskanich represented the UK this year.)

Ireland entered the weakest song it could find this year, 'Mysterious Woman' by Marc Roberts (in reality a country singer from County Mayo called Sean Hegarty), but still found itself joint favourite with the UK and Italy. 'So we're favourites again?' asked one technician off-handedly, perhaps giving the impression that with Eurovision, it's the means that is important, not necessarily the content.

It's easy to understand, as many at RTE had been working towards the big show on 1st May since January. The centre of the action, as far as the audio crew is concerned, is not the auditorium of the Point with its impressive, futuristic, metal set and stage; but the sound distribution room, an orderly jumble of cables and racks sited down one of the former railway depot's long concrete corridors.

'The planning of the sheets started in January,' says sound supervisor Jack Peoples. 'We use the old ones as a base—it's similar year to year, we find.'

The racks in sound distribution were designed and built for the 1988 Final and have been in and out of cold storage ever since. Each cable carries seven microphones, organised in split pairs so that primary users have access, including foldback and the front of house sound system. In all 68 BSS splitters are on site, with only 62 or 63 fully employed. The phrase most often heard around the Point, in technical terms at least, is 'redundancy'. Pairs of splitters are held back just in case of emergency—when you're hosting a showcase broadcast, the last thing you want is dead air or a blank screen.

All feeds appear on the patchbays, organised so that there is as little congestion as possible and quick access. 'We feed to users from both the front and back of the racks,' comments Brendan Lyons, head of sound distribution. 'Everything is using the BSS splitters, except the foldback. For turnaround purposes we've set up A and B feeds on the vocal mics, so that while one country is performing, the next country is setting up.'

Each main sound section is defined in sound distribution: the house system, foldback, monitor output, main stereo and main mono (with stand-by stereo and mono). Mixed programme sound from RTE's OB Truck 7 comes into sound distribution and then is relayed back to the broadcaster's headquarters in the Donnybrook area of Dublin, together with all the voting feeds and visuals.

using sound-in-syncs.

This year's Final may have been hosted by RTE, but, ultimately, Eurovision is an EBU event. Hence the EBU room just up the corridor from sound distribution. 'It's an RTE facility,' explains Conor O'Loughlin, head of television sound, 'but we work to an EBU representative.' It's this room that handles the information that decides the Final—the votes from the juries of all the competing countries. Incoming are four satellite feeds, designated B, E, C, and a stand-by in London. These work in conjunction with four independent sound circuits: X, Y, Hilversum and London, all switched through a centre in Frankfurt. The Irish jury, at RTE headquarters, is brought in on a fibre-optic connection.

This year's Finals marked the third time in-vision juries have been used, something pioneered by RTE in 1994. With or without this, the procedures for making sure that the secretaries of the juries hear the songs, and that each jury comes in at the specified time are complex.

'It takes two minutes to set up each country's jury,' explains Bernard Gibson, overseeing the EBU room for RTE. 'Each one has to power down before the next one powers up. Otherwise...'

If there are any problems with the satellite feeds, the vision mixers can cut to a caption slide of the relevant country, with the votes cast in sound only. For this reason there is 100% redundancy built into the audio system. If the main sound circuits fail, then three telephone lines are standing by. If they go down, two talkback feeds are in reserve. If the unthinkable happens, a fax machine is laid on so that the votes can get through somehow; for extra security no one knows the number of this machine until the day of the show.

The main jury audio signals are mixed through a Soundcraft K1, with a Spirit Live as a stand-by. Both have backup power supplies, to be sure. Another Spirit Live controls mono feeds of the music to the jury secretaries, with the EBU co-ordinator needing to speak to these officials. While the juries received a mono feed, each country broadcast is in stereo, calling for the use of a separate satellite as Eurovision distribution is normally in mono. A backup board is provided underneath this particular Spirit Live, with a battery-powered Shure submixer as a final fail-safe. The EBU loaned RTE six Symetrix noise-reduction units, while lip synchronisation is ensured by a Klark-Teknik time delay.

THE JOB of presenting the Eurovision Song Contest is a coveted one—this year the task is split, falling to up and coming RTE TV and radio presenter Carrie Crowley and Ronan Keating, lead singer of Irish 'boy band' Boyzone, who appear during the interval performing a decidedly Riverdance-influenced new song—the show is carried by each country's commentator. The UK's stalwart for this is Limerick-born Terry Wogan, whose blarney and ironic asides can lift the Contest out of its tedium.

The commentators are positioned at the back of the auditorium, each equipped with a 14-inch TV monitor, script light and combined microphone-headphone headset. From here audio feeds are sent to a cabin containing 10 Glensound mixing stations, overseen by four operators (taking ten countries each). 'It can be bedlam in here,' Conor O'Loughlin remarks dryly, 'because a lot of the circuits aren't booked until just before the show starts.'

In front of this cabin are the various outside broadcast trucks for sound and vision. OB7, with its relatively new Calrec Q-series console, is where the overall audio mix is assembled by Ian Pike. Just as the distracting job of dealing with any mic or cable failures are split off and isolated in the sound distribution room, other elements are delegated elsewhere to allow Pike to concentrate on putting together the production sound.

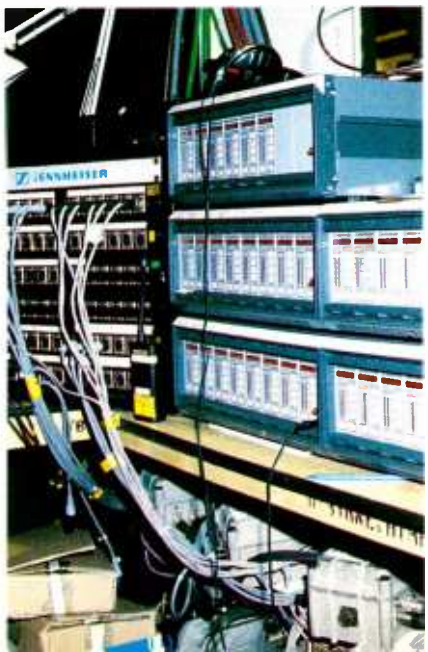
'We found over the years that the vocal mix took nearly all the rehearsal time,' explains Pike, 'so we separated it off.' This, and the stage instruments, is dealt with by Jack Peoples in a truck borrowed from BBC Northern Ireland, known during the show as the Mobile. 'This means that I don't have to deal with the stage crew,' Pike continues. 'Three stereo mixes come in from the Mobile: lead vocals, backing vocals and the stage instruments. I can then concentrate on blending those in with the orchestra.'

Another element broken away from OB7 is the string section, which is mixed on a Soundcraft 8000 located in the sound distribution room. A total of 24 microphones is divided into five mixes, which are sent to OB7, with two going to the front of house system. These mixes come up on the left-hand side of the 60-channel stereo Q-series, with the woodwind (eight microphones) on four channels.

The right-hand side of the desk holds the so-called hardware, including a number of backing tracks. 'About 19 countries are using backing tracks,' says Pike, 'for various elements. Only about four of them contain full orchestras, the others are maybe just bits of percussion or other things.' Also located here are the presenters' mics, the outputs of the voting desks and effects. All the feeds from the Mobile come in dry, with Pike adding reverb in OB7 (a Lexicon LX15 is used for the main vocals).

All processing equipment is under MIDI control, as is a wall-mounted Yamaha DMP7, which handles percussion mixes. MIDI patches are assigned to recall levels and effects parameters for each country. 'It makes it an awful lot easier than trying to follow with fader moves,' Pike observes.

The main desk is similarly fitted with a recall system, a very sophisticated one, as Pike and O'Loughlin are at pains to point out. 'It's total recall, or total Formica,' Pike smiles. A strip of the material, once a common site in rural Irish bars, is prepared for each country, with fader positions written on it. The relevant strip is placed in the space above page 62 >



From the top. 1: Sound supervisor Jack Peoples prepares the vocal mix on the Yamaha 02R, with the Calrec M-Series in the background, in the BBC Northern Ireland mobile. 2: Katrina and the Waves, left to right, Vince de la Cruz, Katrina Leskanich and Alex Cooper. 3: Bernard Gibson sets up the EBU room, waiting for those votes. 4: Presenters Ronan Keating, left, and Carrie Crowley. 5: Radio Mic position (Can you guess whose system RTE was using?)

'Tape' play-in area, orchestra position in the background, with Tascam DA88 and Spirit F1s

< page 61 the faders, with each instruction corresponding to a channel. 'I can re-create instantly the rehearsal mix,' says Pike. 'Okay, so there are changes on the night, but it gives a pretty good start point. We do that 25 times, it's a very high-tech system that has never let us down.'

Each strip is changed over during the visual 'postcards', which introduce each song (along with the commentary), a hectic couple of minutes for the crew, which consisted of Pike, O'Loughlin (as second engineer or 'heart attack man', just in case Pike succumbs during the show) and the musical associate.

Audience reaction is picked up by four stereo pairs of Sennheiser 416s and mixed on a Calrec 8:2 submixer in OB7. A stereo pair of radio mics is also taken into the Mobile. Pike observes that the only thing he really worries about on the night is losing the lead vocal mics, but in keeping with the full redundancy policy, he can assemble a fresh vocal mix if connections to the Mobile are lost.

'All six radios come into me,' he says, 'so I can get at them if need be. That goes for the voting feeds as well.'

Similarly the Mobile can act as the main mix station if something drastic happens to OB7, although the intention was for it to deal with vocals and the live on-stage instruments. Six main radio mics are used, along with two headset combinations, five instrument mics and three DIs. Jack Peoples comments: 'My function is to take those and mix them into three stereo groups (lead vocals, backing vocals and instruments).' To do this, Peoples uses the Mobile's 40:8:2 Calrec M-series console to send pre-insert feeds to a Yamaha 02R, which mixes, EQs and buses the three stereo outputs. This is then returned to the group faders of the Calrec.

'The reason we've done this,' explains Peoples, 'is that in the event of the 02R having a problem, I can release the insert switching and take everything back again onto the Calrec.' The 02R is used to store the individual EQ and level changes for each country, which are then recalled as needed. 'I might have to do a bit of pushing and pulling during the songs,' says Peoples, 'and maybe a bit of extra EQing during the choruses but it gives me a good starting point.'

Like others working on this year's event, Peoples is a veteran of the Eurovision Song Contest, having performed the main mix in 1988 and prepared the vocal mix three years ago, although on that occasion he was using an analogue Neve desk. In addition to the vocal mix, the Mobile's secondary function is as emergency main mixer. The rhythm section is submixed on a Soundcraft console, which can be sent to the Calrec, along with the string instrument mix from sound distribution and the percussion, and the woodwind and brass, which comes through a Calrec 18:2 rack mixer.

The job itself Peoples describes as: 'Mixing the stage sound, but totally in isolation, without reference to Ian's mix. I'm mixing it dry because I'm trying to balance the three groups so that in theory Ian doesn't have to push or pull the faders at his end and he can add whatever effects he wants. The idea is that Ian won't have to touch those three faders.'

Entering the truck as this point, Conor O'Loughlin, referring to the link up of the M-series and the 02R, remarks, 'I don't know whether this has been done before. Nice lateral thinking, Jack.'

Peoples modestly replies: 'If you start thinking about something in January, you're bound to come up with something nearly original.'

Peoples also works with a deputy, although he says that the job is more than just being there in case of heart attacks. 'There's a lot of fader grabbing that goes on and the main mixer needs to concentrate on just the music,' he says. This also counts for OB7. O'Loughlin was in charge of the presenters' mics, postcards and feeds from the Green Room, where artists wait for the votes to come in.

By the nature of the organisation, both Pike and Peoples are isolated from the activity going on around the stage area. As Pike observes, 'If a radio mic fails, somebody else deals with it.' The wireless mic area is positioned behind the orchestra, stage right. Technician Gerry Malone operates a strict regime in handing out mics, with each one colour co-ordinated and assigned to the correct artists and positions on stage.

As in previous years when Ireland has hosted Eurovision, Sennheiser has provided the radio systems, acting as one of the equipment sponsors of the event (along with Soundcraft). A total of 42 channels of wireless mics are used, divided into two individual groups and two different frequency ranges for safety. Providing technical background is Klaus Willemsen of Sennheiser's customer service department. 'Each group of 21-channels is connected to its own antenna,' he comments, 'with four antenna on stage. In each group there are 16 hand-helds and five pocket units, all running with rechargable batteries. These give an operating time of five hours, which is enough for the show.'

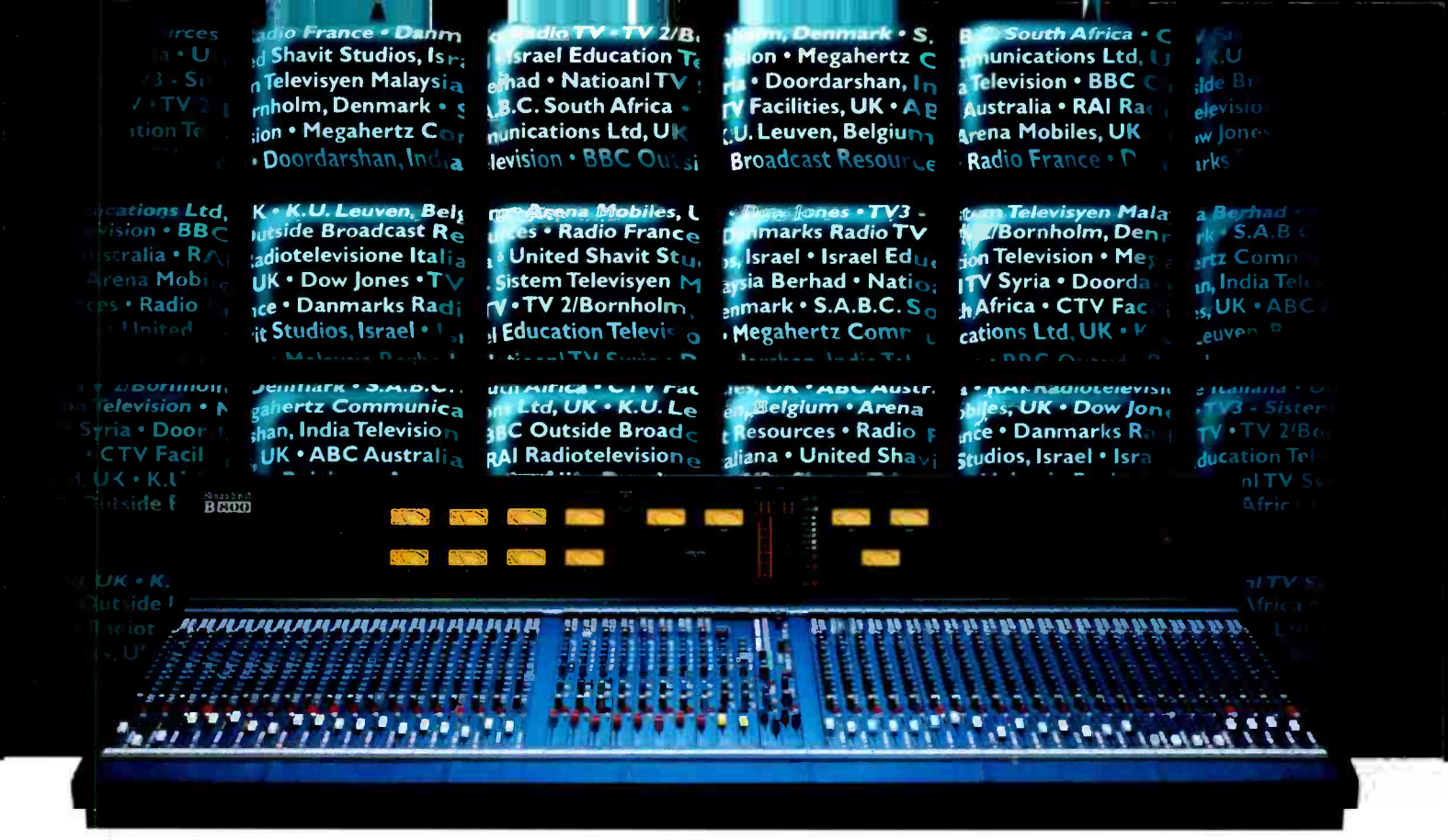
Eurovision uses a 19-inch rackmounted EM1046 diversity receiver system, working in conjunction with a combination of six SKM5000 transmitters for the vocals, another six for on-stage instruments (each plus six stand-bys) and six SK50 pocket transmitters for DIs. A few EM3032s are also used. The presentation mics are six SK50s (two transmitters, two mics and two stand-bys). In addition the majority of orchestra microphones are Sennheiser.

Next to the wireless mic position is the 'tape' playback area, based around two Tascam DA88s, loaded with pre-recorded music, click tracks and metronomes, and two Spirit F1s. 'We wanted the conductors to cue the tapes as if they were drums,' comments Conor O'Loughlin. 'There are two stereo channels and six mono on each Spirit F1. The systems are duplicates of each other and both run in sync. The click-track comes up on pre-fade to the headphone feeds and the conductor and through the stage monitors for the performers. There's a 2-second delay as the machine speeds up, so that has to be taken into consideration.'

Stage monitoring comes from eight Electro-Voice SX200 cabinets built into the futuristic dais, which is set 18-inches above the ground. 'We dismissed the idea of in-ear monitoring,' comments Ian Pike, 'apart from for Ronan and Carrie, who use Garwoods. They wouldn't have worked for the other artists because of the rigging time, hygiene reasons and the fact that many people wouldn't have been used to using them.'

Then, after a year of anticipation and months of planning, it is all over. Sean Hegarty, sorry, Marc Roberts comes in second—the whole of Ireland will sleep easy this night—while Katrina and the Waves storm the competition with 227 points, the highest Eurovision score ever.

There is still wonder at why this event is taken so seriously; the music industry turns its back on it and most people question its relevance today. Even the kitsch factor only goes so far. Perhaps that's all because the Song Contest is in reality a showcase for broadcasters—the songs are irrelevant—and next year it will be up to the BBC to continue pushing the standard, while praying that a really bad song is chosen so that they don't become as proficient as RTE.



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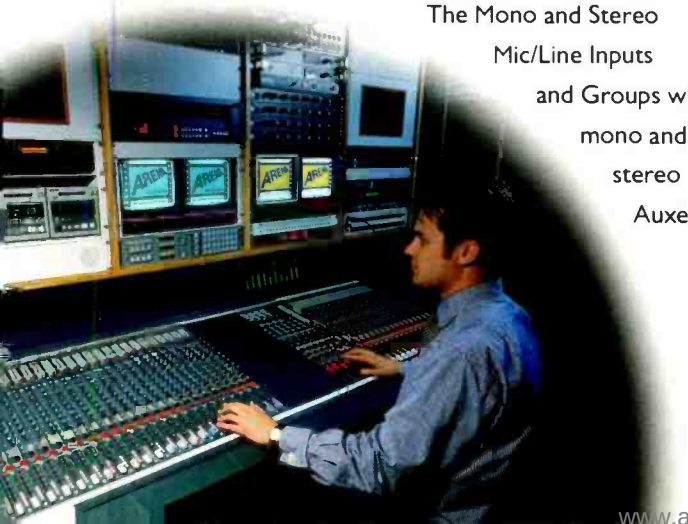

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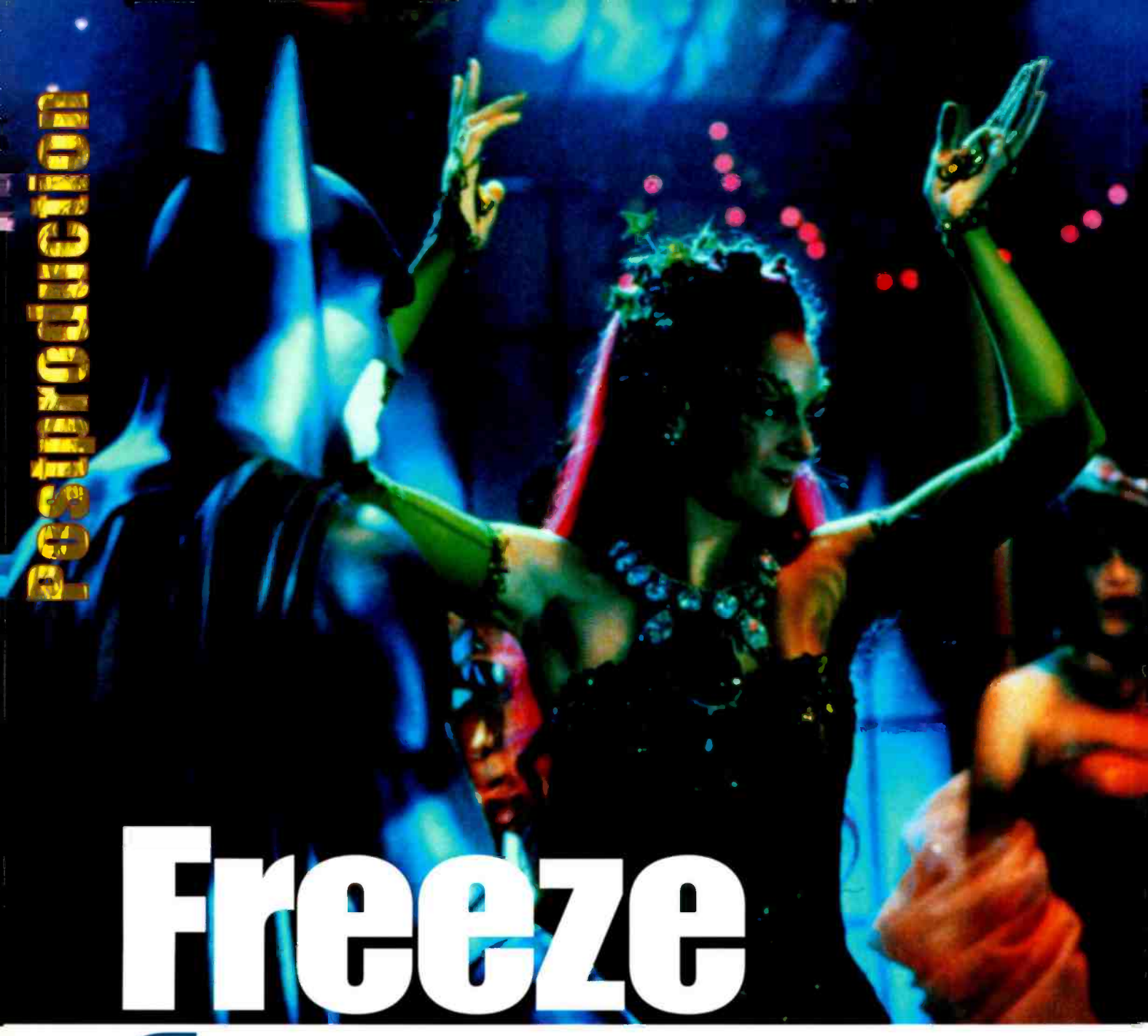


Freeze frame

The Dynamic Duo returns to the big screen, championing high-profile sound design. **Mel Lambert** talks to the co-supervising sound editors and the rerecording crew at Warner-Hollywood about *Batman and Robin*

IF YOU'RE ONLY AS BIG as your last movie success, then the supervising sound editorial team of John Leveque and Bruce Stambler stands head and shoulders above the rest in Hollywood. That Stambler finally received the nod in the 1996 Oscar Awards for Best Sound Effects Editing on *The Ghost in the Darkness* was considered by many in this town to be long overdue as the team has worked on some of the most innovative movies to grace the silver screen in recent years. Now it's turned its attention to this summer's blockbuster, Batman and Robin.

The new offering from Warner Bros. marks a reunion with producer Peter Macgregor-Scott and director Joel Schumacher, the team that brought us 1995's *Batman Forever*. 'This time around,' stresses Macgregor-Scott, himself a former sound editor, 'we have an entirely different look and feel to the characters and story line. Each character and gadget needs its own distinctive sound 'signature.' Having worked with John and Bruce and their team on a number of highly successful films —including *Batman Forever*—it was natural that we do it again with our latest adventure.'

George Clooney, star of NBC-TV's *ER* series, takes over as the Dark Knight, and is joined by Arnold Schwarzenegger as Mr Freeze, Uma Thurman as Poison Ivy while Chris O'Donnell retains his role as Robin.

The film's plot of revenge and misplaced vengeance sees scientist Victor Fries trying to save the life of his wife who is dying from a mysterious disease. The project is unauthorised, and officials come in to clean out Fries. During a struggle, a cryogenic machine is smashed, splashing Victor with fluid. As Mr Freeze, he has to create a suit to keep him at 50° below. His quest is revenge on Batman.

'Sound is now an integral part of any film,'

Above: Poison Ivy (Uma Thurman) the flower-pot femme fatale, greets Batman (George Clooney) and Robin (Chris O'Donnell)

confides director Joel Schumacher. 'For a movie that I can best describe as a living comic book there are extensive sets and costumes that need to be detailed with sound. It's like a pop opera, with sound used to heighten the fantasy. Sound effects have become as essential to the story-telling process as the visuals.'

'It's obvious that movie audiences today expect a film like *Batman and Robin* to sound as good as it looks,' considers John Leveque, who with long-time colleague Bruce Stambler handled the complex task of selecting and editing the thousands of sound elements used in the film.

This experienced sound editor has a clear blueprint of how he will develop the various elements required to complete a film's soundtrack! 'Big, bold, loud, emotional—that's how I like my sound. The helicopter 'screams' as it careens off the deck of the USS Missouri in *Under Siege*. The bus wrapped around the front of the train screeches along the track in *The Fugitive*. The Batmobile zooms. The Batcape flaps. The Batboat growls. The Batarang clangs. All of these—and hundreds of others—are a sound-editor's dream.'

'What Joel Schumacher puts on the screen is the blueprint,' explains Stambler. 'We design the sound around the action, so that it reinforces the focus of a scene, yet without being distracting. There's a very fine line between too much and too little; we carefully select the right combination of sound effects to draw the audience into the film.'

The process of designing sound for *Batman and Robin* began, Stambler explains, with their first meeting with picture editor Dennis Virkler. 'Since most of the action material had been shot without sound there was very little audio for Dennis to use when he did his first cut of the film. He wanted us involved early because he is under pressure to show a first cut to the director, producer and the film company, and he can be more productive if he has a rough soundtrack to work with. We received a list of specific sounds that Dennis wanted to hear, so we went back to Sound-Storm, pulled effects and then cut a work track for him.'

This speeded the process of temp dubs which culminated in the 6-track 'Temp #3 being produced for test screening of the film during early May. 'Normally, we receive a rough cut as the starting point for our sound design,' Leveque continues. 'Here, because we had this unique opportunity of working with Dennis so early in the movie's development, we could get a good sense of what would be needed for such a sound-heavy movie.'

Co-supervising sound editors Bruce Stambler (left) and John Leveque

To handle the spectrum of sounds, the cosupervising sound editors divided up the assignments; while Stambler looked after 95% of the sound effects and handled liaison with the editorial crew, Leveque was responsible for Foley, ADR and dialogue, plus the remainder of the effects. In all of their productions, Leveque and Stambler favour the use of natural sounds that they record specifically for the movie they're working on.

To create the sound of the Batmobile, Leveque and Stambler visited the Rocket Dyne Space Shuttle facility in Canoga Park, California, where they recorded the intense blast of an Atlas rocket engine.

'The core of the Batmobile sound is an 800-horsepower Buick Grand National with a wonderful turbocharger whine,' recalls Stambler, whose rare off-hours are often spent around high-power automobiles. 'All in all, we used 60 different, individual sounds.' A race track near to Los Angeles was also rented to record various sounds of a Ducatti motorcycle and a rare Porsche 917 12-cylinder racer. Both elements were to find their way into the new Batbike sound creation, which we see Robin riding early in the film.

One of the favourite sounds used in the film came about by accident, as Bruce Stambler recalls. 'We had already made a number of recordings of bat wings that were very good but we lacked a sound that page 66 >

John Leveque gathering footsteps at Big Bear Lake, using a Fostex PD-2 time code DAT machine

Above: George Clooney as Bruce Wayne meets Uma Thurman as Pamela Isley prior to her transformation into the lethal beauty Poison Ivy. Left: The crew during production of the final Temp Dub #3 at Warner Hollywood Stage 'D', in late April. Left to right: sound effects rerecording mixer Frank A Montano; producer Peter Macgregor-Scott; co-supervising editor John Leveque; music mixer Jeffrey J Haboush; and lead-dialogue mix mixer Donald O Mitchell

< page 65 would give the idea of Batman rising into the air and moving off at high speed.'

As fate would have it, at Warner Bros Studios in Burbank, north of LA, the facility's world-famous water tower had been covered in a massive tarpaulin. 'During a windy rainstorm,' Stambler recalls fondly with a smile, 'the tarp' came loose and ended up streaming off into the air held only by a single tether. We snuck under the tower with our microphone and a portable DAT recorder. What you hear in the movie is the sound of this enormous tarp flapping, whipping and snapping around in the gale-force wind.'

Most effects tracks were edited on SoundStorm's series of Fostex Foundation digital workstations, which were used to prepare 6-track reels for the series of temp dubs. Fairlight MFX3 workstations were used for Foley editing, primarily because that was the format delivered to the team from Warner-Hollywood's Foley stage.

With the sound elements coming together under the direction of Stambler and Leveque, Foley, ADR and dialogue editors at SoundStorm in late March, the focus of activity moved to two dubbing stages at Warner-Hollywood's facility. Because of a time crunch, Stages C and D were pressed into service during production of a series of three temp dubs.

Unlike what has become standard Hollywood practice, Stambler and Leveque do not consider temp dubs a necessary evil as they move towards the final rerecording of the soundtrack. 'Rather than use the expression temp dub,' offers John Leveque, 'we prefer to refer to them as experimental mixes—the first step in the process of producing the final mix, and one that let us develop the soundtrack as the editing process continues, and we begin to develop the look and feel for the soundtrack.

'We select and cut sounds that will appear in the final film, not ones that will be replaced later. In this way we can concentrate on the

detailing process, and also provide the directors, producer and picture editor with a more finished audio track than if we were just pulling sounds. Also, we are still waiting for opticals. CG and other visual elements; we were also working with a temporary music track for the temp dubs.' Composer Elliot Goldenthal didn't begin the scoring process until a week after the final temp dub. While Stambler supervised temp dub mixes in Stage C with Donald Mitchell, Leveque handled similar duties in Stage D with Frank Montano and Jeffrey Haboush.

Prior to the finals, Don Mitchell handled two weeks of dialogue. ADR and crowd-ADR predubs in C, while Frank Montano took over D for his effects predubs. Foley predubs, backgrounds and additional materials were also handled by rerecording mixer Kevin Carpenter in Studio 2 at nearby Warner-Burbank. Then, in early May, the rerecording crew of Mitchell-dialogue, Montano-effects and Jeffrey Haboush-music gathered in D for the finals and print mastering.

'I prefer to remain in one room for the finals,' Mitchell explains, 'to ensure a consistent mix. Even though they measure identically, I've found that mixes from the smaller Stage C sound softer than those made in D, because of the closer location of the screen to the console.' Both rooms feature identical Harrison PP-1 rerecording consoles equipped with Uptown moving-fader automation.

Lead-dialogue mixer Don Mitchell recalls that 95% of the dialogue on *Batman and Robin* was looped. 'We have so much steam, prop and other noises on the production track, that we had to replace most of the lines. Also, in certain scenes Arnold was difficult to hear clearly, because of his costume.

'During the predubs, we pitch-shifted George Clooney's voice down by page 68 >

Right: Arnold Schwarzenegger as the treacherous Mr Freeze. Below: Rerecording mixer Frank Montano at his extensive sound-effects section of the Harrison PP-1 console in Warner-Hollywood's Stage D, during predubs of *Batman and Robin* in early May. To accommodate the high number of sound-effects elements required during these sessions, a pair of outboard or 'sidecar' automated consoles had been placed to the right and behind his main mixing position

On 1st May, Warner's publicity machine kicked into high gear, including erecting a series of colourful billboards in downtown Burbank

< page 67 about 5% when he appears as Batman, to give it more depth and resonance, and to distinguish the two roles. For voice processing of Mr Freeze, we'll use more of a metallic feel, with a Lexicon 480XL or a flanger.'

For the finals, Mitchell produced a 6-track predub of production dialogue—using one track for each of the principal actors—and a six-track of looped lines. Also prepared were two, 6-track predubs of group dialogue (particularly for the skating sequences in reels #1 and #2), some laid out as LCR sets, and others as full LCR, LS, RS and Boom elements.

'On a high-action movie like this,' reflects sound effects mixer Frank Montano, 'there is the temptation to make the tracks as dynamic as possible. But the end result can be a soundtrack that is exhausting to listen to. I prefer to leave space in the track for the audience to rest, and then build to the next dramatic action. You need to carefully choose where and when to add sound effects.' For the majority of reels, Montano prepared a total of some 20, 6-track predubs and sweetener tracks. 'While I might not need all of them all of the time in the finals, 'I'd rather be wide than tried,' he says.

Because the picture hadn't been locked and still lacked some opticals as the crew started predubs—composer Elliot Goldenthal was also completing the film score—the use of 6-track predubs allowed these elements to be edited straight across to accommodate inevitable picture and editorial changes prior to finals and print mastering of the 5.1-channel soundtrack.

Edited 6-track predubs mastered to 35mm mag were also transferred to a series of SoundStar DMS (Digital Machine-Room System) hard drives, for direct digital playback on the dubbing stage. DMS was also used to replay all elements for the predubs; SoundStorm's edited tracks prepared on digital audio workstations were transferred directly, D-to-D to the system's hard drives. The instant-access and enhanced playback fidelity of these new-generation 'digital dubbers' was described as offering a dramatic improvement over conventional mag dummies. A recorder version of DMS is also under development.

As effects mixer Frank Montano puts in most succinctly, 'Good sound is invisible; bad sound gets noticed. But if the movie has the name "Batman" in the title, don't sit in the first row!' ■

SoundStar digital machine-room system

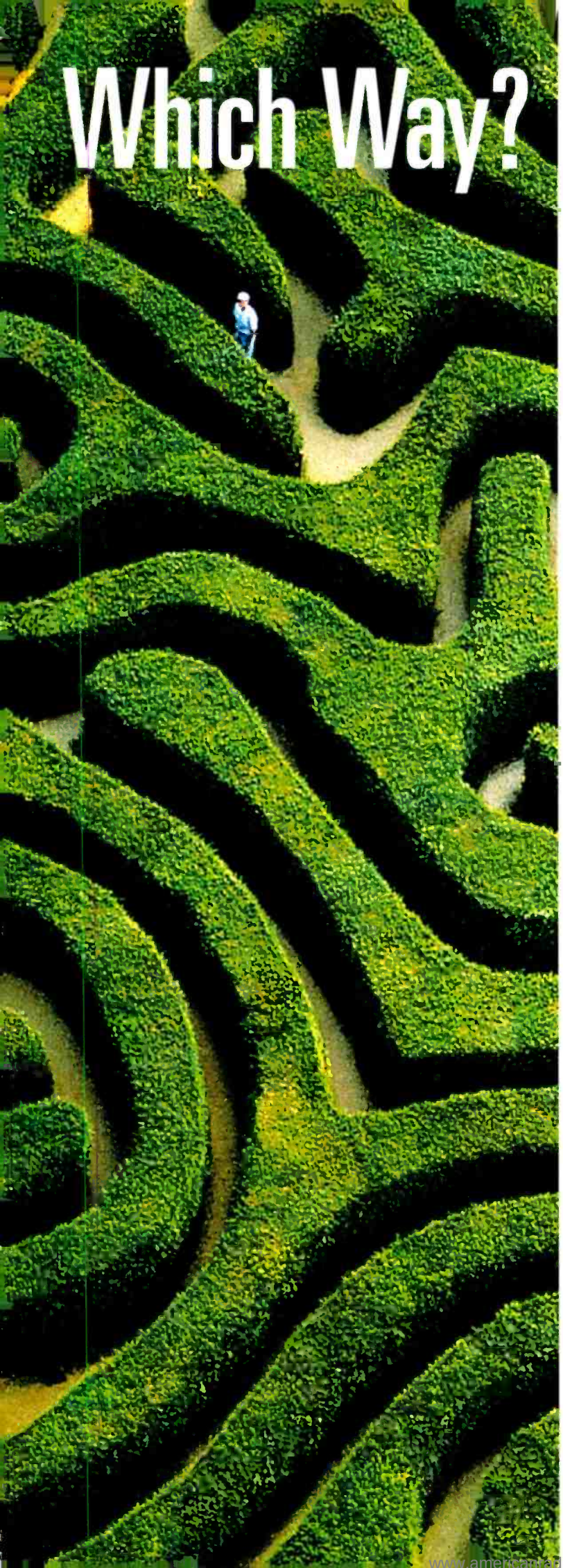
SoundStar was formed in 1994 to develop a digital record-playback system that would combine all of the advantages of mag film with the speed and expanded functionality of a nonlinear digital workstation.

A basic system consists of a Stage Interface (SIU), Master Control (MCU) and multiple hard-drive dubbers. A DMS system can be configured from a stand-alone 16-channel system to a virtual machine room of 1,500 or more, using hard disk or optical drives. The SIU handles all time code I-O and transport control from the console automation system via conventional 9-pin serial protocols, TimeLine Lynx II commands, or a Ketcham Box.

DMS is based on the Euphonix-Spectral Prisma Digital Audio Engine, with highly modified software that handles file management and disk interface. Spectral's Translator 8-channel digital audio format convertor handles Alesis ODI Optical, Spectral SMDAI, Tascam TDIF-1 and Yamaha Y2 digital I-O. Systems are configured for network operation from a single controller for a stand-alone use, or multiple control points for larger systems requiring access from the machine room as well as each mix position, using SoundStar's proprietary TCP/IP, Ethernet-based protocol; custom WAN configurations are also available.

System outputs can be quickly configured as mono or grouped as stereo, 3-track, 4-track or 6-track units. Once configured, the tracks can be advanced or delayed as a unit. Internal patching of tracks to outputs with the graphical user-interface and automated track patching speeds up reel changes, and is as easy as using a traditional patch bay. Output channels are routed through conventional patchbays, allowing DMS to be run transparently with other source machines.

Either 650Mb and 1.2Gb MOs are currently used to transfer material from editorial rooms into the DMS' hard drives, which are capable of handling real-time playback of eight channels per drive configuration. Subsequent versions will offer direct OMFI-compatibility, in addition to direct importing of standard digital-audio file formats.



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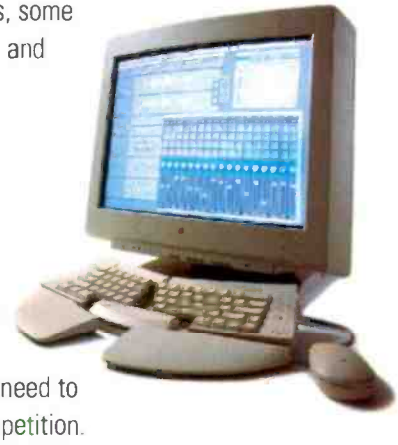

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SONIC SOLUTIONS

Recording Pavarotti

When Pavarotti took off for Barbados with a Decca recording crew, the location presented no distraction from one of the world's greatest tenors. **Simon Croft** discovers that life with a portable editor is on the beach

WITH THE POWER of today's laptop PCs rivalling that of desktop machines, it's perfectly possible to take an audio editing system like the SADiE and work on the beach. 'Another Pina Colada Sir?' says the bartender in the beach hut. 'No thank you,' you reply, perched pertly on a bar stool. 'At least, not until I've tidied up a couple more of these crossfades.'

Technically possible perhaps, but does anyone ever really work that way? The answer for editing manager Nigel Gayler and technical manager Roy Hankinson of The Decca Recording Company is 'Yes'—well almost.

They did take a SADiE portable audio editor to Barbados in order to play and re-edit sequences for the approval of Luciano Pavarotti. Editing actually took place in a private house, but Gayler, Hankinson and producer Ray Minshull did deliver an analogue cassette of the final edit sequences to the great man himself on the beach.

Given the stature of the artist and Decca's long involvement with audiophile digital recording (see side bar) the edit sessions represent a significant thumbs-up for PC-based editing in general, and the SADiE portable editing system in particular.

Gayler, who first cut his teeth on analogue editing with Decca 23 years ago, explains that while some artists are sent a tape for approval, Pavarotti prefers to set aside time to be with the editor and producer when the takes are chosen. With the artist's busy schedule, location tends to be a secondary consideration and editing has often taken place at the Pavarotti's home in Italy.

'With Pavarotti, we will go to a location with editing and playback equipment, so it's all done with him,' Gayler confirms. 'He was in Barbados on holiday after giving a concert at the Holders Festival. For the last five years it's been held on the island: it's an arts festival run by Johnny and Wendy Kidd.'

'The Kidds kindly offered us facilities in their house, so we set ourselves up in a bedroom and there was a dressing room en-suite and that's where I set up the SADiE portable

Recording details

Title: *I lombardi* by Verdi
Key cast members: Luciano Pavarotti, June Anderson and Samuel Ramey
Featuring: Metropolitan Opera Orchestra and Chorus, conducted by James Levine
Location: Manhattan Centre, New York in 1996
Producer: Ray Minshull
Assistant Producer: Morten Winding
Recording engineers: James Lock and John Pelloe
Postproduction and editing: Nigel Gayler editing manager and Roy Hankinson technical manager
Record label: The Decca Record Company.
Projected release: Summer 1997

Change for a tenor: Pavarotti takes a break from the *Three Tenors* tour to record in Barbados

editor. We had the bedroom as a playback room and the dressing room as an editing room.'

The rest of the setup was hired locally from Norman Barrow of A&B Music Supplies, who Gayler and Hankinson say was extremely helpful. The bulk of the system comprised two DAT recorders, an analogue cassette machine, a small Mackie mixer and a pair of active Genelec speakers. Talkback and monitoring for the SADiE consisted of a combination of opening the door between the two rooms and using headphones.

The other important consideration was acoustics, as Hankinson explains: 'We had to do some work to the room obviously, because it was a bedroom. We took all the furniture out of the room and, there was a wooden floor with high plaster walls, so we had to do some treatment. Fortunately, they had heavy drapes on the windows, so we could adjust the acoustic by opening and closing the drapes; and we hired some drapes for the other side of the room. We also treated the floor which helped to improve the room acoustics.'

The sessions in Barbados went exceptionally well,' says Gayler, who feels that the ability not only to edit, but also to play out alternative takes on-demand is a major bonus.

Roy Hankinson explains: 'Where possible, Nigel was using the SADiE portable to play back as well as doing the edit. It's a lot easier to cue up sections. On a DAT machine, the thing spools so quickly, you can't stop it unless you have an ident point.'

'There was a very good example where Pavarotti hits a top note in Act II and he wanted to hear the alternatives,' Gayler offers. 'We were able to play them immediately, which would have been impossible otherwise. So it's an extremely good playback format.

'The way we conducted the sessions, Pavarotti arrived on the Saturday and listened to part of Act II. He then listened to alternative takes and selected a new sequence with Ray and asked me to edit this new sequence straight away, so this is editing while you wait.

'I went into the dressing room and edited it up while he sat in his armchair and listened to the next piece. I was able to edit without disturbing him because I was using headphones. Roy was able to play the DATs of the next section and then when I had finished editing I would call out to Luciano and Ray and he would then listen to these edited sections and make his comments. At the end of the first day we had completed the Act II section and he had listened to his parts in Act III and Act IV.'

Pavarotti felt that there were preferable selections available for Act III and the team therefore arrived in advance of the artist to compile a new editing sequence.

'He didn't listen to that straight away, he wanted to listen to the Act II aria. He made one final change in that, listened to the whole of this aria again, and was pleased with it,' Gayler recalls.

'He then listened to this new editing sequence for Act III. He was extremely happy with that, and preferred it to the sequence we made at Decca's postproduction centre in London. He then proceeded to listen to the remaining sections, and was very pleased. No further work was needed.

'At the end of that, he requested an analogue cassette to be made of all the new editing sequences so that he could listen at leisure. We made the cassette and dismantled the equipment and went to deliver his cassette to the beach,' Gayler explains.

Hankinson picks up the thread: 'Pavarotti said his goodbyes to Ray Minshull the producer, who has worked with him for many years. Ray signed Luciano Pavarotti to Decca and came out of retirement for this, and some recordings which will be released later. Pavarotti said his goodbyes, and said that he was very happy with what we had done. We returned to London the next day.'

Having obtained Pavarotti's approval for the editing sequences, Gayler needed to make only small refinements once he was back in the superior acoustic environment of a Decca editing suite in London.

Everything had gone very well but then again, that was also due to work undertaken in advance of the sessions. After all, the team was going a long way from home with their precious material and editing gear. page 72 >

Below, left to right: Roy Hankinson, Luciano Pavarotti and Nigel Gayler

< page 71 'I had to do a lot of preparation here before we went out to Barbados,' Gayler confirms. 'I downloaded all the Pavarotti takes and edited masters onto the SADiE here (the DATs were used for playback purposes only) so I had everything available, not only on DAT but on hard disk. Had I not done that, it would have wasted an awful lot of time in Barbados. I took out two 2Gb SCSI drives, with everything on both.'

Another aspect of the preparation that went into the sessions was Gayler's experience with the SADiE portable system. Decca invested in the SADiE last year, with the specific intention of assisting the editing process with Pavarotti. In the event, Gayler did not have very much time before he would be using the system on location.

'We knew we would be required to go out somewhere,' he recalls. 'As it happened, we took delivery at the beginning of April of last year and on the day it arrived we were asked to go out to Russia three weeks later to do editing on the *Anna Karenina* soundtrack with Sir Georg Solti and the St Petersburg Philharmonic. So I had to learn the system quite quickly. It was very successful.'

Apparently, Sir Georg took a close look at this new way of working. 'I was editing close to the green room, so he was passing me every day,' says Gayler. 'He was interested in what we were doing because he had seen us editing on analogue in the 1970s, and was quite interested to see this computer.'

Hankinson says the SADiE was a 'major reason' the Barbados sessions were so successful and notes that 'It's very reasonably priced'. Gayler agrees. 'There is no question that I'm happy with it. It's reliable and it seems to be successful. Other editors have used the equipment and been very pleased with it.'

Decca and digital

Decca has a long history of development in digital audio recording equipment, including recorders, editors and mixers. The equipment Decca develops also helps to give sister companies Philips Classics and Deutsche Grammaphon a competitive edge. For this reason, Decca some years ago took the strategic decision that it would not licence its technologies to organisations outside the group.

Roy Hankinson has been involved in Decca's digital development from the start. He joined Decca in 1973 in the R&D department, working first on Teldec video discs and then designing the Decca digital audio recording system.

'We still have in-house development,' Hankinson confirms. 'The editing system that's used here was built in-house. We are about to start recording on our optical recorders, which were designed in-house. We're just finishing a new set of software for that.

'One of the things we find is that because we are a location recording outfit, we are not quite the same as most studios. Our equipment has to fit different parameters. The market doesn't often provide equipment that fits our uses.

'Complex mixers are usually quite big. They are either a big mixer or a small mixer: they're not modular in construction. The philosophy behind in-house design is that first you look outside and if it's there, you use it. Otherwise, if it's required, you design it. There's always an area where equipment is required to interface with other equipment, so you find you need a small design department.

'We are still using the digital recording system that was developed in 1978. That was developed as an 18-bit, 48kHz system, but by the end of this month, we will have a 20/24-bit, 96kHz, whatever you like, system. It's a 4-track 20/24-bit 48kHz recorder, which, of course you can use for two tracks of 96kHz if you want to.

'Resolution greater than 16 bits is required in order to deliver a true 16 bits to the consumer. Higher resolution is needed because each time the signal is processed it is degraded. We record using 18-bit or 20-bit word lengths and use a minimum of 24-bit resolution within the postproduction equipment. A programme might be copied 10 or 11 times before it actually gets out to the consumer.'

Hankinson is cautious about claims for 96kHz sampling; although the new Decca optical system supports this sampling rate.

'We're always interested in anything that improves audio quality,' he says. 'Certainly with DVD coming along, there is the possibility to issue 96kHz recordings; that is one of the things we will look at. It's not clear at the moment that there is a vast difference between 48kHz and 96kHz; although some people claim there is.

'The most critical area of digital equipment is the analogue to digital convertor. As far as Decca is concerned, we sell digits, because we sell programme on CD. Once it is converted to digital, it never goes back to analogue. That has been the case here for 17 years. Our production has been totally digital right from day one.

'You have got to be careful what you do digitally. You have to understand the processes that go on, but the convertor is the most critical element.

'One thing about 96kHz that is an advantage is that you can have a much shallower filter. We've always worked at 48kHz, from the early days because we knew that that would fit with the television systems. Also we felt that it gave us a little area to have a less steep roll-off. We didn't need such a brick wall filter—well, we could angle the wall slightly.

'Some of the major problems early on were the convertors. Some were not even near 16-bit, and some people were still using analogue vu meters and therefore did not monitor the signal properly, hence tended to record at a lower level than they had to. They were giving themselves headroom they didn't need. There are probably quite a few 12-bit recordings out there, but I wouldn't like to say whose!'

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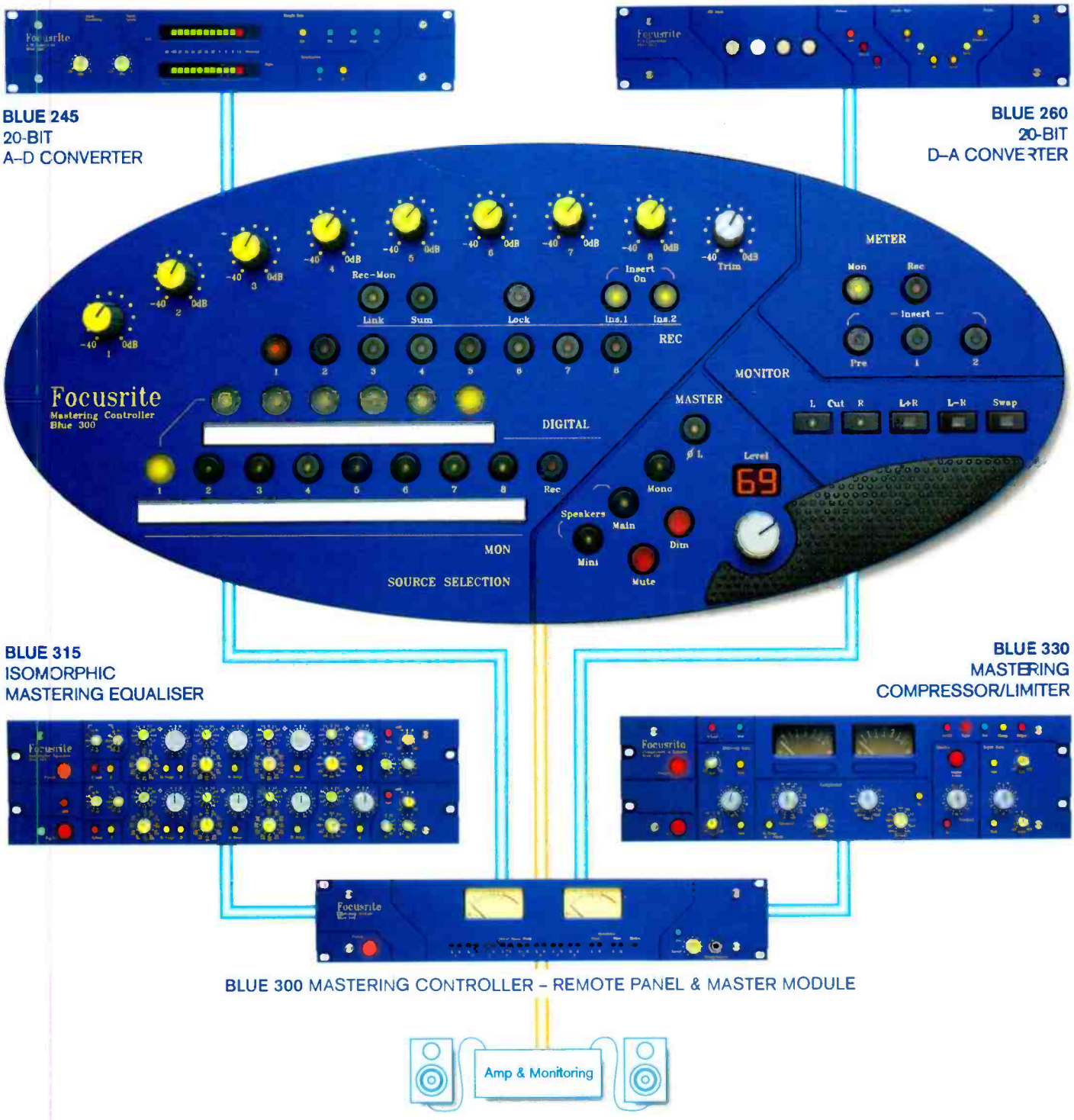

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Stayin' power

Relatively few people have been inside the studio that's been the Bee Gees' home base for some 17 years. **Dan Daley** drops in on Miami's oldest recording facility and one of the world's first personal recording studios

I AM SURE that it was a coincidence that 'Stayin' Alive' was blaring from the Tannoy FSM monitors the moment I walked through the portals of the control room at Middle Ear in Miami. Practically, Studio Manager John Merchant is far too guileless to have set something like this up—soft-spoken, genial and pleasant to the point that you wonder what he's doing in the recording studio business. Merchant is running the multitrack of the 21-year-old classic and the four-on-the-floor groove that helped define disco a generation ago is more than intact—it's eerily contemporary. And just as John Travolta has undergone his own renaissance in film in recent years, the Bee Gees are back with a vengeance. The second break Merchant takes during our visit is a call from the chart trades in London to tell him that 'Alone' the group's first single from its new *Still Waters* album, has just hit No.5 in the British charts and No.15 in the German listings in its second week out. America, he nods knowingly, is still to come.

The reason that the track is rolling is purely as a matter of commerce: telemarketing label K-Tel Records has requested it as the new theme song for its next round of disco commercials. But its presence over the speakers underscores the history of Middle Ear Studios as one of Miami's oldest recording facilities, and as one of the world's first personal recording studios.

Middle Ear opened in 1980 after the Bee Gees followed their RSO Records stable-mate, Eric Clapton, to Miami in the mid-1970s to record. 'The Boys,' as Merchant sometimes refers to them, were charmed by the city and had moved there permanently in 1977. During preparations for their 1979 *Spirits Having Flown* tour, they rented a large warehouse, owned at the time by TK Records, home of KC & The Sunshine Band, located a mile from the South Beach area and converted it into a stage for rehearsals for the tour. By the time they were finished rehearsing, they realised that the location and building were perfect for the studio they had been considering building in Miami. During the tour, their co-producers at the time, Carl Richardson and Ahlby Galuten, oversaw the design of a large-space, single-studio facility within the building. Assisted by Seth Snyder and the Recording Studio Equipment Co of Miami, the studio was ready for the Bee Gees upon their return. The first project done there was the mixing by the Bee Gees for Barbra Streisand's Guilty.

Originally an all-MCI facility with a JH-24 multitrack and an MCI 500-series console like many studios in Miami (where MCI was founded) at the time, the studio implemented a 56-channel Neve V3-series console with Flying Faders automation in 1985 along with a redesign of the acoustic space. The slatted-wood-covered walls are dotted with indentations into which amplifiers and microphones can be positioned to create a wide diversity of niche space ambiences. The acoustical qualities of the sizable—28 x 38 x 12 feet with a 6 x 6-foot isolation booth—main recording room can be augmented by opening doors at the rear that lead into an all-concrete room that is cabled to the 20 x 18 x 12-foot control room for additional reverberation. Other equipment includes a pair of Mitsubishi X-850 with Apogee filters, an MCI JH-24 24-track,

a Studer A820, ½inch, 2-track, Ampex ATR 2-track; Sony 1610, Sony and Panasonic DATs, two each AMS RMX and AMS DMX, an Eventide H3000S, a Lexicon 480L, three PCM70s; dbx, Urei, Teletronix and Avalon compressors; Aphex and Drawmer gates; an API Lunchbox; a Focusrite stereo mic pre, and monitoring is via Tannoy FSMs with custom subwoofers, Meyer HD1s, Yamaha NS-10s, Tannoy PBM-8s, and Auratones. The studio offers a large instrument collection including MIDI gear, a 9-foot MIDI Baldwin concert grand, vintage guitars and amps, and a large selection of Neumann and AKG microphones.

The Bee Gees used Middle Ear exclusively for their own projects and productions they were involved in until 1993, when they decided to make the studio more widely available. 'In a very real sense, the studio has been broken-in in the best way a studio can be,' says Merchant. 'The boys developed the studio into a warm, personal and creative space. And that is part of what comes with the studio. There's nothing cold or impersonal about it. It's a single-room facility, so when an artist rents it out, its all theirs. And everything comes with it; there are no additional rental charges.'

Those who have taken up that offer include Extreme, R Kelly, Arturo Sandoval, and Capitol Records artist Nil Lara, as well as a number of projects by leading Latin artists, including Ilan Chester, Ricardo Montaner. Plus, the Bee Gees continue to use the studio for their own recordings and production projects, which have included a Kenny Rogers-Dolly record, Diana Ross co-produced by Michael Jackson, and a Dionne Warwick album.

The most notable recent project is *Still Waters*, which not only marks the Bee Gees' return to the pop charts but also their first multi-producer outing—tracks were directed by Hugh Padgham, Arif Mardin, David Foster and Russ Titelman. And Middle Ear was not the only studio used—various facilities in New York (Right Track and Unique), London (The Townhouse), Los Angeles (Record Plant) and Miami (Criteria, the first time the Bee Gees had been back at that studio, where they had recorded so many of their hits, since 1980) were also visited. But Merchant, a native of

Virginia who came to Middle Ear while working on a graduate thesis at the University of Miami's audio programme, and who describes himself as chief engineer, studio manager, 'and I still change light bulbs', says that the Miami studio was always the central clearing house creatively and operationally for the project.

'More and more of our work is being done over long-distance lines using the Dolby Fax system,' Merchant explains. 'We'd be cutting vocals here with Russ Titelman working up at Right Track Studios in New York. The boys would be listening into the mixes in real time as they were being done there, too.'

The system was put into Middle Ear about halfway through the project; its benefits became apparent when the schedules of the various producers were requiring the constant FedExing of DATs tapes back and forth from state to state.

'They'd get the DAT from Arif, who had finished the night before in New York, listen to it, make comments, phone them into him and then start that whole process all over again,' recalls Merchant. 'It was taking three or more days to get mixes finalised and approved. With the Fax system, they were coming directly out of the digital outputs of Right Track's Capricorn console, so the signal was staying digital until it got to the convertors here. It made a huge difference, and ultimately I think it gave them better mixes.'

Despite the peripatetic nature of the recording, Middle Ear remained creative central for the entire project. Merchant engineered the demos of the songs that Barry, Robin and Maurice wrote at the studio, all of which served as frameworks for the final tracks, and in some cases provided elements of those tracks, such as lead and background vocals. 'That's the nice thing about them having this place,' he observes. 'They're very comfortable here and that shows in their vocal performances. They can really relax when they're doing a take.'

Recording a Bee Gees vocal is a pretty straightforward affair, Merchant says. 'Typically, we set up a single microphone—a Neumann U87 or this restored M49 that we now have—and they group around it and balance themselves for harmonies. For the first single, 'Alone', Robin's chorus parts came from the demos we did here.'

When composing, the Bee Gees prefer to play and sing simultaneously as though it were a vintage recording session, with Barry on acoustic guitar and Maurice on keyboards, each with their own microphones and with Barry using a Futuresonics in-ear page 76 >

< page 79 monitor so he could move over and adjust the drum machine as they worked. It was a comfortable way of working for them, harking back to the way they wrote and recorded years ago.

'They pioneered the notion of the drum loop,' says Merchant. 'When they were working on 'Saturday Night Fever' in France their drummer had to leave in mid-project, but they still wanted grooves to write to. So they took a 4-bar piece of drums off the 2-inch machine and cut that into a physical drum loop. The trouble was, it was too long to fit properly in the tape path of the machine, so they used a broomstick handle to extend the tape path. The intent was just to use it to compose to. But in the end I think that loop actually became part of the drum track for 'Saturday Night Fever,' as well as several others like Barbra Streisand's 'Guilty', and on the first record of theirs that they cut here, 'Children of the World.' That's what got them so deeply into playing with drum loops and later with drum machines. It got to the point where they would literally spend a week on a kick drum pattern. It also got them more deeply into the notion of producing and engineering themselves. It opened up their personal vistas as producers, and it was at that point, along with opening up their own studio, that their careers as producers blossomed.'

Keyboards and relaxation

It was a propitious moment, since shortly thereafter the Bee Gees became embroiled in litigation with their then-record label, RSO Records. While their own career as artists was put on a hiatus, they moved into outside productions for Streisand, Kenny Rogers ('Islands In The Stream') and Diana Ross.

Some years later, the studio started becoming occasionally open to to outside rentals, and that led to upgrades in technology and design in the mid-1980s.

'The people who have used this studio outside of Bee Gees tend to like it because of its privacy and its collection of vintage gear, like the Mellotron we have,' explains Merchant. 'And everything comes with the rental, we view the studio as an organism, complete the way it is.'

It's a tricky room to book, however. The Boys still get first crack at it. And even tangential projects, like the K-Tel remix of 'Stayin' Alive', takes priority over an outside booking. (Although the Dolby Fax and ISDN lines have increased the studio's potential as a commercial facility—that same week Sylvester Stallone was scheduled to come in for a voice-over on a film project.) Discussing a track sheet that he seems to have memorised, Merchant notes that the 24-track master from that song had the drums on two tracks and two tracks of automation data, with everything else sandwiched in between. 'The entire drum track was on only two tracks,' he says. 'When you listen to it, you hear their impression of what drums should be and it was quite different for the time. They were the first to really put the kick drum out in front. Up to that point, people were still mixing for AM radio, which couldn't handle much of the low-frequency information. But with disco coming in, their drum mixes to multitrack helped define how that music would sound.'

Also on those tracks one can hear a combination of real strings and early analogue synthesisers, such as the Prophet 5, and a Rhodes electric piano all blended together. You can also hear things that no one has ever heard, such as the brass parts that were replaced in the second chorus by the 'life goin' nowhere' vocal parts. There are also numerous instances of vocal ad libs scattered across the tracks, what Merchant calls 'track salad', that were combined via automation.

Middle Ear will undergo another redesign sometime in the near future, Merchant says. There is no sense of urgency to that project, which reflects both the laid back nature of the studio and of Miami itself. But the genesis of that does come, interestingly, from The Boys' experiences in using multiple studio and producers on this most recent record. 'The experience of using other studios seems to have had an effect,' Merchant says. 'Especially at Record Plant in LA, which really sounded amazing. This record had a real eye-opening effect on them.' No designer has been retained yet, but Merchant says it's likely that they may go for a 'name' this time. Other recent enhancements that bring Middle Ear forward include the implementation of black burst video synch.

'If this Bee Gees record is a monster, then I'd expect that we'll be having more enquiries about this studio,' Merchant figures. 'I don't know that it's going to be able to be rented any more than it already is, which isn't that much. But if that happens, then we want to be ready for that. But mostly, the upgrades we've done here over the years are to make the place more functional for The Bee Gees themselves. It's their studio at heart. And it's nice to hear them and this studio back on the radio together.' ■

Contact

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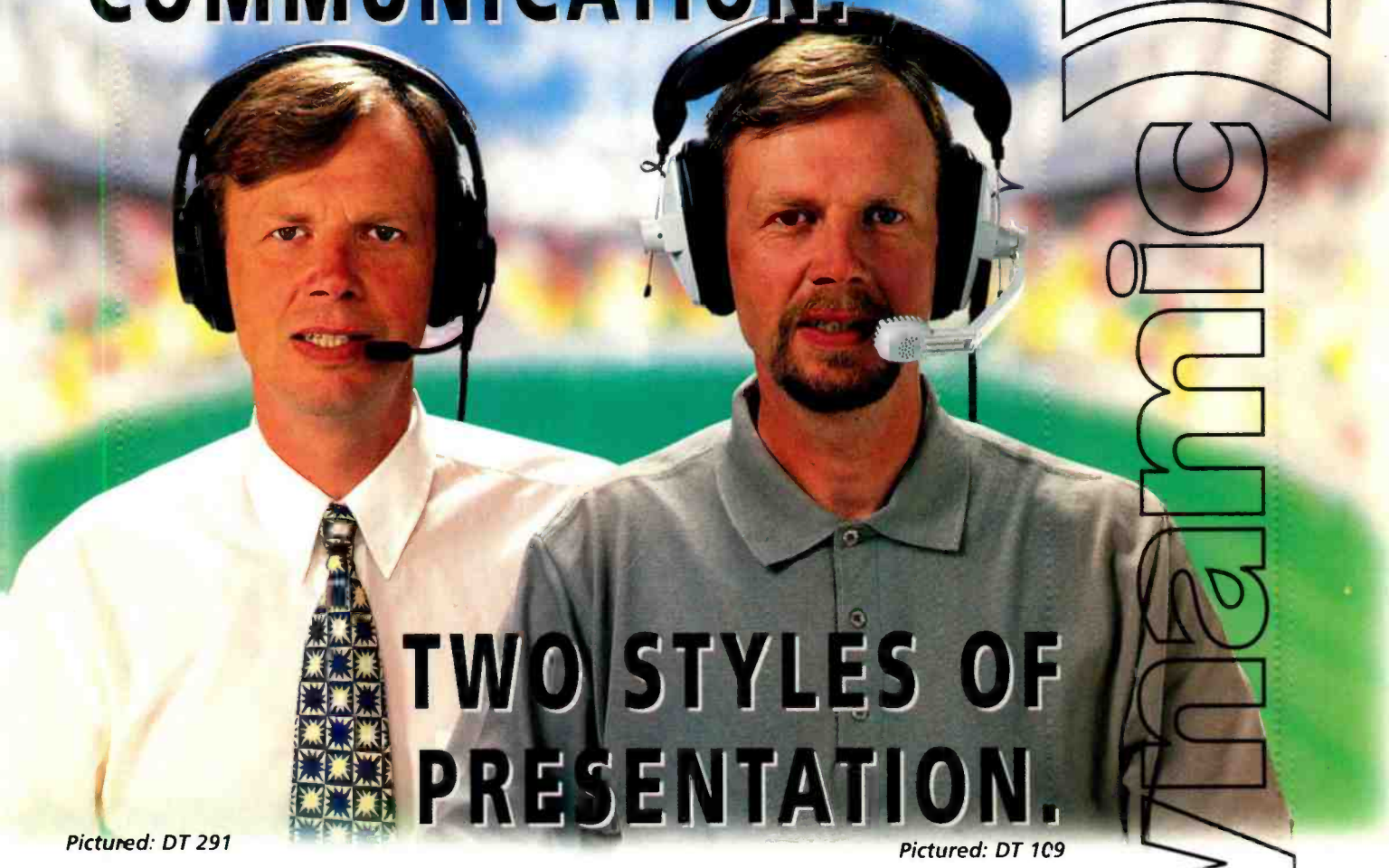

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Gimme Five

Some 15 years after the launch of Channel 4, the UK has acquired another terrestrial television channel. **Kevin Hilton** investigates the kit and controversy of C5

ON 30th MARCH, Channel 5 finally lived up to its name and became the UK's fifth terrestrial TV channel. This after several false starts and continued doubts about the validity of such a service, competing against the two public service stations, BBC1 and BBC2, the commercial ITV network, the more specialist commercial Channel 4 (S4C in Wales) and a host of satellite and cable services.

Regardless, C5 launched in a fury of publicity, posting advertisements around the country, pushing its retuning capability and dragging in the already over-exposed Spice Girls—on the grounds that they are a powerful symbol of five—for the opening ceremony. This just gave the press something to bash the new arrival with, homing in on C5's high proportion of bought-in material and the trouble some people were having receiving the signal.

What everyone seemed to miss was that a fifth service had launched at all, given the idea was put forward towards the end of the 1980s and has taken this long to come about. One person who remembers the initial plans is Chris Collingham, now controller of engineering and operations at the new channel.

In 1987 he attended an Independent Broadcasting Authority committee meeting in his capacity as chief engineer of TV-am, the then national commercial breakfast TV contractor. The idea for a fifth channel was mooted then and was eventually included in the 1990 Broadcasting Act. Collingham, whose CV also includes the BBC, left TV-am in 1989 to become a project management consultant and then became technical director designate for one of the groups interested.

This consortium did not bid when the Independent Television Commission (which replaced the IBA as regulator and licensing body for UK commercial TV) advertised the licence in 1992. Only one group, C5 Holdings (Thames TV, Pearson TV, Time Warner, Associated Newspapers and City TV) put in an offer, which was rejected by the ITC on the grounds that it did not think that the bidder could sustain a service of acceptable quality. 'The rest of us said to the ITC that a fifth channel would work one day, but not with the plan put forward at the time,' Collingham recalls.

The cause of concern was the frequency slated for the proposed fifth TV service, because whatever was suggested could cause interference to domestic VCR machines. As Collingham says, the aim was to reduce the amount of interference from the beginning. Channel 36 was initially offered up, but this was seen as unsuitable.

Collingham explains, 'If the original plans had gone ahead, the transmission polarity would have been opposite to existing services, vertical polarisation as opposed to horizontal. This would have reduced interference but meant that virtually everyone would have needed a new aerial.'

Most experts regarded Channels 35 and 37 as better bets, but when the relevant authorities came up with a second plan—Channel 37 for nine of the main sites plus Channels 48 and 56—Channel 35 was not included in the package as it had been reserved for a proposed BBC satellite single frequency network.

In 1995, the current Channel 5—a consortium of CLT, MAI, Warburg, Pinus Ventures and Pearson—won the tender race, although there was a judicial review as it had bid the same amount of one of the losing page 80 >

< page 79 contenders, the Virgin Group. The licence was eventually confirmed and, after a process of lobbying, Channel 35 was released for use by the new service, as it was deemed to be inefficient for a single frequency network.

In the build-up to its launch, C5 made viewers aware of the potential need to re-tune their existing equipment, fighting off criticism of its retuning programme, which was knocked for employing non-qualified staff. Although it was clearly stated from the beginning that the new service would not reach the whole of the UK population, C5 has also had to make provision to provide as much coverage as possible.

'We've got the retuning programme up and running and in place,' Collingham responds. 'The Department of Trade and Industry has now agreed to the use of Channel 35 for at least the next five years—they're talking about using it for possible digital mobile TV on trains and coaches—but we will have re-tuned by then. We're building another nine Channel 35 transmitters at the moment and they should be ready by the summer. That will bring us up to about 80% coverage terrestrially.'

This 80% is conventional UHF transmission, while the rest is a mixture of satellite feeds. C5 is now on Astra, and distribution through cable head-ends. On its terrestrial network, C5 has been accused of moving away from the policy of co-siting, where broadcasters transmit from one antenna for a given area, thereby doing away with the need for multiple domestic aerials.

'Channels didn't have to co-site before we came along,' Collingham replies, 'but they generally did. With some of the existing sites it's not physically possible to get another antenna on the mast, for example Sutton Coldfield and Crystal Palace. In both cases NTL [the terrestrial and satellite transmission service provider] said that it had another mast, the old VHF site, only five miles away.'

While C5's London transmitter at Croydon is only a matter of 3° away from Crystal Palace, which houses the other terrestrial channels, there are still complaints that the new service's pictures are prone to the snow effect.

'In London, for example, most people should be able to put up the worst of aerials and get away with it because it's only 6dB down on Crystal Palace,' Collingham explains. 'There are other problems with aerials. Some are not designed to cover up to the frequency we're on—for example a Group A aerial goes from Channel 20 to Channel 34. We're on 37 and already 6dB down so an existing aerial may be 15–16dB down. Newer aerials extend to 37; Group Bs only fall off at 38. When it comes to it, it was pretty difficult to get a fifth channel in. During the 1960s the UK allowed for four channels, which was pretty far sighted; most other European countries only went for three.'

Encouraging viewers to buy proper aerials is all C5 can now do, having made the best provision for transmission and reception that it could. At the same time it had to prepare its play-out centre ready for launch. As a publisher contractor as opposed to programme making and production licence holder, C5 required a transmission facility rather than a full studio complex.

Collingham observes that such page 82 >

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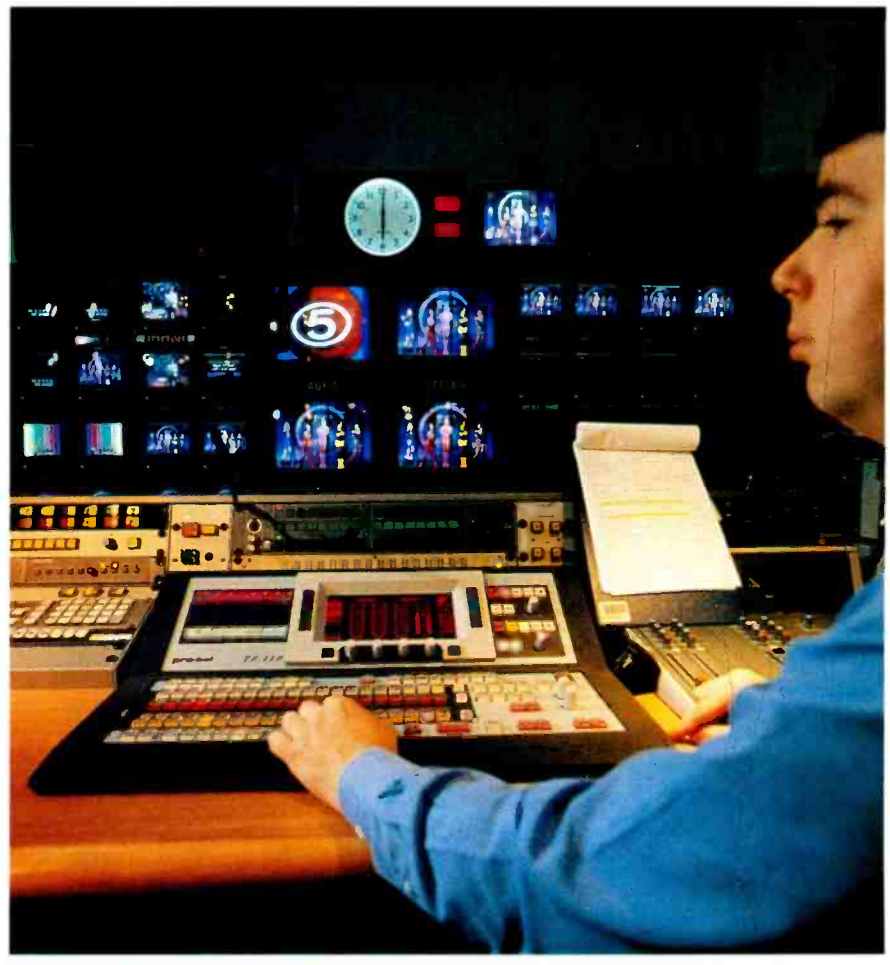

Getting five; switching on Channel 5

< page 80 decisions come down to cost. 'We decided to contract out our programming as much as possible,' he says. 'So ITN (Independent Television News, which already supplies ITV and C4) makes our news programming [which, unusually for TV, has bulletins on the hour every hour, as well as main programmes at specified times) and other live shows are produced by various companies, while the rest is bought in from independent producers.'

The play-out centre is based at the new Pearson Television building in the West End of London, housing Pearson's own studios, which have been moved up from the old Thames TV centre in Middlesex, as well as new technical facilities for other publisher broadcasters (including Disney and Discovery). Collingham says that Pearson being one of its parent companies was not necessarily a factor. 'We looked at a number of facilities companies before settling for Pearson,' he comments. 'It was a brand new building but had it still been in Teddington we wouldn't have gone because we wanted a central London facility. Now we're only a few minutes walk from our programming and administration offices.'

Cost-effectiveness also defined the nature of the play-out centre. 'It meant that the facility had to be highly automated,' explains Collingham, 'and serial digital throughout, so there was no lining or drifting on the output. It either works or it doesn't. Also, from day one it was a 24-hour service with four separate outputs [each one corresponds to as specific region for more directed advertising], so we had to be careful how it was built. We also wanted to run with as few people as possible.'

In putting this together, C5, like other UK commercial broadcasters, is bound by its licence to provide a reliable service with good quality output and programming. 'The ITC had to approve the installation,' Collingham continues, 'ensuring that it meets the Code of Practice. Since we're not making programmes, we have to check everything that comes in to us from our suppliers. If we do it ourselves, we know we can trust our staff, but in the long term we need to trust the independent producers as well.'

While the independent programmes are checked in-house in two dedicated technical viewing rooms (where technicians look for colour balance, sound quality and overall production values), acquired material—TV movies, feature films, overseas programming—is checked by the ITFC facility, which started out as the technical checking centre for the ITV Network, but is now a stand-alone company. It is also here that films and programmes are edited or graded for content and panned and scanned.

The main transmission suite is where programmes are played out and the overall output monitored. Central to this is a Pro-Bel TX220 vision mixer, which handles the serial digital signal output as a single stream (with embedded audio). Around this unit are three stand-alone Sony digital Betacam VTRs. There are a further three such units in a FlexiCart automation loader, but in general programmes are loaded into the stand-alones, because if there are any failures it is would be easier to swap machines. Also built into the page 83 >

< page 82 control console is a small Soundcraft audio mixer to bring up the continuity announcer's microphone. For overnight services the announcer will prerecord all the necessary links onto MiniDisc. Two Denon MD players and the output of the Soundcraft are taken into the TX220, and then digitised.

Live programmes, including the news and the afternoon show 5's Company (produced by Pearson subsidiary Grundy TV in the downstairs studios), come in either as land lines, line links or from circuits via the Telecom Tower if they originate outside London.

Other important inputs are the commercials, which arrive at C5 either on digi Beta or, more usually, down feeds from various agencies. Technicians record these onto tape, top and tail them and check them for quality. The clips are then recorded onto tapes already loaded into a Sony LMS (Library Management System) 1000 machine, with the information relating to each commercial entered into a database. 'It's the only way to know exactly what is in the system,' observes Collingham.

From here the commercials are cached onto Tektronix Profile disk play-out machines, with one assigned for each of C5's four 'regions'. Both the commercials and the main presentation output are under automated control, with Pro-Bel's two specific software packages being used. Compass for programming and MAP for the advertisements and caching. Compass runs the majority of the output and instructs MAP when a commercial break is scheduled.

The output of the main transmission suite —programmes, promos, captions, voice--overs—is split to four small Pro-Bel 2-input vision mixers stored in the equipment room, one for each region. This is fed into one input of each submixer, while the relevant Profile player takes up the other. 'The main programme feed goes straight through,' outlines Collingham, 'but during a break, the submixers switch to the Profiles to feed the break. At this time the main mixer is doing nothing, but it does stand-by with captions in case of failure.'

This arrangement was decided upon to keep down the amount of equipment used and for easy expansion. 'As each of the four sub-mixers has its own Profile, we could just add on if we ever wanted to go to more regions,' says Collingham. 'It would have been unmanageable with four big presentation mixers, and then trying to add a fifth or sixth.' Although this is a high level of automation, there is still a human over-ride, as Collingham acknowledges: 'The transmission controller is ultimately responsible for it all happening. They have to make sure that the right tapes are in the machine and that the correct feeds are selected.'

Downstream of the four vision sub-mixers is a Teletext inserter for subtitles and data services. A feature of C5's ad breaks is a colourful station ident that pops up between each clip; this 12-frame animation is generated on a Leitch device and is inserted as a cached trigger.

Like the transmission suite, the other technical facilities used by C5 are leased from Pearson. These include a graphics area with a Quantel Paintbox, rostrum camera, two Macs and caption generator; two edit suites, one a digi Beta room, the other an off-line, nonlinear facility with a Quantel Editbox; and the technical checking areas. Pearson also offers a stand-by transmission room when the on-air suites require maintenance and is currently building an audio dubbing booth.

In C5's equipment room the four outgoing feeds are prepared for distribution. The channel is confident that it is the first terrestrial TV service that distributes its signal to its transmitters using satellites. One set of two is sent to an MPEG compression system at the nearby NTL Oxford Street Teleport and thence to the Orion satellite. The other feed goes to a compression hub in the basement of the NTL building and then onto that company's headquarters in Hampshire, from where it is sent to Intelsat 725. Both feeds are sent out over cable links.

Collingham explains that this split serves two purposes: any one of the four regions can be isolated if the station wishes to target a particular area of the country for marketing reasons, while there is redundancy built in if one of the uplinks were to fail. It also means that all the regions can be transferred onto one satellite if the other is about to experience a sun-out (when the bird is caught directly between the sun and the earth).

Like any new service, C5 has taken its fair share of knocks since the launch, but it is consolidating and looking towards the future. Being so new, it has seen the proposals put forward for digital terrestrial TV and planned accordingly. 'We've been gifted half of a multiplex,' says Chris Collingham, 'which means we have 12Mb to use, some of which will go on simulcasting our existing terrestrial signals. Because we're serial digital, we could take the existing service as it stands, put it into another multiplexer and go. All we've got to do is build another 50 transmitters and, ultimately, it depends on people buying the new receivers and aerials. The frequencies will be within the existing aerial groups, but there will still be people out there with inadequate aerials.'

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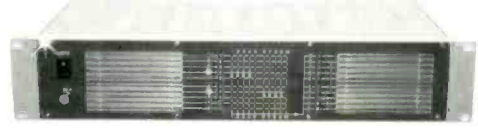

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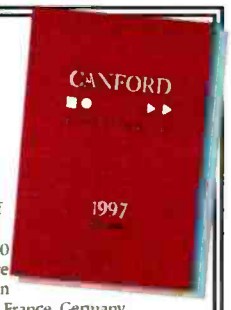

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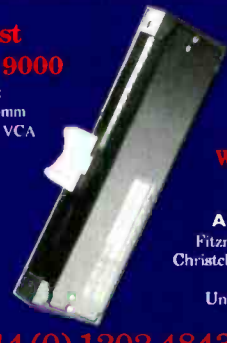

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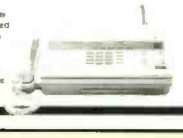

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US: Post-it notes

With punishing overheads and a declining market, the temptation is strong to jump ship from music recording to postproduction. But the waters between are deep and dangerous writes **Dan Daley**

There has been considerable record industry concern in recent months over the flat performance of the US music market. Last year's growth was pegged at less than 1%, and the first crests of what could turn out to be a wave as high as 40% of retail returns of independent labels could ultimately make 1996's true net a negative one.

What this upheaval has obscured, though, is the robust health of postproduction. Evidence of acceleration and advancement abounds: Todd A/O is adding new stages and is reportedly looking to add new geographical locations; Creative Cafe, a venture started by Record Plant owner Rick Stevens and Academy Award-winning sound designer Steven Hunter Flick, just opened its third location in Los Angeles; several high-end, high-profile studios, such as Bad Animals in Seattle, have made visible transitions from music recording to post in recent years; Orlando, Florida, which once threatened to become the White Elephant of post when it was realised that technical talent—the key element in all post-production—was reluctant to leave California and move there in the 1980s, is now a busy place indeed, with Universal and Disney expanding their operations there and several independent post facilities thriving.

Like most of the trends that have affected virtually every aspect of professional audio in recent years, the prime motivators behind the burgeoning of audio post come not from within but without. The expansion of cable systems in the US and their voracious appetite for programming has put the already production-line nature of Hollywood onto a virtual war footing, churning out movies-of-the-week and features, some of which were never meant to linger long in the cinema before shooting over to slots on HBO and Star TV. Not that Hollywood was lacking for work already—foreign distribution and rentals now account for more revenues for some major American-made films than their US showings, and overseas markets are hungry for America's most visible export: really big, really loud movies. (*Batman Forever* was mixed at 112dB.) The home video market rivals the cinema in its demand for programmes, and its catalogues are swelled by compilations of anything from rock videos to best-of shows to the peculiarly American obsession with replaying World War II, which on video is now longer than it was in reality. Finally, the needs of an incipient 500-cable future has meant that as fast as post houses seem to be expanding, it's not fast enough; more audio scoring and post is being sloughed off onto project-type studios.

> Costs to keep technical post talent are spiralling upward as the number of new facilities and suites grows

In addition to lots of work, audio post facilities can take advantage of the economies of technology more readily than can music studios. Post suites can proliferate and go on-line quickly with GUI-based DAWs that cost a fraction of what large new consoles and scarce vintage equipment can cost, and post suites' acoustical design requirements are often less intense than those of large music studios.

And when high-priced technology is needed, such as 100-plus-input mixing consoles, the 3-digit hourly rates that many post houses can charge helps defray those costs. Considering that many well-equipped music recording studios often squeak by on 3-digit daily rates, and how those same studios need to keep adding new levels of technology to beat off the hordes of home studios that are nipping at them from below, it's a wonder that any of them stay in music at all.

However, rosy as postproduction sounds,

Europe: Welcome to DVD, the digital volatile player

DVD's international soap opera continues—integral to the development of the plot are subversive issues including copyright, increasing technical complexity and the growing political consequence of failure writes **Barry Fox**

DVD-MOVIE in Europe remains an extraordinary mess and a serious risk for any company investing in software production. Of all the companies that promised to launch DVD Movie in Europe this year, only Panasonic is pressing ahead. Panasonic's players are on sale in Germany, Denmark, Spain and France, with sales in the UK promised by June. But this is despite the fact that there is just one, yes one, PAL disc available (*Twelve Monkeys*) and it has an exclusively German-language soundtrack.

The strength of software that will play on European players is on NTSC discs from Japan which falls into the same Regional Code zone (2) as PAL Europe—but only 12 of the Japanese discs have English language sound. NTSC discs from the US will not play because they are coded for Region 1. Under British law anyone selling them risks prosecution because they are not certificated (or rated) by the UK censors.

Back on the hardware, the Panasonic player connects only to an AC-3 Dolby Digital surround decoder, not a decoder for the MPEG-2 digital surround system required by the DVD standard for PAL discs. All the other companies (except Thomson which is badging Panasonic players and using them for in-store demonstrations of wide-screen TV) have now shelved plans to launch DVD in Europe until the Berlin Funkausstellung at the end of August. They are waiting to see if there is any software available and, if so, what digital surround system it uses. If it uses MPEG2 surround, all the Panasonic players so far sold will be obsolete; if it uses Dolby AC-3, the DVD standard is obsolete and MPEG2 surround is dead in the water. Either way, early commitments to DVD programming in Europe risk being suicidal.

As one trader put it, there is only one real hope for DVD, and that is to put a record button on the player and make it a disc-based VCR that plays pressed discs as well as its own recordings. This is now possible.

As expected, the Working Group setting the standard for DVD-RAM voted in favour of the compromise brokered by Hitachi (the WG Chair) between the rival proposals from Panasonic-Toshiba and Philips-Sony.

The DVD Forum now defines the new erasable disc as coated with phase-change recording material, pre-grooved with a wobble tone to provide a clock signal, and pre-pressed with bursts of pits which provide address headers for the erasable data. Although the pressed pits are robust, and do not degrade with use, the fact that they are in short bursts makes them susceptible to any blemish on the surface. Hence the DVD-RAM standard provides for caddies to protect the disc. And here things get very confusing.

The DVD-RAM standard says that caddies are optional for single-sided discs but obligatory for double-sided discs. This raises two questions: how, as the standard suggests, can a DVD-RAM be played on a standard DVD-Video player if the disc is in a caddy and DVD players use tray loading for bare discs? And why should it be necessary to protect the double-sided discs but not single-sided discs?

Simply the promise of DVD-RAM playback on standard DVD players is misleading. Owners of first-generation DVD-Movie players and DVD-ROM drives will not be able to play DVD-RAM discs. The optics will probably not read the low reflectivity disc and certainly the tray will not physically be able to take a disc in a caddy.

What the DVD Consortium should be saying is that they have seen ways of making future generation DVD-Movie and ROM players capable of playing back DVD RAM recordings. Whether this happens, on what timescale and at what price and whether all

it's not a piece of cake. Costs to keep technical post talent are spiralling upward as the number of new facilities and suites grows, creating a peripatetic class of increasingly affluent and itinerant mixers, sound designers and engineers. Music studios learned long ago the pitfalls of building rooms to suit the tastes of one major client; a Nashville studio once built a room for producer Jimmy Bowen that had no wall between control and recording rooms, a situation that ultimately gave Bowen, not the studio, the most leverage. But that's precisely the model of post, in which the suite is built around the technical talent, who is, in essence, the client. So while the technology may cost less than in music, that's being offset by personnel costs.

Secondly, the same personal studio issues that plagued conventional recording studios are making themselves felt in post as affordable technology encourages more small but very capable facilities and those increasingly affluent technical talents go off and open their own facilities. In addition, as the number of suites grows, so does competition; there are reports of, if not the wars seen in music, then at least rate skirmishes as new post facilities vie for clients.

In short, audio post is on an upward swing. But it is becoming, as did music recording before it, much more complicated as a business. The US postproduction industry is the world's most evolved and will be the playing field upon which future paradigms for that business are cast.

manufacturers make the transition, is an open question.

The issue of caddies is one of marketing and political correctness, not technology. The Japanese market is happy with ROM discs in caddies. The North American and European markets hate caddies. By making caddies optional for single-sided discs the DVD Consortium keeps everybody happy.

There is a completely different reason why caddies will be compulsory for double-sided discs, and it is purely logistic. There is no room on a DVD disc for a label at the centre. The only way to label a disc is on the reverse side—with a double-sided disc there is no blank reverse side so the only place to put a label is on a caddy. Hence caddies are obligatory for double sided discs.

In its basic form, the DVD-RAM recorder will work like a D-VHS cassette recorder, taping whatever bitstream is thrown at it, regardless of whether the bits convey audio or video, and of coding quality. The code can equally well come from a digital TV or radio receiver, digital camera, audio encoder or studio mixing desk. For the DVD recorder to work like a VCR it must encode analogue video in real time. Until now this has seemed a long way off, but both Sony and Hitachi have now announced MPEG2 single chip codecs. Multichannel audio codecs are sure to follow.

I report all this because commercial success of DVD-RAM in the computer and consumer markets will stimulate mass production of blanks which professionals can then use.

DAB hand in radio

With the dawn of practical digital radio broadcasting finally upon us, the inevitable political power plays have begun for the British broadcast market writes Kevin Hilton

DAB—REMEMBER THAT? Now there was something that caught the attention for a while, not that it should have done, of course. All those involved in the Eureka 147 Project tried very hard to play things down, saying that September 1995 wasn't a launch, it was just the start of pilot schemes.

That didn't stop broadcasters like the BBC having big get-togethers where executives spouted ill-informed soundbites about the new golden era of radio. Everyone outside of the groups immediately involved in DAB were bemused—and that included a couple of BBC presenters who interviewed me about what it all meant.

The conversation ran along these lines: 'So there's this brand new radio system that nobody can receive at the moment because there's no affordable equipment on the market?' Yes. 'So what're the benefits of DAB, then?' And so on, and so on. And I still haven't been paid for that interview...

Every so often the subject pops back onto the agenda, but the reaction is still pretty much the same: everyone is underwhelmed and just a little fearful that this is yet another format that means they'll have to fork out money for new equipment, with the sinking feeling that something else may come along in five years time.

Despite all the talking there has been the feeling that it was something that was still a way away and therefore might not happen. Like all things one tries to forget about, DAB has snuck up on us, because, later this year, digital audio broadcast *c'est arrivé!* Or whatever that phrase is in German, because it will be at the IFA consumer electronics fair in Berlin that manufacturers will unveil the first DAB receivers that won't require an overdraft and a long waiting list.

During the AES Convention in Munich, DAB was to the fore, largely because Bavaria has led the way in terms of Germany's pilot scheme for the new technology. Enthusiastically pushing the format were representatives from research facility IRT, which has been heavily involved with test transmissions throughout the region.

The conversation I had with one of IRT's principals was in dramatic contrast to an earlier conversation I had with a representative of a German technical consultancy that has been heavily involved with the implementation of digital technology for radio. According to this source, the initial enthusiasm for and interest in DAB had waned, particularly in the private radio sector.

The commercial stations are concerned whether they will be able to compete with the centrally-funded public broadcasters, which also have the advantage of being national services, something that makes more sense in the implementation of DAB multiplexes.

A way round this could be the joining together of commercial services to create a viable national 'network', something that has recently happened in the UK. Two of the country's biggest groups, Capital and Virgin, were merged when the former bought out the latter for a reported £87 million (a £65m price tag plus £22m of Virgin debts).

The deal depends on approval from the Radio Authority, but, if it is passed, then the former fierce rivals, which were fighting for supremacy in London particularly, intend to take on BBC Radio 1FM. Richard Branson, who founded Virgin Radio four years ago, explained that the merger would enable the company, which would be the largest commercial group in the UK, to challenge what he sees as the BBC's dominance in DAB.

Looking behind this independent bullishness, giving the impression of noble privateers taking on the establishment, there is financial and commercial expediency. With its London FM frequency Virgin had cut into Capital's prevalence in the city. If the RA allows the take-over, Virgin will become what industry cynics are already calling Capital 3 and 4, joining the existing FM and AM services.

> Behind this bullishness there is financial and commercial expediency

The plan is for Virgin FM to become a rock station aimed at the 24–40 age group (which is what it originally was before trying to take on R1 and Capital FM), while the national AM service will be retargeted towards the 15–34 age group, directly at the revitalised R1.

As the AM station has already been promised a place on an independent DAB multiplex, its amplitude modulation status will no longer be a barrier to taking on the BBC's nationwide service directly.

The only stumble could come from the RA deciding that it is not in the public interest for Capital to have three local licences in London, although the body has allowed the UK's one-time-independent local radio structure to become neither independent nor local, with a few big groups each controlling a large number of stations.

There should be a commercial alternative to the public service plans for DAB, but if the financial markets are anything to go by, the apathy towards the new technology, caused by a mind-numbingly long gestation period, appears to be spreading. On the night the Capital-Virgin deal was announced, Capital's shares fell by 5p. Hardly a welcoming sign.

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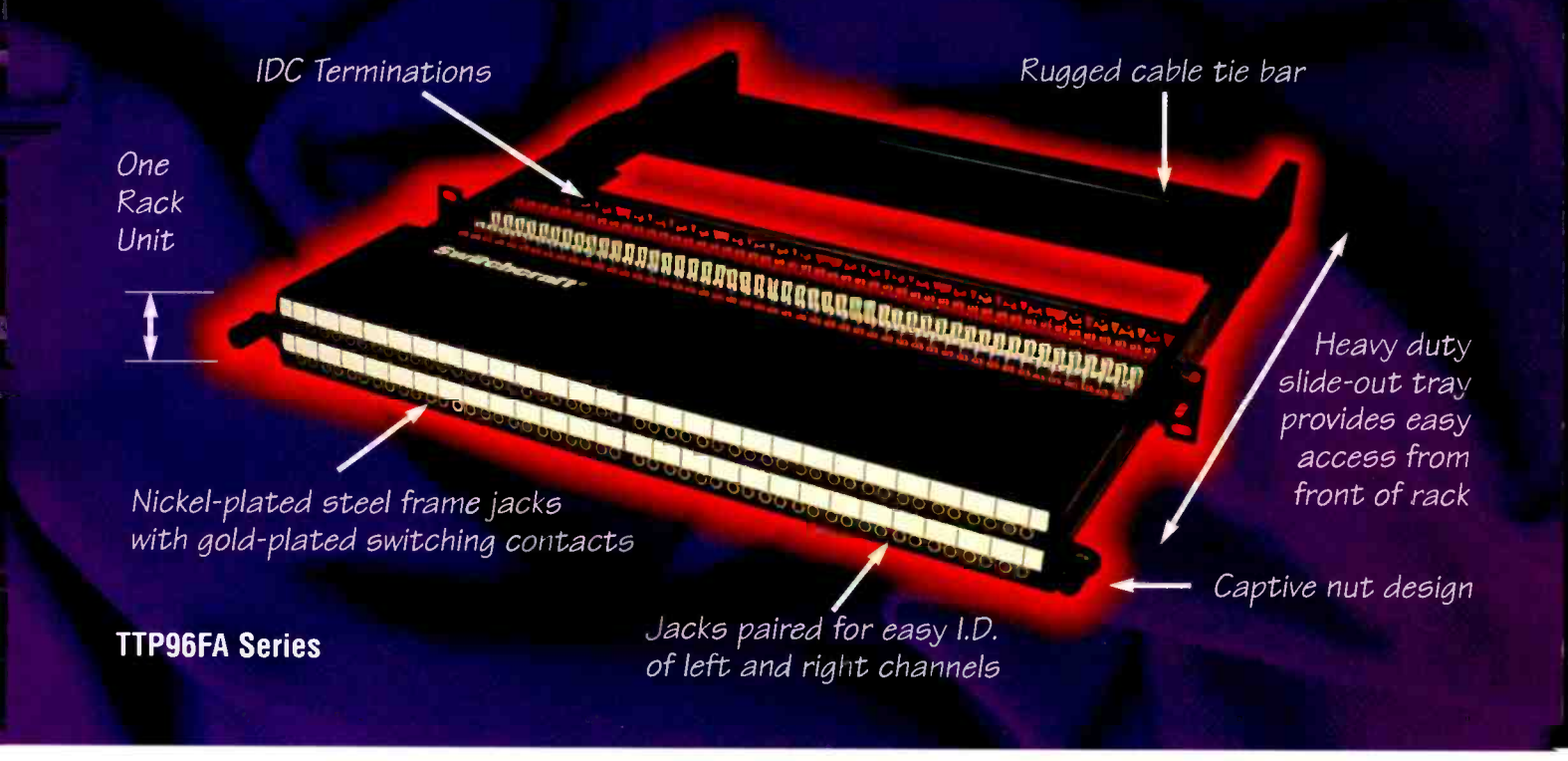

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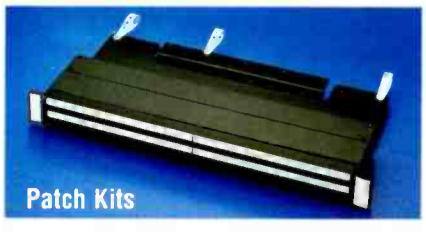

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AC power explained

With virtually every component in a studio requiring electrical power, understanding the principles is essential. **John Watkinson** gives a tutorial on a neglected subject

ELECTRICITY is an excellent servant but a dangerous master. Used wisely, it's an essential ingredient of music recording. Used wrongly it can kill, either directly or through fires. Even if you escape the fire, the loss could kill your business. A poor electrical installation can cause hums, crackles and buzzes on the audio, making your output quality lower than the studio next door that has fixed all its problems. Still not interested?

Electricity is a convenient way of transmitting power from one place to another. For our purposes, the power source can be considered to be the output shaft of a steam turbine. This shaft drives a generator which converts mechanical power to electrical power. The power is transmitted elsewhere by wires, and these suffer from two problems.

Fig.1a shows that a complete circuit is necessary so that current flows in a loop to the load and back. The first problem is that the wires must be insulated to prevent the current flowing from one to the other, forming a short circuit. The second problem is that some of the power is lost because of the resistance of the wire. The power loss is proportional to the current. **Fig.1b** shows a 1kiloWatt generator driving a motor with a nominal 100Volts. From Ohm's law this requires 10Amps. If the wiring resistance is 1Ohm, then 100Watts is lost in the wiring. **Fig.1c** shows a 1kiloWatt generator which has a 1kiloVolt output. Now the current is only 1Amp and the wiring loss is 1Watt. So going to a higher voltage has improved the efficiency from 90% to 99.9%, hence the use of incredibly high voltages on electricity pylons.

Unfortunately, using 132,000V to drive a cassette deck is not really on, and a lower voltage is needed for local distribution to consumers. In order to change voltage, a transformer is needed, and these only work on alternating current (AC). **Fig.2a** shows that a transformer is the electrical equivalent of a gearbox. The voltage ratio is the same as the turns ratio between the primary and secondary windings. In a practical power generation system shown in **Fig.2b** the AC generator, or alternator, works at a few kiloVolts to allow the use of moderate amounts of insulation. This is stepped up to a phenomenal voltage for transmission, and then dropped again in a substation for distribution at a safer voltage.

Fig.2c shows that the voltage waveform of an ideal AC power source is a sine wave. Clearly the voltage and the power fall to zero twice per cycle. **Fig.2d** shows that the instantaneous power is proportional to the square of the voltage, so the average power is given by the mean of the square of the voltage. The DC voltage which would produce the same power is given by the square root of the mean of the square of the AC voltage. This is the origin of Volts RMS (Root Mean Square). Ten Volts RMS delivers the same power into a resistor as 10V DC. The peak voltage is obtained by multiplying the RMS voltage by the square root of 2. Thus if you are dumb enough to stick your finger in a 230V RMS socket, you will actually find out what 325V peak feels like.

When the power involved is measured in megaWatts, as is the case with the alternator shaft, the power fluctuations of a single phase AC system would tear the machine apart. The solution is to use 3-phase AC. **Fig.3a** shows that 3-phase AC has three wires with 120° between the voltage waveforms. If the same peak current flows in each phase, the total power turns out to be constant even though the power fluctuates in each phase. **Fig.3b** shows a three phase transformer; the windings can be connected as a star or a delta configuration. The delta configuration is ideal for loads such as motors that must take exactly the same current from each phase. The star configuration produces an additional terminal at the star point, which is called the neutral point. Note that because of the 120° phase relationships the phase-to-phase voltage is somewhat more than the phase-to-neutral voltage. In the UK the phase to neutral voltage is typically 230V at 50Hz whereas in the USA it is 110V at 60Hz.

The star configuration is used for domestic distribution because each house can be supplied with a single phase and neutral. **Fig.3c** shows that the phase selected rotates from one house to the next so that the phase loading is more or less balanced. If the loading is not balanced there will be a net current flow in the neutral wire at the substation transformer.

Larger premises will be fed with all three phases and the wiring within the building must be arranged so that a reasonable load balance is obtained.

When the load is resistive, like an electric fire, the current is in phase with the voltage and the maximum power is delivered. However, many loads are reactive, mostly inductive, and in this case the current and the voltage will not be in phase. Take the case of a powerful class-B audio amplifier with no audio input. Very little current will > page 96

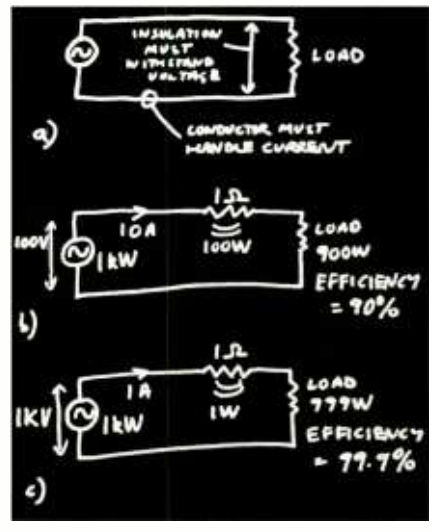

Fig.1a: Complete circuit is needed to pass current. Fig.1b: 100Volt working gives 90% efficiency. Fig.1c: 1,000Volt working gives 99.9% efficiency

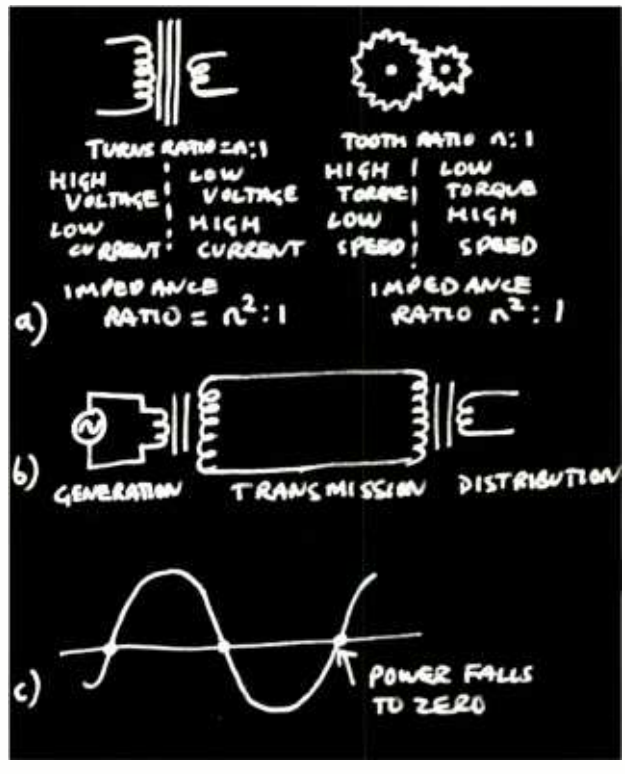

Fig.2a: The transformer is an electrical gearbox. Fig.2b: Practical power system steps up to high voltage for transmission. Fig.2c: Single phase power is not constant

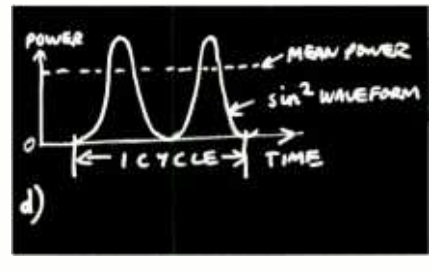

Fig.2d: Single phase power sin² waveform. Mean power is the same as that delivered by DC voltage equal to RMS voltage

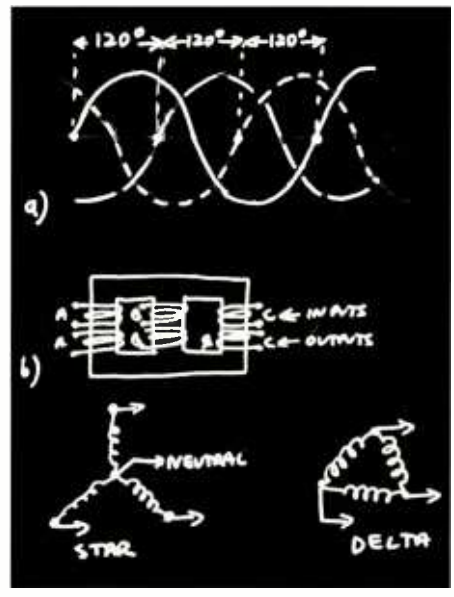

Fig.3a: Three-phase power has overlapping waveforms so that total power is constant. Fig.3b: Three-phase transformer has three limbs, can be connected in star, with neutral, or delta without

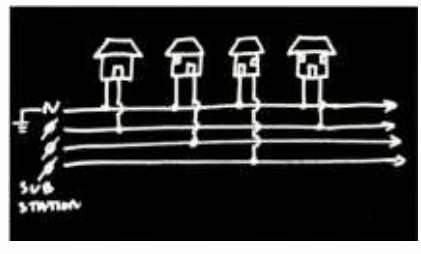

Fig.3c: Domestic single phase supply rotates between phases from one house to the next.

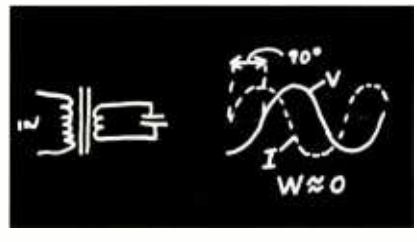

Fig.4: Although capacitor dissipates no power, full current still flows, causing transmission losses. Transformers are rated in VA not Watts to protect against this.

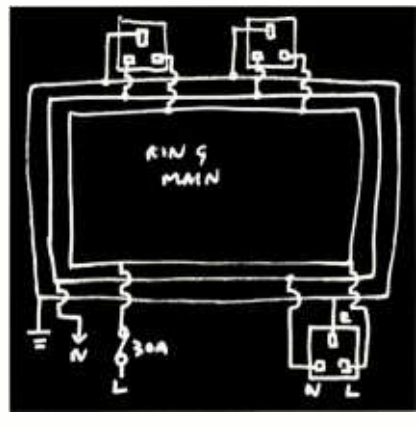

Fig.5: Ring main system used in the UK is only legal with 13A sockets because plugs must be fused to protect the load, not the ring. Ring has its own fuse

< page 95 be drawn from the power supply and the reservoir capacitor voltage will rise almost to the peak voltage of the transformer secondary. No secondary current flows for most of the cycle as the rectifier is reverse biased. The primary current is limited by the inductance of the winding and the phase of the current will be nearly at 90° to the phase of the voltage.

Although little power is being used, the Ohmic loss due to the current in the entire transmission system is undiminished and its heating effect is unchanged. This is why transformers still get warm even if the secondary is unloaded. **Fig.4** shows a transformer driving a capacitor. The secondary current is at nearly 90° phase to the voltage so the power delivered is almost nil. The term 'wattless power' is sometimes used to describe the situation. The ratio of the power delivered to the power which would be delivered into a resistor is called the power factor which is given by the cosine of the phase angle. As the Ohmic heating is due to the magnitude of the current, and independent of the phase angle, transformers are rated in VA (Volt Amps) rather than Watts. Clearly using a transformer at its maximum VA rating with a near zero power factor is pretty stupid, but is what happens in many practical cases, including our idling class-B amp.

Your household electricity meter does a vector multiplication of the current and the voltage. So you only pay for the actual kilo-Watt hours, irrespective of the phase angle. If you run at a miserable power factor, the electricity company has to supply all of the quadrature current and sustain all of the losses. In larger installations, this situation is wasteful, and if the load is predominantly inductive it will be necessary to install capacitor banks to improve the power factor.

As a fault in an electrical device can result in its resistance or inductance falling, the result is high dissipation and a temperature rise, leading to burnout. This is avoided by fuses which are the equivalent of the weak link in a chain. If the normal current is exceeded the fuse link melts and interrupts the current. The fuse must be in the live wire as an open neutral fuse would leave most of the circuit live, but with no power flow to indicate the condition.

The traditional way of installing single-phase power wiring was to run a separate set of wires back to a terminal box for each socket. The wires need to be able to carry the full-rated power of the socket, whether it is in use or not. A more efficient system is to use a ring main. **Fig.5** shows that within a given floor area, an unlimited number of 13A sockets can be connected with the live, neutral and earth wires in a ring. A fully loaded socket at the furthest point on the ring will be supplied by both halves of the ring in parallel, allowing thinner wire to be used. For every powerful device in typical use, such as a kettle or fan heater, there will be lots of low-powered devices such as radios, fans, power supply bricks, PCs and so on, so the ring main can operate with a 30A fuse, even though there are enough sockets to supply much more. A 30A fuse only protects the ring wiring, so it is essential that each plug has a fuse suitable for its connected appliance. ■

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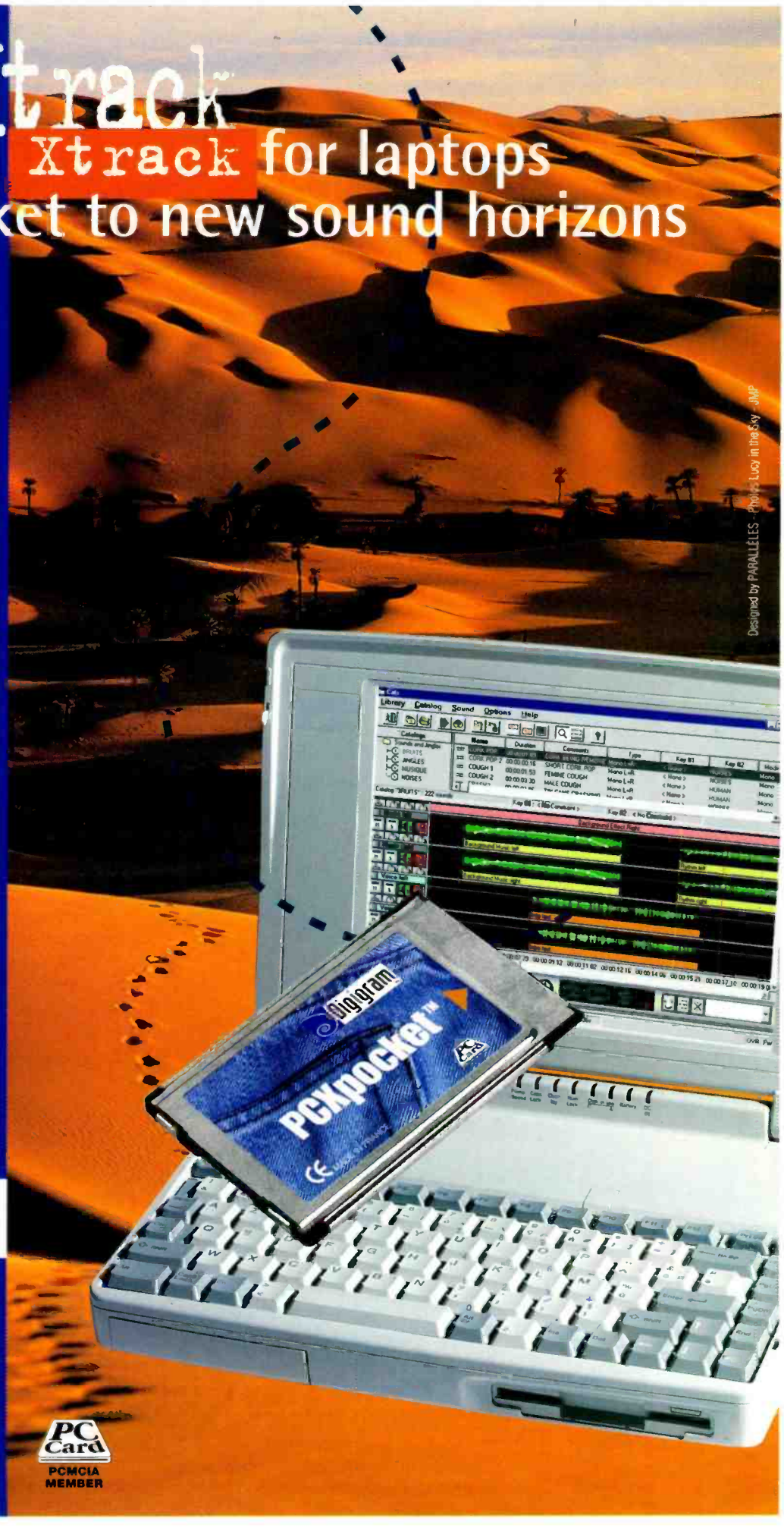

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Station to station

From buzzword to byword, DAW systems have steadily gained in reliability and desirability, and they now define postproduction. **Yasmin Hashmi** gazes into the future of workstation solutions

WITH THE PROMISE of broadcasting entering the digital era, the past year has seen much talk of convergence. This is understandable when manufacturers and service providers begin to give demonstrations of radio accompanied by pictures and text, or when telephone companies show how pictures, text and audio can be broadcast over the Internet. However, the term convergence seems to apply to media rather than applications. In practice, the gradual digitisation of virtually all forms of media has seen manufacturers responding by addressing increasingly diverse markets, ranging from cart replacement for radio or theatre, to postproduction editing and surround sound mixing for video, television, film and DVD (Digital Versatile Disc), to production for the Internet or mastering for multimedia CD.

There will inevitably be a certain amount of crossover between these markets in terms of technology, and this potentially offers new opportunities for multifunctional studios. However, whether this will lead to the digital audio workstation market converging into the 'seamless studio', based on a handful of standard platforms differentiated only by software applications and customised controllers, remains to be seen. According to Hazel Simpson, group commercial director for SSL, 'Digital audio markets have been migrating towards an integrated system model and away from the "stand-alone box" approach of the past.' Given the special needs of the professional audio industry against the more general goals of the computer industry, must such a model rely on proprietary solutions? Or can off-the-shelf platforms be adapted appropriately?

As personal computers become more powerful, a reasonable assumption would be that the days of the proprietary platform are numbered. Indeed this has been said ever since Digidesign, now well known for its Mac-based systems, took the audio industry by storm with its 2-channel Sound Tools card and software package. When the multichannel Pro Tools arrived, established players certainly felt a dent in their sales as many users understandably chose the cheaper alternative.

Nonetheless, while Digidesign has been one of the most successful digital audio workstation companies in terms of packages sold and the number of third-party software products developed for its cards, it could also be reasonably argued that with the arrival of built-in audio capabilities for computers, the days of such cards are also numbered. Silicon Graphics for example, recently demonstrated its professional digital audio board which is built into the Octane and Onyx2 platforms, as well as being available as an option for the O2 workstation. Supporting eight channels of 24-bit ADAT optical and two channels of AES-EBU, the technology is apparently already being beta tested by Steinberg. According to Kord Taylor, audio marketing manager for Silicon Graphics, 'Developers are recognising that the speed increase curve for CPU-based processing is faster than that of dedicated DSP chips.' This may be true, but not everyone is in the market for an SGI platform.

In fact, Macintosh computers have had built-in audio capabilities for some time, and there are a number of software packages such as the Macromedia Deck II which already take advantage of this. Deck II was one of the first software-only multitrack recording and mixing applications available, and the recently launched v2.6 now claims support for playback of up to 64 tracks in real time. It also features multiprocessor support for increased real time effects processing, and has addressed timing problems associated with the Apple Sound Manager. Indeed Digidesign itself responded to audio-capable Macs by launching the software-only Pro Tools PowerMix, and also began to address the even lower-cost IBM-compatible market with a PC version of Session 8.

Although the Mac is generally considered to have a more friendly operating system, the PC wins hands down in terms of cost and availability of peripherals, accessories and software. There are many PC-based audio editing software packages to choose from, which only require the addition of low-cost sound cards. Furthermore, as PCs begin to offer audio capabilities in the form of MMX technology for example, there will no doubt be a surge in the number of software-only products developed as a result, promising to make the digital audio workstation even cheaper and more accessible to the wider market.

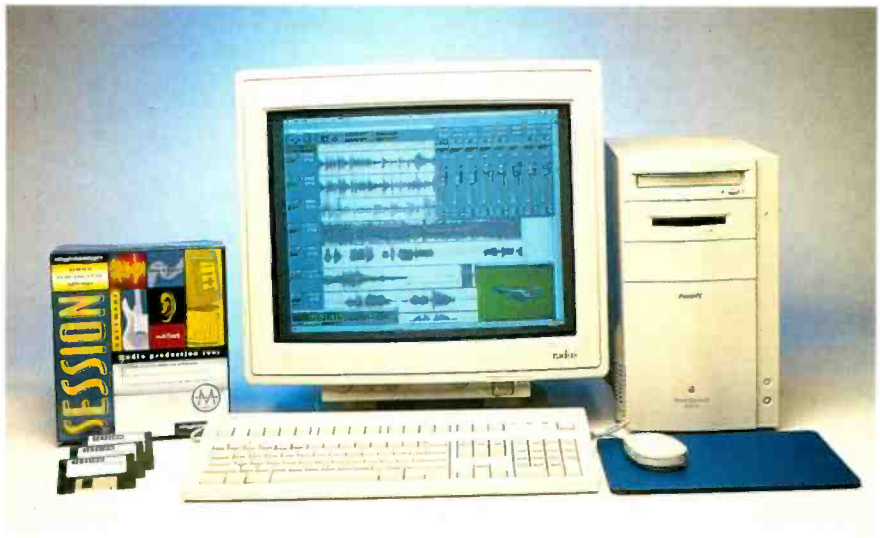
While on the surface, software-only solutions sound like the answer to many people's dreams, the reality is that off-the-shelf computers are still not generally mass produced with the audio professional in mind. If professional audio and sync I-O are required for example, then hardware with appropriate interfaces must generally be page 100 >

Fairlight MFX3 plus—talk of interchange

Akai has come a long way from the DD1000

Omnimix—moving away from 'stand-alone box'

DAR's offering of full and cut-down systems



Digidesign—still prolific

< page 99 added. Macromedia has always recommended Digidesign Audiomedia, Sound Tools or Pro Tools cards for audio I-O, although version 2.6 now supports the new Korg SoundLink DRS 1212 I-O multichannel audio interface PCI card. In fact the SoundLink DRS range of products also includes the dedicated 168RC recording console, and the 880 convertor units, which provide conversion between analogue and 8-channel ADAT optical.

Apart from the I-O issue, relying solely on the personal computer platform can have other drawbacks. For example, the overall performance of audio software can vary, depending on the host CPU, amount of RAM, and disk throughput. This means that there will still be a demand for proprietary engines geared specifically to handling and processing audio, such as the Spectral Prisma, the Sonic Solutions SonicStudio and the Windows 95/NT-based Merging Technologies Pyramix. These products are provided as plug-in cards accompanied by proprietary software applications, although Sonic Solutions recently introduced the SonicStudio Engine application programmers interface for third-party software developers such as Opcode.

However, not all engines consist of plug-in cards. The SoundScape Digital SSHDR-1 comprises modular rackmounts with all required processing, I-O, storage and sync interfaces. Provided that the specified drives are used, the new version 2.0 supports up to 12-track playback per unit, and the new SSAC-1 accelerator card, which retrofits existing units, allows each channel of the GUI-based mixer to run any number of real-time processes, while also providing an extra eight channels of TDIF I-O.

Some people are quite prepared to mix and match software and hardware products, and to source a platform and peripherals with the correct specifications, but for many, especially those seeking guaranteed performance, the turnkey proprietary system is a far more convenient and attractive solution particularly if supplied with an integrated control surface. Manufacturers obviously agree, since the number of such systems is on the increase, ranging from low-cost, compact self-contained desktop units such as the Roland VS-880, to high-end systems such as the new SSL Altimix, which includes integrated nonlinear video and surround sound mixing for the emerging new television formats.

Proprietary hardware is an expensive route for a manufacturer to take, particularly if the product is not intended for mass production, and with so many companies competing for the lower volume, higher cost end of the market, there are bound to be casualties. The most recent was the Fostex Foundation which was officially withdrawn from sale in March. The StudioFrame has changed hands yet again, having been sold off by Timeline which, after a restructuring process, is now concentrating on its MMR-8 modular recorder. Nonetheless for those established systems which remain, the market over the past year appears to have been fairly buoyant.

Having been purchased by Managing Director Mike Parker, Digital Audio Research (DAR) is keen to re-establish itself in the US, and recently showed its Sound- page 102 >

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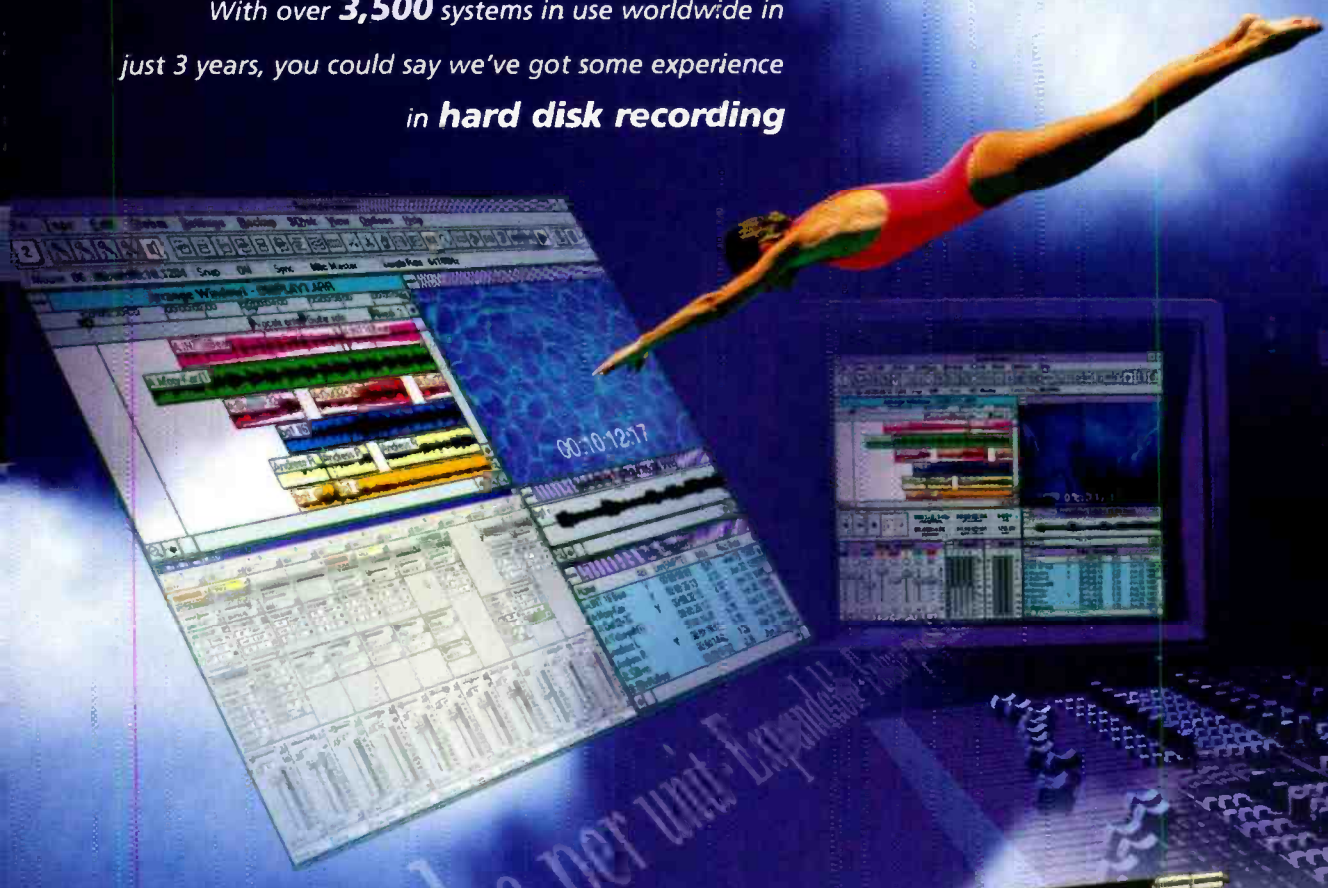

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Fairlight dubber—part of a trend

< page 100 Station Gold and Sabre Plus at the NAB Convention in Las Vegas. The company also showed a beta version of its OMR8 modular recorder, which is fully compatible with other DAR systems and supports recording to hard or MO disc.

In fact, an increasing number of manufacturers now offer recorders or dubbers, aimed at providing cost-effective record/replay facilities to complement their existing editing systems. Most of these are aimed at replacing mag machines or multitracks for film and video postproduction, a specific area of the professional market that until recently has not been well served. However, having announced the availability of the DaD dubber, a 24-track unit that can play back files created on Fairlight MFX3 series workstations, Fairlight USA chief executive John Lancken confidently expects to have 'well over 1,000 tracks of playback in Los Angeles by the end of June'. While the DaD promises to satisfy the demand by MFX3 users for a cost-effective playback system, there is also a need for more general purpose units which can record as well as playback.

The Akai solution is the DD8. Designed as a modular 8-track machine, the unit can use hard disk or removable media and supports a range of I-O options including TDIF and ADAT. It can be controlled via its front panel, RS422 or the DL1500 edit controller, and is compatible with other Akai systems such as the DD1500 editor and DR8 and DR16 recorders. Furthermore, according to Akai, the DD8 can sync to virtually any timing reference, including biphase, in forward or reverse, at any speed—a feature that will no doubt be particularly welcomed by film mixers. The Genex 8000 8-track recorder also supports this feature.

Following on from the DMT-8 low-cost compact workstation, Fostex launched the D-90 8-track and D-160 16-track series of low-cost recorders. Using removable hard drives, both products include SPDIF and ADAT interfaces, selectable 44.1kHz or 48kHz sampling, and varipitch support. For the larger multitrack replacement market, Otari has launched a 48-track version of page 104 >

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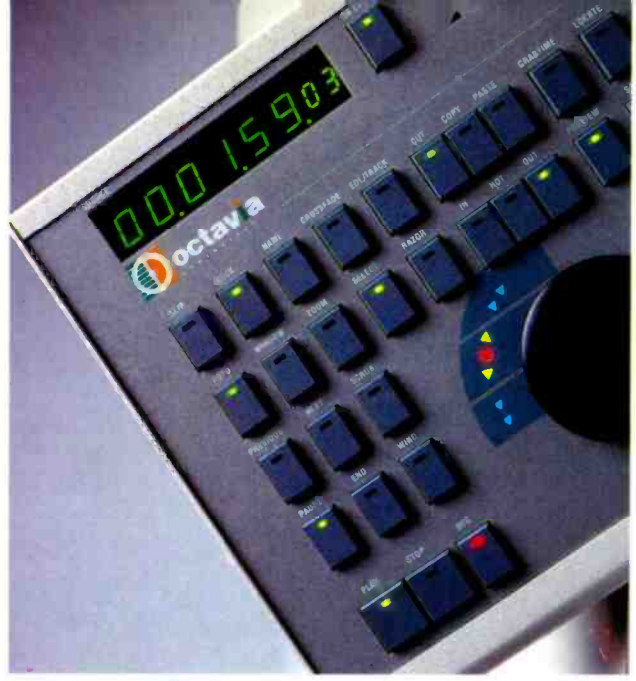

SA&V—supporting BWF

< page 102 RADAR, which is now able to display in feet and frames as well as in bars and beats and SMPTE time code. For those who need fewer tracks, but more storage, Studer has released v1.1 of its D424 4-channel recorder, offering more than two hours of recording, as well as a choice of 16-bit, 20-bit and 24-bit resolution, PQ editing and transfer to CD-R via SCSI.

Another marked development that can be seen as a logical response to the threat of cheaper computer-based systems, is the availability of cut down versions of proprietary systems. By using removable media and-or networking, these have the clear advantage of being compatible with their larger counterparts, as well as offering familiarity in terms of operational control. Such systems include the DAR Sabre, the Fairlight MFX3 Mini and the recently launched AMS Neve AudioFile Prolog. Designed as a feeder system for larger 24-bit AudioFile-Logic mixing systems, the 16-bit capable Prolog supports 16 channels with six hours of storage, includes a comprehensive ADR package, and can be upgraded in a variety of ways.

While manufacturers attempt to provide total solutions by offering a range of compatible products, in practice many users will still be faced with the need to deal with material from different makes of audio system. Furthermore, as video editing becomes increasingly dominated by tapeless technology, there is also a growing need for direct transfer between nonlinear video editing systems and audio workstations.

The OMF initiative promised to address the need for file transfer between both audio and video systems, and packages such as the AirWorks S/Link and Spectral MediaFiler already provide format conversion for audio files. However, the general lack of compatibility between systems is still one of the major obstacles in the way of the seamless studio. Furthermore, the emerging digitisation of broadcasting for both radio and television is presenting manufacturers and users with new challenges, not least in the acquisition, preparation and archiving of current material in a suitable and compatible form, for future use in digital transmissions. Users have been aware of this for some time, but have been frustrated by the industry's inability to agree on standards.

Various companies have been cooperating on a one-to-one basis in order to provide at least some form of transfer capability between their respective systems and those which they see as natural partners. Orban for example, which recently launched it new generation Audicy workstation, already provides direct file transfer capabilities between its DSE-7000 radio production system and the Enco DAD and Broadcast Electronics AudioVAULT on-air delivery systems. AMS Neve, DAR and SSL support audio files created by the Lightworks video editor, and Fairlight recently announced an initiative for direct file interchange with a range of companies including Studio Audio & Video (SA&V), Doremi, Timeline and Lightworks. However, this is seen as an interim step, in the hope that the industry will eventually agree on a standard format such as WAVE or the new BWF, a version of WAVE designed for broadcast applications.

Already supported by Merging Technologies and SA&V, BWF is apparently fairly straightforward to implement, and contains the minimum information considered necessary for all broadcast applications. It accommodates both linear and MPEG audio, as well as higher level descriptors which can be used to refer to other files containing more complex sets of information.

While the computer industry as a whole cannot be expected to address the specific needs of the digital audio workstation market, it is clearly driving its overall direction. Hardware will become ever more powerful, costs will come down, and software will increasingly support cross-platform operation. However, just as advances in other industries have led to a certain amount of homogenisation in terms of design, there will always be a case for individuality and proprietary solutions.

Indeed, there is a striking similarity between the reasons for purchasing a particular type of digital audio workstation, and those for purchasing a motor car. Some people will be satisfied with limited functionality, so long as the system is within their budget. Some are more concerned with how it handles, others rely on a proven track record and support from the supplier, and yet others make their choice for reasons of prestige. As the owner of a high-end post-production facility recently pointed out, 'My clients are paying a lot of money and expect to see a lot of impressive kit—I couldn't get away with just having a computer screen on the desk'. Whether or not a peek beneath the bonnet will, in future, reveal a standard engine, remains to be seen. ■

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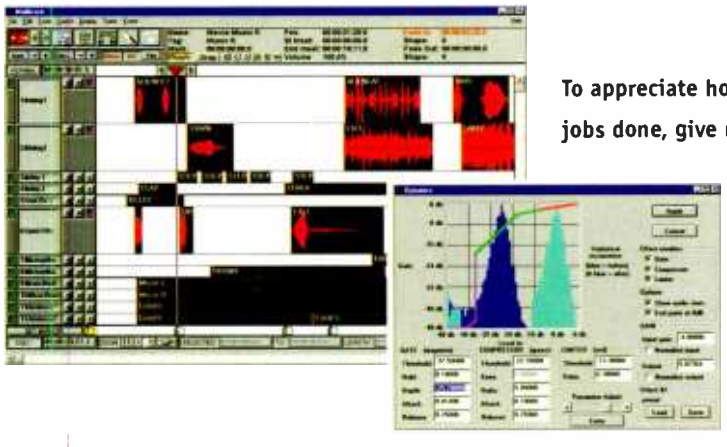

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True lies

Assessing digital conversion systems in terms of numbers of bits is at best misleading and potentially disastrous. Prism Sound's **Graham Boswell** asks a few key questions and offers a few answers

One of the most perfidious habits in our industry is the propensity of product marketeers to take liberties with engineering specifications. There are few better examples than the number of bits attributed to analogue-to-digital convertors. The problem arises because manufacturers of both chips and equipment have taken liberties with the 'n-bit' measure, and because conversion systems don't exactly fit the concept of 'n-bits'.

Let's drop the 'n-bits' claim and replace it with measures such as dynamic range, amplitude linearity and distortion (including non-harmonic products). I place them in this order because the nearest useful number we have approximately corresponding to 'n-bits' is dynamic range while amplitude linearity principally concerns low-level performance. An important feature of convertors is their ability to behave like analogue systems and allow audio to fade into the residual noise.

In analogue recording ¼-inch, 30ips is widely regarded as providing reference-quality recording. Analogue tape is often favoured because of its 'sound' even though tape compression can introduce quite large amounts of distortion. How does digital compare?

Well, it's band-limited (at CD sampling-rate); has no compression effect, low distortion, and provides wide dynamic range. It's worth pointing out that the limiting factor in digital is the A–D, with comparable-quality D–As needed for listening, whereas with analogue systems the tape recording system is the key.

Early digital equipment had subtle flaws but developments in the last ten years or so have improved matters. Typical distortion figures for A–D and D–As usually fall well short of the so-called 'theoretical' figure. How many high-bit convertors manage distortion (THD&n) of better than -105dB (about 18 theoretical bits) for large signals? Few, if any. Unlike the onset of noise, these products are signal-related and become significant when the highest amplitudes are recorded, rather.

In professional applications we need to have dynamic range (as AES17) of more than 90dB to provide working headroom. Equipment in popular usage offers between -90dB and -110dB THD&n, while high-end equipment typically offers up to around -120dB.

If we record 20 or 24 bits, we have a recording format that imposes no significant noise as a result of wordlength limitation; noise is usually dominated by the A–D process itself. Do manufacturers claim 24-bit performance or 24-bit output wordlength? Does anybody offer a 141dB dynamic range?

It may sound perverse, but a 16-bit convertor can resolve signals less than its LSB if properly designed. An A–D can retain information of less significance than its LSB provided that the right sort and amount of dither (noise) is present prior to the quantisation. This helps digital systems emulate analogue by fading cleanly into noise. It can be shown that tones are converted successfully down to around -140dBFS in better quality A–D and D–A convertors, albeit in the presence of noise.

Reducing to 16-bit wordlength for CD could be a problem but the noise penalty can be reduced using noise-shaping—redistributing the quantisation noise spectrum without affecting the audio spectrum. This allows noise energy to be taken from the 3–6kHz band, where the average ear is most sensitive and added to regions above 15kHz. The result is that total noise amplitude is increased (above 15kHz) but sounds quieter to the ear.

But how should we assess wordlength with noise-shaping? Low-level programme might be exposed by apparently reduced noise, while our measurement equipment shows poorer results.

There is a case for recording long wordlengths as storage is rapidly getting cheaper and throwing bits away could compromise things—depending on the convertor and the number of bits you decide to throw away and whether or not you use dither.

Meanwhile, 96kHz sampling digital systems have been available for some years and are available now in the form of both comparatively inexpensive DAT recorders and high-end equipment. Rumours abound as to the future of audio on DVD. A 96kHz sampled multichannel, 16-bit linear PCM variant has been mentioned for DVD. With extra bandwidth, wordlength becomes less important because we can distribute the quantisation noise energy over a wider bandwidth—as a result noise-shaper peak levels can be much lower even with 'subjective' noise levels equivalent to around -108 to -110dBFS.

A key issue is measurement method: all measurements should be quoted using the recommendations of AES17 and any weightings should be identified. A recently published A–D chip spec includes a claim of 120dB dynamic range without making it clear that this could only be obtained using A-weighting. Another favourite trick is to quote D–A noise floors with the digital input at zero—as soon as any audio modulation is present, the residual noise rockets.

Would it not be more useful if that chip maker quoted dynamic range (weightings specified), amplitude linearity and distortion? Of course, noise-shaping clouds the issue, but dynamic range is about the only useful inference from the 'number of bits' and the mathematical integrity of current claims seems doubtful. More than dynamic range, clean reproduction of low amplitude material above and below the (unweighted and unshaped) noise-floor seems to have been the big issue over the last few years and great strides have been made improving it and reducing noise levels.

Answering the question, 'What's a true high-bit convertor' inevitably becomes a dialogue. Those who recall the line, 'What is the unladen weight of a swallow?' will remember the consequences for the bridge-keeper... ■

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Trim Control. –20dB to 40dB line gain, 0dB (Unity) to 60dB mic gain. (Mono chs.)

Polarity switch.

Aux sends 1-4 level controls & pre/post switch. Separate level controls for each Aux. Pre/post switch for Auxes 1-4 selects pre-fader/pre-EQ or post-fader/post-EQ operation.

Aux sends 5-8 level controls & pre/post switch. Again, separate level controls for each of Auxes 5-8. Different, though, is the pre/post switch. On Auxes 5-8, you can select pre-fader/post-EQ or post-fader/post-EQ.

12kHz Hi Shelving EQ. ±15dB of boost/cut.

Hi Mid EQ. Sweepable from 500Hz to 15kHz, with a 1.5 octave bandwidth. Separate controls for ±15dB boost/cut and frequency. (Fixed at 3kHz on stereo aux return channels)

Lo Mid EQ. Sweepable from 45Hz to 3kHz, with generous 1.5 octave bandwidth. Separate controls for ±15dB boost/cut and frequency center. (Fixed at 800Hz on stereo aux return chs.)

80Hz Lo Shelving EQ. ±15dB of boost/cut.

EQ In switch.

Low Cut Filter. Sweepable from 30Hz to 800Hz, 12dB/octave. (Fixed at 150Hz on stereo aux return channels.)

Constant-loudness Pan control. For consistent signal level across the stereo panorama, whether it's cranked hard left, right, or center.

Mute button and LED. Mutes channel and all aux sends. LED lights when channel is in mute, as well as when it's being muted or previewed by Ultra Mute™.

Submix 1-8 assign switches. Assign a channel to any or all of the 1-2, 3-4, 5-6 and 7-8 submixes.

L-R & Center assign switches. Ditto for L-R and Center mixes.

100mm log-taper channel fader. New logarithmic faders provide consistent fades throughout their travel, and feature a dust-shielded super smooth design.

Solo button and LED.

Never before has so much been so affordable. Using the automated manufacturing processes and design expertise that established our 8•Bus and SR Series, we've created a no-compromise, 40x8x3 large-format live sound console that's equally suited for sound recording.

- **4 extra stereo line channel strips** for aux returns, which include: 4-band equalization with 12kHz Hi shelving EQ, 3.5kHz Hi Mid EQ, 800Hz Lo Mid EQ and 80Hz Lo shelving EQ, plus 150Hz fixed low cut (high pass) filter at 18dB/octave, low cut in/out switch, other features same as mono channels.
- **8 submix section bus strips** feature 100mm log-taper faders, "Air" EQ controls, center & L-R assign switches, pan controls, mute & solo switches with LEDs.
- **Center-mounted master section** includes fader link L/R switch, center master fader, center solo + LED, L&R/L-R master level faders. L&R solo switches + LED indicators.
- **Independent L/R/Center bus assignment.**
- **On-board snapshot group muting** with external computer control capabilities.
- **12x4 matrix mixer** w/Center, L & R input controls, matrix mute button + LED, matrix master level control, and matrix solo + LED.
- **40 (or 56) mono channel strips** with Mackie's renowned high-headroom/low-noise discrete mic preamps, polarity switches & sweepable 30–800Hz low cut (high pass) filters at 12dB/octave.
- **Flip switch** for stage monitor applications exchanges the master control of any selected aux send with the corresponding subgroup fader and routes the aux send to the sub insert jacks, slide master fader, "Air" EQ, and balanced output connectors.
- **UltraMute section** has 99 mute groups and 99 snapshots capability, mute select switch + LED indicators, Store/Preview/Snapshot/Clear/Do It switches + LEDs, 10-unit snapshot indicator, group indicator, Mode/Up/Down switches.
- **Talkback section** includes talkback mic input (phantom powered), assign switches for Matrices A-B-C-D/Aux 1-4/Aux 5-8/Ext./L-R, talkback level control, solo level control, program level control, talk button, recessed phones jack (there's one on the rear panel, too), and 400 Hz/pink noise source with separate level control (uses talkback routing switches).
- **All inputs & outputs fully balanced** (except RCA tape inputs & outputs).
- **Each channel** has rear panel XLR mic & TRS line in jacks, insert with separate send & return (balanced), & direct out.
- **Built-in Clear-Com™ compatible interface:** ties the SR40•8 into any Clear-Com party-line intercom system, with ground isolation, ignore switch, call button + LED, and a trick or two that even Clear-Com doesn't have.
- **Rear panel** includes RS232 data port & MIDI In and Out jacks, as well as an exhaustive list of balanced inputs, outputs, throughputs and shotputs.
- **External 400-watt power supply** with redundant power capability.
- **4-pin XLR lamp sockets and dimmer.**

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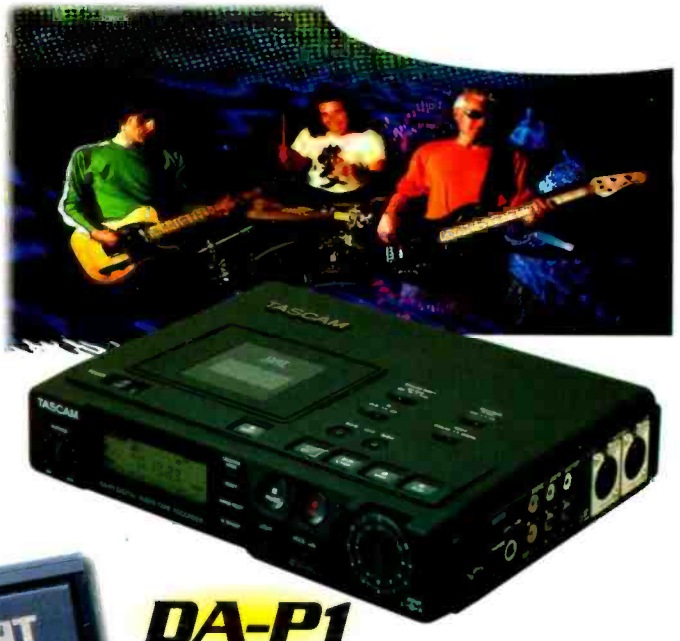

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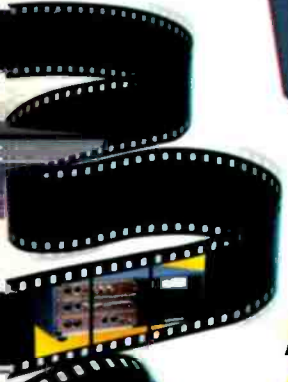

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