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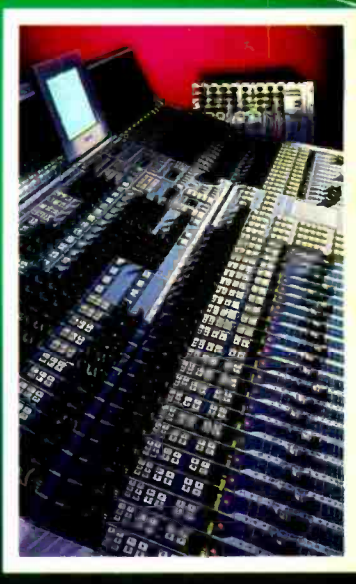
Studio sound

THE INTERNATIONAL TECHNICAL MAGAZINE FOR PRO AUDIO, POSTPRODUCTION & BROADCAST



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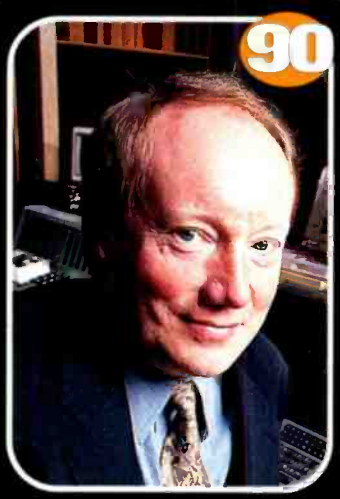
MAY 1996

Surveying the set of The White Room—the clinically clean front of a sophisticated television production and cool music show. Page 110



CORNALL INTERVIEW

One of the most influential and respected classical producers: Andrew Cornall in an exclusive interview with **Studio Sound**

REVIEW SECTION

PRODUCT OF THE MONTH STUDIO SOUND

COPENHAGEN AES

START PAGE 22

For some record companies, only one console is good enough.

Nippon Columbia

Chief Recording Engineer Mr Kazuhiro Tokieda (seated), with Recording Engineers Mr Katsuhiro Miura and Mr Takahi Sasaki.

"Studio 1 is used mainly for acoustic recording. It was important that the new console met Nippon Columbia's sound quality requirements, and provided enhanced operational use. The SL 9000 easily did so."
Mr Tanaka, Executive Recording Engineer, Nippon Columbia.

"I truly believe that our choice of the SL 9000 will make a significant contribution to the quality of our music releases."
Mr Iida, Recording Department Manager, Nippon Columbia.

SL 9000

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THIS EDITION'S CONTRIBUTORS

PATRICK STAPLEY began in pro audio in 1972 at London's Abbey Road Studios and has worked with artists as diverse as Paul McCartney, The Damned, Sting and Kraftwerk. Recently shortlisted for a Press Association award, Patrick has been writing for *Studio Sound* for over a decade.

ZENON SCHOEPE is a freelance journalist and regular contributor to *Studio Sound*. As an audio scribe he has a mighty reputation as a slayer of sacred cows and a spotter of the King's New Clothes. He's been a project musician and recordist for much longer than his 15 years writing.

JIM BETTERIDGE has spent his 18 years in audio as a professional musician, training at Advision Studios and as Odyssey Studios' Operations Manager. He also helped set up the first City and Guilds recording course, and now runs his own Copper Blue postpro studio in London.

Critical mass

While some of us actively seek recognition for our work, others are commendably modest. Such is the power of other peoples' attention that the word itself can inspire greatness or terror almost without qualification. Yet without attention of some kind our work–and often our personalities –can go unrecognised and unacknowledged. You did know you were in the ego business, didn't you?

One of the problems of making your work the focus of a review, however, is the reviewer. Too often the reviewer assumes the importance of the artist. Try this: 'The slow introduction is dominated by the four-note Death motif on the timpani, with protesting cries from the brass, marking the essence of the personal struggle which Suk fought at this time in order to overcome the overwhelming grief which beset him. At the time he wrote of the anguish he went through to achieve the ultimate goal of a peacefully confident C major and by then referred to *Asrael* not so much as a work of pain but one of "superhuman energy".'

Sometimes the whole process of review assumes the proportions of a competition: 'But Cowper's self-emancipation from the ornate words of ancient use and wont and the more elevated themes supposed to be essential to poetry was more complete even than Wordsworth's. He had no notion that his system was a new one, nor purpose of establishing a changed rule in the canons of poetry. Indeed his own poetic successes and fame have an accidental character altogether, as things which were never calculated upon in his own conception of his life, but stumbled into unawares in his endeavours to escape from the enemies of his peace.'

It seems that dealing in words or melodies gives you a better shot at popular recognition; perhaps it's because 'ordinary people' write letters and whistle tunes. Design a revolutionary console automation system and your audience is reduced to those who use such things–although the profile of the Fairlight CMI and SSL console do seem to have broken the general public's indifference barrier at times. Engineer a particularly difficult session, however, and even your peers are likely to be largely unaware of your achievements.

A CRITIC'S WORDS can be as important as the subject of the review. And the reviewer must have an appropriate sense of perspective if the review is to serve its subject and its readers properly.

The issue is one of recognition. Where, for example, do equipment manufacturers, music producers and engineers look for their reviews? Who, for example, gave a nod to the neat way that the theme music to the BBC production of *Spider-man* faded up as a link at the close of one scene and was hard panned and EQ'd to become part of a radio announcement at the start of another?

It is gratifying that *Studio Sound* is able to play a part in recognising the work of those in the pro-audio industry–through its reviews and features. And it is rewarding for the magazine to be recognised, in turn, for its success in terms of the publishing business.

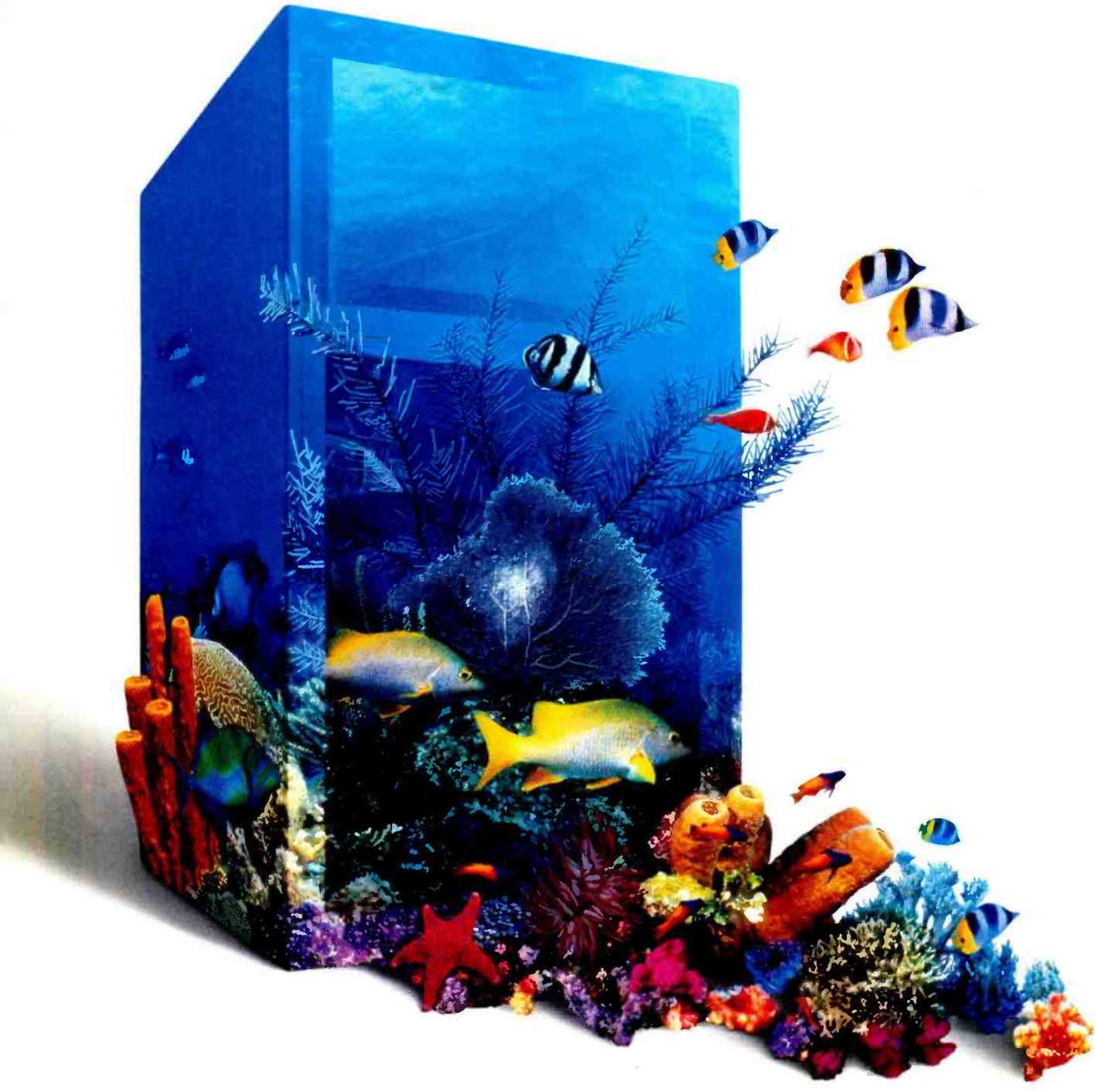
The advent of the 100th AES convention provides a timely opportunity to recognise the role that the Society and its annual conventions play in pulling pro-audio people together and offering them opportunities for communication and education that would be difficult to secure elsewhere. Congratulations, AES–for identifying and serving your audience with enthusiasm but without pretension.

Congratulations too to *Studio Sound* authors Patrick Stapley and Dave Foister. Patrick's world exclusive in breaking the story of the Beatles' *Anthology* series of recordings earned himself and *Studio Sound* a nomination in this year's Press Association awards and secured the Scoop of the Year award in Miller Freeman's editorial awards. Dave, meanwhile, had no trouble in being dubbed Technical Writer of the Year.

These reviewers did not secure their awards by being preoccupied with their own position or importance; they were recognised for the accuracy and accessibility of their work.

My personal thanks and congratulations to both.

editor



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Parp parp!

THE SOUND you just read is *Studio Sound* blowing its own trumpet as we once again confirm our status as the industry's leading magazine. In a recent awards ceremony held in London's historical Skinners' Hall, *Studio Sound* collected two of the most coveted editorial accolades in the form of 'Scoop of the Year' and 'Technical Writer of the Year'. These awards went to Patrick Stapley and Dave Foister respectively. The annual awards, which acknowledge journalistic excellence within the genuinely international Miller Freeman publishing group, are regarded as a measure of editorial achievement and are hotly contested by over 200 eligible magazines.

Patrick Stapley's world exclusive reporting of the Beatles reunion story received widespread acknowledgement in the international press last year and saw *Studio Sound* commended in the recent UK Press Association Awards. NICK SMITH

NEW YORK: Producer Mutt Lange recently approached Harris, Grant Associates for an acoustic and technical design package for a world class—but private—facility in upstate New York. Discrete Systems was engaged to provide a custom solution to the acoustic control required by HGA and went on to build the technical installation with Discrete Systems and Coastal Acoustics supplying the technical consultancy and a Boxer T4 monitor system.
The Control Room is equipped with one of the first 80-input SSL 9000j consoles (built onto a curved frame designed by HGA), over 400 rack units of outboard equipment, 96 tracks of digital recording, 24 tracks of analogue. It is complemented by six small varied recording rooms and a main live area featuring large-scale variable acoustics remotely controlled from the console.
The environment is fundamental to the quality of the complex but precludes any recourse to rapid external maintenance or rental companies and so the studio has been conceived to accommodate all requests that may be made of its capabilities whilst remaining a technically accessible and comfortable environment for creativity.

AMERICA'S NAB SHOW may take much of its lead from video, but the stand made this year by the audio sector went some way to casting it as more than video's 'poor relation'. Apart from the Radio-Audio Hall—which housed around 250 audio exhibitors—there was a healthy audio presence throughout the main halls and the newly established Multimedia Hall. Healthy, that is, unless you happen to be one of the 85,000 or so visitors doing the phenomenal amount of legwork necessary to visit them all.

The effort was well rewarded, however, not only in terms of new and developing equipment but in terms of the video industry's increasing awareness of audio issues. Nowhere better was this illustrated than through NVision's *The NVision Book: An Engineer's Guide to the Digital Transition*. Building on the success of its predecessor, the John-Watkinson-penned *The Video Engineer's Guide to Digital Audio*, The *Book* is a reference book published by a video-orientated company to help video people deal with the 'difficult subject' of audio. TimeLine also contributed a book to the cause of education through an update to their *SMPTE Made Simple, a Time-Code Tutor*.

Other notable NAB paperwork came in the form of Otari's PicMix white paper, A Surround Sound Primer, which proposes the PicMix as a standard surround-sound upgrade for consoles lacking the routeing facilities essential in for surround mixing.

Notable equipment innovations embraced high-end developments such as Otari's Elite recording console; Sonic Solutions' SonicStudio Post 24-bit, 96kHz system; Nagra's 24-bit, 96kHz capability for its Nagra-D digital recorder; Avid's Audio Vision IV preview; and Digidesign's ProControl and Pro Tools v4 software—and extended down to Whirlwind's modest but invaluable Brick mic-line driver. Even the unfashionable—in video terms—issue of loudspeakers received worthwhile attention in the form of KRK's new Rok Bottom subwoofer and the excellent Trapagon Monitor Systems showing.

Interestingly, certain of the audio exhibitors had chosen stands outside the Radio-Audio Hall in an effort to increase their catchment beyond those visitors interested in dedicated audio systems. Specifically, SSL took the audio specialists' approach and reported fewer, but high-quality, visitors while AMS Neve was pleased to be able to woo visitors from Sony's adjacent OXF R3 console display onto their own display of Logic. Sonic Solutions, meanwhile, had taken up residence in the

Las Vegas: audio means big business to the gaudy city

Multimedia Hall (set away from the main Convention centre at the Sands and housing around 600 exhibitors) and showed no evidence of regret or reduced interest—but curiously found itself keeping the company of a handful of camera and lighting manufacturers—it seems that the NAB's organisation had fallen just short of the last hurdle. But if you needed convincing of the erosion of disciplinary boundaries, NAB was a good place to get it.

More predictable aspects of the audio side of the exhibition included a slough of new consoles (including those from D&R, Harrison and Otari), the announcement of a forthcoming audio-video production system called Trinity and involving Play, Graham Patten, Soft Image and Merging Technologies,and updates to many high-end recording, mixing and processing systems (AMS Neve, SSL, Avid-Digidesign...) and an abundance of broadcast (Orban's Optimod FM2200, 360 Systems' Shortcut) launches. In spite of many reservations, the Multimedia Hall —stuck in another hotel complex apart from the main exhibition —attracted traffic and provided the busiest audio stand many of us encountered through the Avid-Digidesign obstacle course.

In short, the NAB Convention had something for just about everybody.

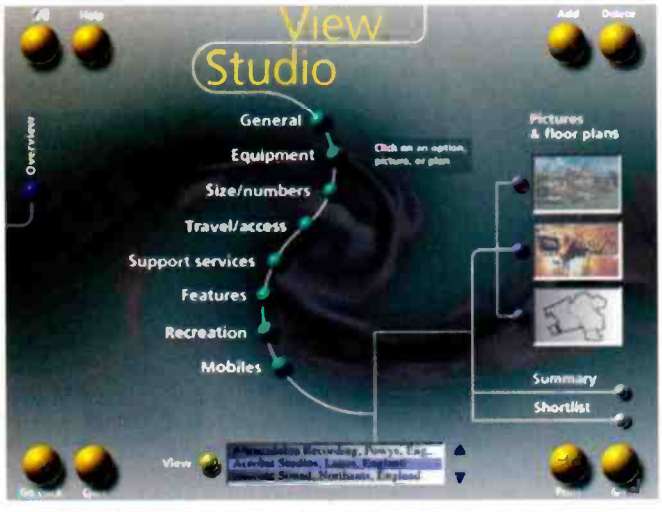
NAB boasts that it is the second largest show (to the US Comdex) hosted by the vast Las Vegas exhibition. Certainly, it represents big business to the gaudy city.

TIM GOODYER

THE NET: more audio manufacturers, distributors, studios and associations are getting their acts in gear and doing the decent thing by getting themselves webbed-up.

Everything you ever wanted to know about Spirit is now available from the comfort of your own computer—from product specifications to baseball caps, it's all in there at: http://www.spirit-by-soundcraft.co.uk.

Digidesign has announced its new site offering 24-hour customer support. In the coming weeks the site is going to be expanded to include product information, schedules of seminars, demonstrations and other news. The site will also provide links to Avid ☞



COPENHAGEN: The 100th AES Show will see the launch of the 1996 Studio Encyclomedia, the music industry's first interactive directory of the world's top 500 professional recording studios. Distributed in CD-ROM format, browsers will be able to access information by any number of parameters, allowing decisions to be made based on specific requirements, such as type of mixing console, location of facility or even car parking. Studios interested in further information about the directory should contact Encyclomedia on +44 181 455 1008.

News in brief

◆ Parisian post facility Point 12 is to incorporate a new 16-fader AMS Neve Logic 3 console in its relocation. The new facility will also house Point 12's four AudioFiles and continue its work in TV commercials.
AMS Neve, UK.
Tel: +44 1282 457011.

◆ San Diego recently witnessed the installation of six TimeLine Studioframe DAW-80 workstations at Sondelux—the world's largest independent postpro facility.
As a sister to Signet Sound in LA, Sondelux' group uses over 30 DAW-80s.
TimeLine Vista, US.
Tel: +1 619 727 3300.

◆ China Central Television has installed a 56-input Euphonix CS2000 console, bringing the CS2000's presence in the PRC to six. Elsewhere in the Far East, Taipei's Creator A-V Studios has installed a 72-input CS2000 in place of its previous SSL G-series console. Creator will be using the new console primarily for music tracking.
Euphonix, US.
Tel:+1 415 855 0400.
Creator Studios, Taiwan.
Tel: 8862 218 9768.

◆ German state radio station, SDR, has bought a Fairlight MFX3 for use in its new radio drama production studio. SDR joins Frankfurt's HR radio in its support of Fairlight through its purchase of an MFX3 Mini system. London's Town House studio claimed the first UK order of Fairlight's FAME mixer-editor for use in a new audio-for-picture post room which will also handle CD mastering.
Fairlight, Australia.
Tel: +61 2 975 1230.
Fairlight, UK.
Tel: +44 171 267 3323.

◆ American new media company, N2K (Need to Know), has chosen Rane's THX44 EQ and SSA6 amps for its 'multimedia boardroom' at its New York HQ. N2K provides a number of on-line sites including: Jazz Central (www.jazzcentralstation.com) MusicBlvd (www.musicblvd.com).
Rane, US. Tel: +1 206 355 6000.

◆ London's Metropolis Studios has recently installed a 56-channel Mackie 8-bus console, Digidesign Pro Tools system, eight channels of Sample Cell and Emagic Logic Audio sequencing in two new production rooms.
Mackie, US. Tel: +1 206 487 4333.
Digidesign, US. Tel: +1 415 688 0600.
Emagic, Germany.
Tel: +49 4101 4765 44.
Metropolis, UK.
Tel: +44 181 742 3111.

◆ New-York-City-based Howard Schwartz Recording Studios has become the first independent postpro facility in the US to purchase an SSL Axiom Digital Production System. The Axiom purchase follows the facility's claim to being the first New York owner of ScreenSound and before that an SL6000.
SSL, UK. Tel: +44 1865 842300.

◆ Australia's ABC Television has purchased ten Micron TX-501 UHF/MDR 550 pocket transmitter-diversity receiver sets for use on its award-winning Four Corners current affairs programme production. Also in Australia, the Sydney-based hire company, Jands Production Services has invested in 32 Meyer Sound MSL-4 active loudspeaker systems.
The Meyers are expected to see service in productions ranging from 'Disney-type shows' to jazz festivals.
Micron by Audio Engineering, UK.
Tel: +44 171 254 5475.
Meyer Sound, US.
Tel: +1 510 486 1166.

◆ The 1996 Olympic Games is set to benefit from the use of some almost 800 Audio Technica microphones. Set in Atlanta, Georgia, the Olympics' broadcast body, AOB, will source all aspects of sound and vision for distribution through the world's media.
AT microphones performing at the event will include AT4071s, AT825s, UniPlate, UniPoint and AT4073 shotguns.
Audio Technica, US.
Tel: +1 216 686 2600.

◆ The BBC has recently ordered four Akai DD1500 DAWs—one for the Arabic Channel, one for the Japanese Channel, one for BBC Postproduction and one for BBC Leeds. The Leeds operation has also purchased a DR8 nonlinear recorder. The BBC World Service has recently installed a Sonifex HDX2000 hard-disk automation system at Bush House.
Akai, UK. Tel: +44 181 897 6388.
Sonifex, UK. Tel:+44 1933 650700.

◆ Santa Monica-based Encore Video has installed Graham Patten Systems D-ESAM 400 and 200 Digital Edit Suite Mixers in a new all-digital suite where they will be used in conjunction with an Accom Axial 2020 Edit controller, and Sony DVS-8000 and DME-3000 DVE.
Graham Patten Systems, US.
Tel: +1 916 273 8412.
Sony, US. Tel: 201 930 1000.

◆ London's Goldcrest post house is the first UK facility to have installed an Otari RADAR nonlinear system. Goldcrest claims to be one of the largest post facilities in the UK and handles a wide variety of programme material.
Otari, Japan. Tel; +81 4 2481 8626.
Goldcrest Films & TV, UK.
Tel: +44171 437 8696.

◆ America's Kansas City broadcaster, Bonneville, has recently purchased four Orban DSE7000s and Minneapolis' ABC O&O KQRS-KEGE broadcaster has added two further DSE7000s to its strength.
Orban, US. Tel: +1 510 351 3500.

soundings

information, Digidesign Development Partners' sites, and other sites related to digital audio. http://www.digidesign.com.

Re-pro, the Guild of Recording Directors, Engineers and Producers has just opened a site containing information about the organisation, how to join, newsletter articles, current issues and news of forthcoming events. http://www.aprs.co.uk/repro/ Re-Pro's e-mail address is repro@aprs.co.uk.

Studio designers and builders AVD are claiming to be among the first acoustic construction companies to have its news and views swirling around in Cyberspace. http://www.avdco.com/

Liverpool's Parr Street Studios is making it presence felt with information on all its studios as well as gossip about recent sessions. http://www.connect.org.uk/merseymall/parr/

APT has announced a new interactive web site featuring a searchable database to enable studios using WorldNet codecs to quickly locate other studios using the same technology. With over 500 studios now connected to WorldNet, the site enables users to search by country and city. http://www.aptx.com

Giving full track listings of its 12 sound effects CDs and custom sound effects service, Bits & Pieces' new web site is at http://www.users.dircon.co.uk/-bits/

Bigger than the bible, and with considerably more jokes in it, the web site that will stop you doing any productive work for weeks at a time is here, and it's from Sonic State. Aside from the power-house of industry listings, there's reviews of pro-audio equipment and multimedia software. There's also the world's first stolen equipment directory. You will weep with joy. http://www.sonicstate.com

Apogee's new web site includes data on its entire product line. A downloadable area also provides access to a range of useful files including the manual and installation information, plus screen shots, for MasterTools. http://www.apogeedigital.com/

Anyone using computers to create music should visit Opcode's new site, which includes news, information and downloadable software. http://www.opcode.com

NetRadio Network, the first live 24-hour Internet-only radio network uses real-time audio to offer Net users an experience similar to listening to the radio. The music plays in the background while users visit other web sites, answer e-mail or do any other Internet-related function. Sad but true. http://www.netradio.net

Studio Sound's e-mail address is cz73@cityscape.co.uk.

NICK SMITH

DIGIDESIGN AND MACKIE have announced and agreement on a joint product development strategy. Initial plans call for the development of a new low-cost hardware control surface for Pro Tools as well as other DAE (Digidesign Audio Engine)-based products. Scheduled to be released later in the year, it will feature touch sensitive moving-fader technology, transport controls and basic editing functionality.

Dave Froker, Digidesign's VP of Product Marketing, said: 'For Digidesign and Mackie to be working together on new products is a very natural development considering many existing Digidesign customers already use Mackie products as an integral part of their digital audio workstation setup.' NICK SMITH

INITIALLY BILLED as the first off-line MPEG encoding facility in the UK, London's Town House appears to have a hotter baby on its hands than it's prepared to admit. Talking to

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Eurofin Technologies' Peter Williams (Senior Consultant for the project) it quickly becomes evident that the level of sophistication of the facility is unequalled outside a couple of US sites and Philips itself—and none of these are oriented towards offering a commercial postproduction service.

As a potential world leader then, the Town House now offers advanced MPEG video and audio encoding service for CD-i (and forthcoming formats). The facility is based around Philips' Delta VS system running on a 32-bit RISC Sun Sparc platform with a peripheral Androx-Sun video capturing stage fed from Betacam or Digital Betacam. Audio is transferred to DAT and preprocessed separately, being run through a nonlinear audio system first if necessary, before capture at 44.1kHz via a Digigram PCX5 board.

The combined MPEG encoding is performed in non-real time and may involve several passes in order to optimise the process. The advantage of this processing-heavy approach over cheaper, real-time processing is readily evident in the quality of the encoded video—movement suffers little of the blurring common in lesser systems and the artefacts thrown up in video dissolves is almost eliminated.

The MPEG post suite marks the Town Houses's first move into video and represents part of a £1.5m investment in post. Later in the year, the Town House is looking to play a significant part in the DVD market with full MPEG2 coding. **TIM GOODYER**

SONY PICTURES recently unveiled a pair of new digital-ready rerecording stages at its revitalised Culver City postpro complex. The 300-seat Kim Novak and William Holden theatres offer identical 244-input, 72-bus Harrison-GLW MPC-1 digitally controlled analogue consoles equipped with 256 x 256 routeing matrix. Each room features an 8-channel monitoring system, Sony Theatre speakers, 35mm Kinoton high-speed 35mm–70mm simplex projection systems. Also included is a producer's soundproof conference centre.

The new dubbing stages are linked to an all-Sony digital machine room that houses a number of the firm's hard-disk recorders and players; PCM-800 MDMs and PCM-3348 DASH machines plus access via fibre-optic and similar links to other hardware located on the lot. Manufactured by Sony Electronics the new digital player-recorders were developed by the Advanced Digital Systems Group. Magnatech 35mm 6-track, single-stripe recorders and reproducers are housed in a companion machine room. Also available are Sony APR-series and Ampex 2-inch, 24-track analogue multitracks, 3348s and PCM-800/DA-88s.

The Stage 3 and Stage 5 complex houses a sitcom rerecording room, ADR suite and Foley suite effectively doubling the size of available facilities. ADR1 and Foley A each include MPX-3000 mixers while the large sitcom rerecording room houses a 70-input, 40-bus MPC-1 console.

The system is based on a collection of digital audio servers that can be assigned via a high-speed network to individual editorial suites and linked to the dub stage. Inload systems will also accommodate digital audio in a variety of formats or native sound files that can be directly transcoded to the Sony format media.

'In essence,' commented Executive VP Michael Kohut, 'we have been looking at the way these tasks have been handled for 50 years using mag, and then expanded that ability as we move forward into digital.'

In terms of the facility's choice of consoles, Kohut admits: 'We have always been a big supporter of the Harrison MPC. One difficulty we faced when we started modernising [this facility] six years ago was that we couldn't wait until digital became a reality. But we learned that Harrison planned to develop the MPC-1, a new digitally controlled analogue console that could be upgraded to digital. We were excited by the fact that we would be able to upgrade the console as we progressed from analogue to digital.'

Evaluation of the prototype all-digital system begin this year. **MEL LAMBERT**

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Dirty solutions department

Following the controversial exposure of Sony's DSD system to the press, its technical merit is open to appraisal. Does it offer a future for digital audio?

The lid now seems to be well off regarding Sony's Direct Stream Digital recording system, consequently I don't feel I need to remain silent about my experiences. I was invited to a demonstration of DSD at West Side some time ago and this piece is based on my experience then. The thing that sticks in my mind about the event was that the presentation was couched in terms which appealed to one's emotional side rather than the rational side.

Observations such as David Kawakami's bizarre 'Digital is getting very analogue in character' gave me more reservations than the Red Indians. When I tried to steer the debate in the general direction of technology I was met with total silence. This might be considered polite behaviour in an advanced civilisation such as that of Japan, but we European peasants find it gets up our noses.

Sony claims that DSD is a new improved method of recording audio, but seems to be completely unable to explain why. I intend first to set out what I understand DSD to be. Readers must understand that I have had to make up a lot of it from first principles in the absence of hard facts, but it is to be hoped that Sony will point out where I have gone wrong. In many of today's digital audio systems, the A-D converter uses a high oversampling factor so that the word length at the convertor element can be reduced. Using noise shaping, the convertor element can be reduced to a single comparator outputting a single bit per clock provided that a very high clock speed is used. To obtain a conventional PCM (16-20 bits at 48 kHz or 44 kHz) output, a digital filter called a decimator is necessary. This trade bandwidth for resolution, lowering the sampling rate while increasing the word length. We record such PCM on virtually all digital-audio formats, including CD, DAT, DASH and so on. On playback, again many D–A convertors use oversampling, and similarly, in the limit, the A-D element can be a single switch, driven from a 1-bit signal. Again a digital interpolator is need to convert the PCM to a 1-bit signal. What Sony appears to have done in DSD is to omit the decimator and the interpolator from pair of 1-bit convertors. The recorder handles the single-bit signal. Naturally omitting the decimator and interpolator must remove two potential sources of degradation, but before we congratulate Sony on this move, we need to know how much degradation these stages actually cause and whether this is audible. I asked Sony whether it had made such tests, but I have yet to have an answer. Also before we congratulate Sony on DSD, we have to face the fact that recording the 1-bit signal required a much higher data rate than regular PCM. DSD uses 64x oversampling, so requires four times the bit rate of a 16-bit PCM system. That's four times as much tape or disc space, four times the transmission bandwidth.

It's a sad but incontrovertible fact that (compression apart) PCM is the most efficient way of recording a waveform. The 16-bit words on a CD have 65,536 combinations, whereas a single bit in DSD has only two. For combinations read information. Consequently, recording single-bit audio is the more inefficient method known. 32,000 times less efficient in fact. The only way that DSD will work at only 54x over-sampling is that it has to use noise shaping, which requires a filtering process. I asked Sony how it arrived at the bit rate which DSD uses: specifically I asked what psycho-acoustic data were considered or what research was done to choose the bit rate. I am still waiting for any evidence that they heard the question. I am sure that a better solution exists using the same bit rate as DSD. It has to; Shannon's theory has yet to be disproved. A further problem with DSD is that the only way a DSD bitstream can be faded or post-produced is to convert back to PCM.

AT THE DEMONSTRATION

I heard, it was clear to the whole invited audience that there was something wrong with the low frequency reproduction and there was a contraction of the stereo width when comparing DSD with anything else. It seemed that the problem must have been in the analogue circuitry either preceding the A-D or following the D-A. As I said to Peter Eastty, at 64x over-sampling bass frequencies are virtually DC and it is inconceivable that the problem was digital in origin. If Sony wants us to believe that it should listen to what it makes before we have to. As Barry Fox has pointed out, Sony has a monumental archiving problem in the CBS vaults. It is my belief that a urgent solution is required, one which was in some way (emotionally if need be) perceived to be better than PCM. What I believe Sony did was to take existing 1-bit A-D and D-A and produce a quick-and-dirty solution by getting rid of the decimator and interpolator. The bit rate was, I suspect, not chosen at all, but was dictated purely by the characteristics of the convertors which were to hand. There is no great harm in this. We have all been forced into quick-and-dirty solutions at one time or another. I don't have a problem with Sony using DSD to copy the CBS archive, provided it sorts out the analogue stages, it should do the trick provided it doesn't mind the enormous data storage that it will need. but that's Sony's business.

> Omitting the decimator and interpolator must remove two potential sources of degradation, but before we congratulate Sony, we need to know how much degradation these stages cause and whether this is audible

What I do mind is a quick-and-dirty fix being put forward as a great break-through. It the audio industry does wish to find an improvement in its recording equipment, this should be done at a measured pace, based on some genuine research. In the absence of some coherent answers from Sony I continue to believe that DSD is no more than an interesting diversion.

John Watkinson

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Teach your children well

The snobbery that pervades pro-audio attitudes towards education and training is a damaging luxury for those who practise it writes **DAN DALEY**

Knowledge in our little corner of the world is increasingly notable by its absence. It manifests itself in brief blank stares and the steady proliferation of equipment manuals–which seem to have progressively more in common with advanced weapons systems operating procedures than records ('Align DIP switches on bus card only when in slot and only during Full Moon algorithm cycle! Failure to do so could result in accidental arming and subsequent loss of volatile Eastern European nation!').

Those of you who own studios may also have noticed the prolonged absences of assistant engineers, who are generally out in search of eye drops to correct the vision problems associated with spending all night reading over manuals for the new pieces of gear that clients keep requesting and threatening not to book time with you unless you have and then forget about them a week later anyway.

Simply put, there is a lot of stuff out there to know and learn, and the mountain of it is growing, rendering Sisyphean the effort to acquire and process it all.

So how do you learn the ropes of the pro audio? To listen to studio owners speak of the process, I can't help but be reminded of how law clerks metamorphose into lawyers after 10 or 15 years under the watchful eye of the head barrister. In bygone days there were no diplomas to be handed out; as with Scarecrow in *The Wizard of Oz*, any documentation you achieved regarding your personal fund of knowledge was simply window dressing. Instead, you acquired your knowledge by going out and doing whatever your field of endeavour called for, day after day, year after year, until someone patted you on the back and said something like, 'Hey, kid, here's the keys to the supertanker. Take it out into the inlet for spin. You've earned it. And watch out for those rocks on your left!'

I've done enough stories on the growing academic pro-audio infrastructure in the US over the last five or six years to have gotten a pretty good sense of where audio schools stand in the minds of studio owners. And I'll tell you–these people tend to look condescendingly upon them, partially because of their own careers and partially out of experience with the graduates, some of whom emerge with little or no more than they went in with. However, some things are changing. More and more audio postproduction facilities are finding, often grudgingly but finding nonetheless, that someone who is a graduate of one of these programs is likely to have an edge over someone who isn't–enough of an edge to get them into one of a facility's multiple rooms and running dub sessions and such and earning their keep a bit sooner. Howard Schwartz, owner and chief raconteur of Howard Schwartz Recording in Manhattan, told me just a few months ago that he wouldn't hire an unexperienced assistant engineer if that person had not come through an accredited academic curriculum of audio technology. Like others of his generation, Howard often has a less-than-complimentary opinion about the worth of the school's effect on neophyte engineers' ability in the studio. But, he adds, the kids at least come in knowing what a digital audio workstation is, if not how to drive it. And they often also come with their own insurance carried over from their days in school, which is a useful thing to have during their obligatory 6-month, minimum-wage internships.

Regardless of how one views the schools ideologically or experientially, the sheer number of individual, incompatible systems out there is placing more focus and lending increased legitimacy to the idea of going to school to become an engineer. But how US studio owners view these schools and collegiate programs–which number well over 100 in the States at this point–is also a function of who is choosing to attend them. And herein lies a major source of the often acrimonious perceptions that arise between audio schools and studios. No-one can ever accuse American business of under-utilising hyperbole, and hype is a regular recruitment tactic of audio schools. When recording studios show up on television –which they often do as settings or music videos and celebrity interviews–they look enticing to the mind weaned on MTV. The well-proclaimed placement policies of audio programmes often appear as tickets into showbiz careers, and low failure rates at schools seem to indicate that even students with less than average aptitude are encouraged to finish and pay a full tuition. Of course, the placement after graduation often consists of a nonpaying internship, and that in, and of, itself is not always sufficient to weed out less serious entrants. It's usually the period after that probation-ary six months are up that tells the tale. But go to school they do, in ever-increasing numbers. And to the benefit of pro-audio's future, many of them go on to become engineers, either because of an academic background or in spite of it.

Just a bunch of good ol' boys that never needed no teachin'. Crosby, Stills, Nash & Young shoot from the hip

So studio owners also have to adjust their attitudes towards audio academia. Blanket dismissals of the worth of what audio schools have to offer is counter-productive in the long run. In assessing the value of what the schools put forth, they'll have to spend more time looking at not just what courses the prospective engineer has taken but also what his or her motivations are.

One of the most important questions an employer can ever ask a potential employee is: 'Why do you want to do this?'. When we speak of having 'ears' I've always taken it to embody a combination of intuitive and intellectual capabilities. As the business grows more complicated in terms of its technology, the schools, which can't be expected to teach taste, can help by providing an organised way of assembling and processing all that information. Studio owners can make use of those evolving curricula by accepting the fact that the schools can't instil in their charges either motivation or talent, but can give them the fundamentals upon which they can build the basis of a good engineer in the future.

In a highly technical industry, we can no longer use agrarian concepts for building the future. Teach your children well, but know that they will have to learn things differently from the way previous generations did.

Copycat crimes

While the studio industry seeks to improve quality, the record industry backs a system which compromises the music writes **BARRY FOX**

At first both the IFPI in London and RIAA in Washington simply ignored my many requests for further information on the BBN system. After much time and effort, I found the patent on the system–along with the technical description, this document obligingly informs me that the RIAA funded the patent application. So I wrote about it and–belatedly filled with the spirit of cooperation–the RIAA wants to comment.

To recap, BBN is a means of adding noise to music to electronically 'watermark' it as a means of copyright protection. It's Copycode all over again. While the studio industry busts a gut to improve recording quality, and the hi-fi industry strives to deliver studio quality into the home, the record industry's trade bodies back a system which protects copyright by intruding on the music. They then play the ostrich when asked what effect the intrusion has on the sound. They start to talk sensibly only when the cat is out of the bag.

In a published letter, David Stebbings, the RIAA's new Head of Technology (previously Mr Copycode at CBS), reassures on BBN. The RIAA's system, he reminds, uses psychoacoustic masking to ensure that the code is 'inaudible'. This is possible because the code signal is very similar in character to the accompanying music. So similar, in fact, that if the code is listened to on its own the associated music is easily identified. This means that when the encoded music is compressed by another psychoacoustic process, the embedded signals are not distinguished from the audio and transmitted unharmed.

So when analogue music, with buried BBN signal, is digitised by a compression system like those used for DCC or MiniDisc recording, or DAB or Internet transmission, the BBN watermark gets compressed too, not lost.

According to Stebbings, tests with encoders have 'confirmed' that it is very difficult to separate the code from the audio signal, and the code services several coding and decoding processes. But isn't all this just another way of saying that BBN adds signal-modulated noise?

David Stebbings also reassures that the BBN code survives 22-bit PCM or Sony's new DSD bitstream coding, because the code-to-music ratio of -19dB is maintained right down to the noise floor. The BBN coder has the 'rather easy task' of matching the buried data to the random noise at the floor, which has a low correlation with the embedded code.

'Far from being a problem,' assures the RIAA's technical chief 'high definition or high signal-to-noise ratio systems benefit from the the signalling methodology employed by the proposed RIAA-BBN system.' This seems to add up to the claim that BBN improves the sound by adding signal-modulated noise.

In the case of Copycode, all manner of claims were made by CBS, the RIAA and IFPI, and swallowed hook, line and sinker by the record companies and music press. The truth only came out after the audio industry challenged the assertions.

It may well be that the RIAA is right, and tracking the music signal with a fog of similarly shaped noise at -19dB actually improves the recording quality. The fog may also survive even the most brutal compression coding, as, for instance, proposed by Bell Labs and NEC to make a solid-state Walkman a cost-effective product for joggers by the turn of the century. But if the IFPI and RIAA are so confident that this is true, let them now arrange demonstrations to prove it. This should not be difficult because David Stebbings refers to the test use of 'different encoders' and 'multiple codecs'.

ANOTHER RESEARCH COMPANY is already challenging the RIAA's claims for BBN, albeit as a sales pitch for its new system.

MusiCode was developed by a startup technology company called Aris Technology, in Chelmsford, Massachusetts. Aris is now telling the 'big six' record companies and the music press that MusiCode is a 'one-stop solution' to piracy and the Internet transfer of music. According to Aris, the record company can bury a unique message, such as the name of a song, artist, album and record label, inside the music. So MusiCode will identify pirated software, and track on-line or broadcast transmissions. Or it can trigger sensors in a recorder, to limit home dubbing.

The factsheet on MusiCode may excite the music industry but it is nothing more than a wish list. It lists a string of problems that the music industry would dearly love to solve, but gives absolutely no indication of how MusiCode can solve them. It is like advertising a car that runs on air. A lovely idea that everyone would buy if only someone could make it work.

The only information in the wish list that halfway resembles hard fact, is the claim that there are two versions of MusiCode. One carries data inside the music at a rate of 100bits/s, and another streams data at 1000bits/s. Only the slower data rate can survive recording onto cassette tape. Neither 'degrades the sound quality' and both can survive digital compression.

Aris unambiguously claims that MusiCode is inaudible, while dismissing the BBN spread spectrum technique and Thorn-EMI's ICE notching system as audible or unreliable or both.

I challenged Aris to talk hard fact, or be treated with jaundiced suspicion by all but the ever-obligingly gullible music press. The company is very reluctant to give away any details of how the system works, but has now released just enough useful fact to tease that MusicCode might, just might, be more than a wish list.

MusiCode looks for naturally occurring statistical events within the music. The events are relationships between nearby waveforms. A digital code bit '1' is signalled by many events within a given time. Code bit '0' is signalled by very few. Although small alterations must be made to the music host signal, these are infrequent and take the form of greatly attenuated replicas of the host original. As well as inaudibility, Aris also claims resistance to compression coding.

Like the RIAA and IFPI, Aris must demonstrate to win credibility. But as Richard Gastwirt, Director of Marketing at Aris admits, the company has a special problem. Through the RIAA, the record companies funded the work on BBN. How readily will they now accept that the work may have been flawed, or overtaken by events?

Exactly the same dilemma faced the British record company some ten years ago, when their trade body the BPI funded research on spoiler signals by Southampton University. Wild horses would not drag a copy of Southampton's report out of the BPI. But the organisation learned from the mistake and has kept a low profile on copy-spoiling systems ever since.

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Enter the East

The diversity of culture and language in the Asian region is a obstacle in its path to the world entertainment stage writes **STEPHANIE GOH**

It's a typical Friday night at Singapore's Boat Quay and the drones are filing out of their concrete and steel hives to become individuals again. The uniform is smart casual and the most common language spoken is English. They come from all over the world to meet, work and live sampling the cultural diversity Asia has to offer.

Asia is a complicated place in which two-thirds of the world's population resides. It speaks more than ten different languages, with economies and political systems in various stages of development. These languages, cultures, economies and laws directly influence the creation and distribution of entertainment.

In certain Asian markets, local heroes often sell better than foreign ones—no matter how popular they are elsewhere. Unlike Western Europe and North America where a handful of languages will see you through, almost every Asian country has its own language which its neighbours do not necessarily understand. For example, between the Japanese and Thai time zones, there are at least ten different languages

spoken—including Mandarin, Cantonese, Bahasa Malayu, Bahasa Indonesian, Thai, Vietnamese, Japanese, Korean and various other North Asian languages such as Laotian Cambodian and Myanmese.

Each of these countries—with the exception of Singapore —has a strong market or one that has great potential for growth as their economies transit to capitalism. One of the symptoms of growing pains is piracy which is a still flourishing trade even in the more economically developed nations like Hong Kong, Thailand and Malaysia. Piracy, however, is a necessary evil which offers cheap and ready access to entertainment to a greater part of a disadvantaged population. We're encouraged to believe that think piracy damages everyone concerned but out here it offers an idea of what other cultures have to offer. Western music and even music from their own region is introduced to them in the form of pirated product.

'This creates a demand within these societies to produce their own product and when they start producing and exporting their own product, they will join the fight against piracy in order to protect their own product,' says Keith Ng, Strategic Marketing Manager, EMI Singapore.

TWENTY YEARS AGO, pirated product was readily available from most record stores in Singapore or Malaysia—in broad daylight, at S$2.00 a cassette. These days consumers tastes have changed along with their economy and lifestyles; as society becomes more affluent, the consumers buy better audio and video hardware for their homes, cars and offices, they also want quality software. Consequently, consumers are currently prepared to spend up to S$25.00 dollars on a CD.

The Singapore market mainly consumes foreign product because there is little appreciation for local talent. As a result, local singer-songwriters such as Eric Moo and Dick Lee have had to seek friendlier markets in Taiwan and Japan in order that their record companies could use their success abroad to sell them to their home market. Talent is difficult to nurture in Singapore because emphasis is placed on education towards commerce rather than the arts. Tertiary institutions are now offering educational programmes in film, video and performing and graphic arts in a move to rectify this shortage of creative talent. According to Barry Butler, Managing Director of Speakeasy Digital, Singapore still has a way to go before it can be a creative hub.

'A certificate in audio engineering does not necessarily make an engineer who knows how to bring out the best in a piece of work. It has to come from inside. It takes a passion for the work that takes nurturing. This is only the first generation.'

In agreement is Lillian Chan, Managing Director of Schtung Music Singapore. She believes that local talents have a chance if they are given

'Once we in the business achieve some amount of success, we have to put it back in terms of investing in local talent in order to help the industry grow. Singaporeans seemingly lack talent compared to Malaysians, Indonesians or Thais who have their own defined culture through which they can express themselves.'

These three countries have their own local artists who do very well both at home and internationally, selling into territories which speak their own, or similar, languages—Malay and Indonesian, for example, are sufficiently similar to allow a large crossover. Convenient, because Indonesia boasts a very large population and ready access to media networks.

In contrast, Hong Kong has a self-sufficient market and exports it's products with great success because 'Canto-pop' not only appeals to the Cantonese and non-Cantonese speaking in Malaysia Singapore, Thailand, Taiwan but also to the large concentrations of immigrants settled in North America. The situation is neatly illustrated by the fact that Jackie Cheung is one of PolyGram's top four best selling artists worldwide—after label mates like Bon Jovi and Sting. When it comes to live shows, Anita Mui has proportionately as many sold out concerts as the Grateful Dead.

Asia is finally coming into her own with the establishment of channels such as MTV Asia and Channel V; and satellite broadcast stations such as MEASAT that cater specifically for the regional market. Facilities such as Synchrosound Studios in Kuala Lumpur, Malaysia now house some of the best equipment money can buy—from the likes of AMS Neve and Solid State Logic. As in Europe and the US, it is here to offer world class production centres to its local cultures. This heralds an exciting new age in the audio-visual industry as Asia clears the hurdles of cultural difference, poverty and civil war, and prepares to compete as an equal on the world stage.

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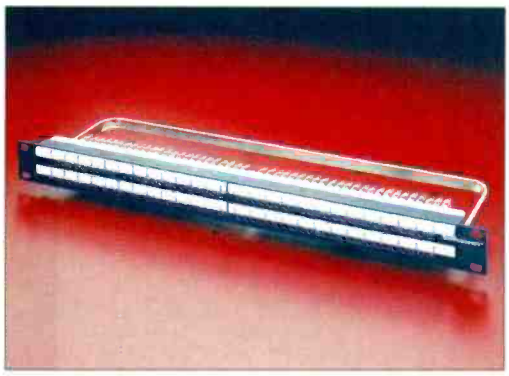

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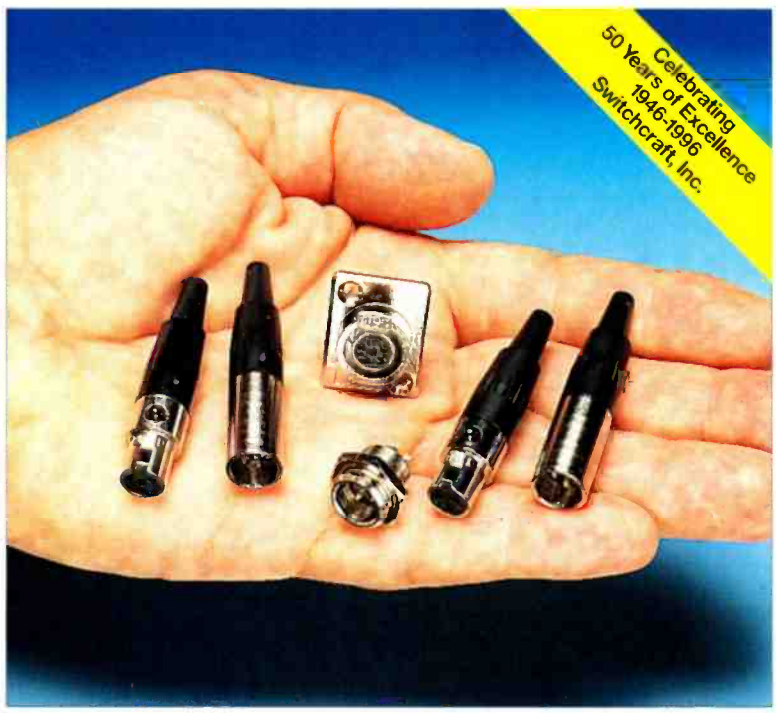

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Stage Tec CANTUS

As the live and broadcast sectors move progressively away from hybrid and into totally digital systems, one German company has responded by producing a truly versatile high-end digital console writes PATRICK STAPLEY

THE GERMAN PRO AUDIO market has a reputation for embracing new technology with a certain alacrity. Witness the early adoption of digital consoles by German broadcasters and the number of Neve DSP desks that were sold into that sector. Germany has been an important market for digital console manufacturers, and according to Stephan Salzbrenner, Managing Director of Stage Tec, the sound-reinforcement sector has also recently championed the technology.

'We see more and more German theatres installing digital systems,' he says, 'and I would say that any major theatre now thinking of a refit here would install a complete digital system with, of course, a digital console. Analogue is seen as the technology of the past.'

Stage Tec has responded to the demand by producing a high-end digital console of its own which has initially been aimed at the live and broadcast sectors—although it is equally suited to post and tracking.

Stage Tec was originally formed in 1989 as a sister company to the Salzbrenner installation company, and was set up to specialise in stage-related technology and act as a pro-audio distributor. Three years later a second Stage Tec company was created to cover the broadcast sector with ex-Managing Director of Studer Revox, Walter Derrer, brought in to run it. In addition to the planning, distribution and installation activities of the new company, the concept further evolved, and by 1993 a third company, Stage Tec Development, was created specifically with the aim of developing digital-audio equipment and ultimately a full scale digital console. Stephan Salzbrenner explains the events that turned Stage Tec from a services company to a manufacturer.

'At the Vienna AES in 1993, Sennheiser announced that following its takeover of Neumann it was to close the mixing console division of the company. This meant that there were a lot of extremely well qualified development engineers in Berlin who were suddenly without a job, and this was the opportunity we'd been looking for. Seven of those engineers immediately came to work for us and by the end of the year were joined by a further six personnel. Just seven months after creating the new company we had built and installed our first product in a prestigious venue in Berlin.'

This was the Nexus interconnect and routeing system designed to provide distribution and control via fibre optics for theatres, broadcast, OB work, and so on. This paved the way for the next development stage, the Cantus digital console.

Although Cantus was a brand new design, a great deal of the groundwork had already been covered due to the R&D team's previous experience in developing the Neumann Strategy digital console. This meant that R&D proceeded very rapidly, and by November 1994, just 17 months after the company had been formed, the Cantus console was launched at the Tonmeister Exhibition at Karlsruhe, where it immediately won two orders.

Since then a further 14 consoles have been sold, including four desks to Bayerischer Rundfunk (Bavarian Broadcasting) in Munich, and La Scala opera house in Milan.

Bayerischer Rundfunk was one of the first companies to order the Cantus —interestingly it is also owns the last Neve DSP console to have been produced which is still in regular use. So what were some of the criteria that swayed BR in favour of Cantus? Studio Manager Hans Schmid explains the reasons why Cantus was chosen for the station's newly refurbished Studio 10 radio-drama facility.

'First of all we wanted a digital console, and as far as we were concerned there were four contenders—Capricorn, Studer, Lawo [another German digital manufacturer] and the Cantus. For us the Cantus was best because it was the most ergonomic and it was very easy to get hands to things quickly, which is obviously important for live radio work. Also communications and talkback were very comprehensive which again is necessary for a drama complex like this. Having MS stereo built-in was also a big plus point.

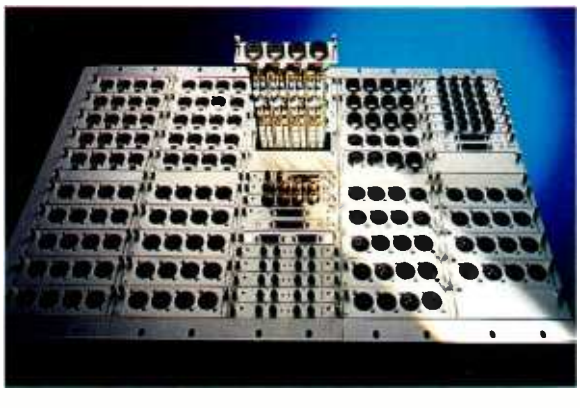
'Another consideration was that being an old studio, there was no provision for air conditioning in the machine room where the console signal-processing rack would be positioned, and the Cantus DSP, which fits into just one 3U-high space, required no additional cooling or air treatment. However, one of our major reasons for choosing Cantus, was that Stage Tec was the only manufacturer to offer the option of a small slave console that could be run in conjunction with the main desk although independently, and this was perfect for our requirements.'

The Cantus system comprises a relatively flat design, modular console equipped with its own power supply (and backup power supply) and integrated computer specifically for control-surface functions. Being designed with live use very much in mind, any circuit board may be removed and inserted during operation, and according to Stage Tec a defective or missing board will never cause complete failure.

The console connects to a remote DSP and control rack via a single bidirectional fibre-optic cable. This 3U-high rack provides up to 128 fully featured channels, but this number can be decreased or expanded as required. One rack will also provide 64 mix buses (for groups, auxs, mains, and so on) and an additional 256 signal buses for a wide variety of nonsumming input and output functions. Processing is based on 40-bit extended floating-point architecture and features a channel-orientated structure which basically means that if an error occurs on a DSP board, only individual channels will be affected.

Cantus fully integrates the Nexus system, again using fibre optic, allowing an elaborate audio, control and synchronisation distribution network to be created. Each Nexus base unit can be fitted with a variety of modules to allow for specific input and output requirements, and this includes mic amps allowing base units to be placed as near to the source as possible prior to digital conversion, thus maximising audio quality. A–D conversion is via a proprietary designed 22-bit convertor offering a dynamic range of 124dBA—at a headroom of 22dBu this produces a noise floor down at an impressive -102dBu.

More than one Cantus console can be connected to the same Nexus network allowing simultaneous signal sharing, also



Nexus—the digital-audio routeing and interconnect system

PHOTOGRAPHS BY JO FISCHER

Studio Manager Hans Schmid at the Cantus console installed at Studio 10 Bayerischer Rundfunk (Bavarian Broadcasting) in Munich

a slave console or consoles can be connected to the main desk allowing two possibilities. Firstly a slave can be used as a satellite console, for example in a theatre auditorium connecting back to a main console, where both consoles have access to the same sources and destinations. The second method is as a split configuration where the slave shares the same resources as the master console, but operates totally independently to it.

As with some other digital consoles, Cantus has been designed to be designed, allowing the user to configure the system to suit the way they work, and, of course, being digital it only requires a button press to completely change between one user setup and another.

The control surface itself has been arranged into three distinct operational sections: channel strips, central channel and centre section. The desk operates as a totally free, assignable, structure, meaning that channel strips can function in many different ways depending on how the desk is configured—that is as an input channel, a group, a master output, an aux master and so on. Up to 10 banks or Pages of channel strip assignment can be stored and recalled either locally or globally, thus allowing many different layers of control to be structured to suit the operator. Thus a 36-channel-strip console will effectively offer 360 channels.

All channels include clear electronic labelling to keep the operator in touch with current assignment, and two ID windows are included above the fader: the first shows the current assignment (brightly lit) and the second displays the next page in line to be set (dimly lit).

The desk can be supplied with as many or as few channel strips as the user requires, but an average configuration is between 36 and 56. All channels are identical in appearance and are made up of three modules—Fader Module, multifunction module and Metering Display module.

The Fader module contains the proprietary designed digital opto-coupled motorised fader (mechanics by Penny & Giles) and is topped with the two 8-digit ID windows. It also includes two rotary encoders with associated displays and a selection of keys. The Multifunction module incorporates a further six rotary encoders arranged into two sections (4 and 2) and several more keys.

Metering is again a proprietary design using two high resolution LED metering displays in each unit—these can read mono or stereo programme which can be viewed at three points through the channel—input, pre-fade or post-fade. The meters can also be switched to display gain reduction for dynamic processing. The base of the meter provides a visual overview of the channel strip showing its source, types of processing assigned, output and so on, there are also a further two ID windows which can display channel number and electronic scribble-strip description.

Desk configuration including matrixing to Nexus and individual channel structuring is all organised from the centre section. The main user-interface is a pen and tablet (a QWERTY keyboard also slides out from under the front buffer) which interacts with a colour TFT display built into the meter bridge. From here all interconnection, console architecture, and individual channel configuration is constructed.

Each channel has access to a range of software processing modules: fader, gain, limiter, compressor, expander, gate, high-pass filter, low-pass filter, bell EQ, shelving EQ, notch filters, insert, delay, input, output, panpot, metering. And using the graphic interface these can be arranged and repeated (within the confines of system processing power) for each channel type fitting in with the bussing structure the user has defined, that is the number of groups, auxiliaries, and so on. Other channel features such as metering points, solos, cuts, and so on, are predefined within the system.

Although each channel has a finite amount of processing allocated to it, the system does not operate on a pool of processing principal where set numbers of processing elements are available to the console. So in this respect, the operator does not have to plan where in the console to distribute channel elements in order to ☞

The Satellite editing room showing small Cantus slave console

conserve processing. As far as processing delays are concerned, Stage Tec claims that whatever processing is assigned to the channel, that interchannel alignment will remain within a single sample tolerance. However, provision is made to introduce delay into the channel signal path to compensate for external processing, time alignment of microphones and so on, and this can be added either in milliseconds or samples.

Channel operation may be carried out in two ways, either from individual channels in the more traditional sense, or from the central channel. The central channel provides a full complement of controls for all channel functions, thus maximising the channel-display element and making adjustment much quicker as well as concurrent. Where more than one type of processing block has been incorporated into a channel—for example two 4-band equalisers, the relevant section can be Paged to provide access to the additional parameters. Control sections that have not been included in the accessed channel will simply blank out, thus a channel that has not been assigned dynamic processing will be readily visible as no values will light-up in the dynamics panel.

The individual channel strips have been constructed with a mix of dedicated and assignable controls. At the top of the channel is a simple routeing matrix that accesses eight groups (either mono or stereo), to extend beyond this the user makes selections centrally.

Other fixed control switches include solo, cut, insert points in-out, automation status, fader group assignment, paging, and so on. The remaining switches assign the various channel functions to the eight local rotary encoders such as EQ, dynamics, auxiliaries, gain, panning and so on. With the more elaborate functions such as EQ, the function button will cycle through the various control pages. Additionally, there are two MASK buttons that can be programmed by the user to custom assign parameters to the rotary controls.

The rotary encoders which have been specially adapted by Stage Tec, feature tall, slimline, rubber clad knobs which have been offset to minimise visually interfering with their respective display areas. The knobs include gearing logic, so that a fast turn will cause a coarse change while a slow turn will implement a much finer change. Each respective display area is built into the face of the channel to the left of the knob and includes a 2-colour arcing scale (16-LED) display and a 4-character alphanumeric. The knob itself includes a push down switch action, and this provides a toggle for the display switching between function descriptions and a numeric value for the setting.

Channels can be mono or stereo, and stereo channels include a split function that enables the left and right leg to be split across two channels at the press of a button, thus enabling easy offset adjustment.

Apart from system setup and manage-ment, the centre section of the console contains the usual array of control-room-monitoring controls for main and close-field speakers, communications with talkback to up 18 destinations (a customised panel is also available), full internal and external monitoring with provision for 80 sources (here again can be specifically customised), machine control with a selection of drivers for popular machines including most recently integration with Sonic Solution's systems.

Also included centrally are all the controls for static and dynamic automation. At present the finishing touches are being added to the dynamic automation which is scheduled for release around or just after the Copenhagen AES. However, the static automation is operational and allows up to 100 console wide snapshots (per project) to be arranged in sequences and recalled for scene change during live applications and so on. Crossfades and dynamic snapshots will also be available shortly. Automation data along with general project data can be stored on the system's M-O disc for archiving purposes. Another aspect currently being developed is multichannel monitoring and Stage Tec says that the console will be able to accommodate all formats from Dolby SR up to 8-channel.

In the short time that Cantus has been available it has proved extremely successful, and signs are that this will continue. Stage Tec is aware that sales so far have been confined to its home and neighbouring markets, and is currently planning marketing assaults further afield: this will begin in the summer with a tour of the US and Canada. Meanwhile, Cantus is on show at the Copenhagen AES and if you have the chance, its well worth having a look at this fine piece of German engineering.

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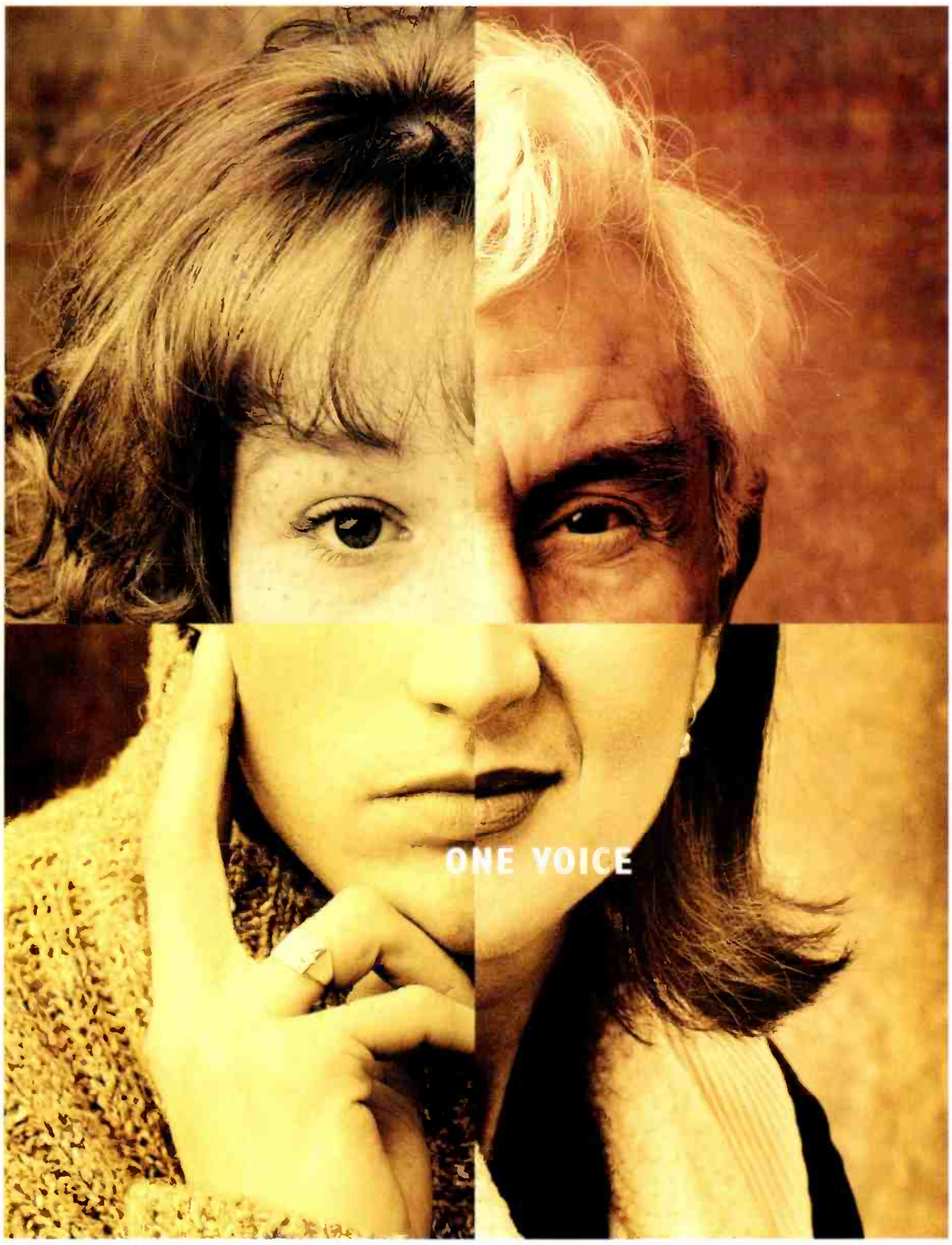

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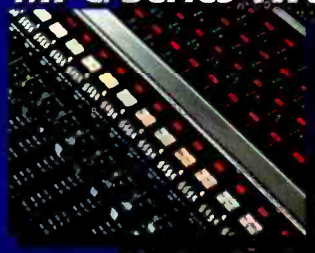

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Fostex DAT RECORDERS

Subject to constant revision, ranges of professional DAT machines can prove confusing. JIM BETTERIDGE runs down Fostex' current range of machines and places them in their appropriate working environments

UP UNTIL A FEW years ago, anyone wanting a time-code-capable DAT recorder, bought a Fostex D20. It was simple. The D20 was the only DAT machine Fostex made and—to the 'surprise' of various august bodies—with it the rebel newcomers managed to establish an international industry-standard for professional DAT applications using their own unique format for time code.

In January 1991 the IEC time-code format for DATs was approved as the official industry-standard and now several manufacturers make excellent IEC time-code DAT recorders. The Fostex range, now also IEC compatible, has expanded to include two portable machines (PD2 and PD4) and four rackmount models (D5, D10, D25 superseding the D20, and D30). All machines provide balanced +4dB analogue connections on XLRs and full AES-EBU, and SPDIF optical, digital interface connections. With the exception of the D5, all have IEC time-code facilities to some extent or another. Also, all Fostex machines, again with the exception of the D5, support all 799 ID numbers given in the DAT specification rather than the more usual 99 provided on most machines. This is particularly handy when storing lots of short recordings such as effects or stings-jingles and so on.

THE D5 comes in at something under a grand in the UK. It's no secret that this machine is based on a Pioneer model, though with a number of Fostex customisations, plus, of course, a Fostex cosmetic, and their professional service backup. It's designed as a good basic workhorse machine and will record digital or analogue signals at 44.1kHz and 48kHz. It will also record analogue or digital at 32kHz using a 12-bit word to double the record time (Long Play). Recording digitally, it is also possible to select 32kHz, 16-bit standard play with no extension of recording time. The D5 is a 2-head machine and hence cannot be used for punching in or out. It does not support SCMS and so multiple digital copies are not a problem.

A wireless remote is provided; this works well but there is only one channel so should you have two machines, operational confusion might ensue. There's also a 5-pin DIN socket providing GPI (General Purpose Interface) facilities for Stop, Play, Skip forward and Skip back. As is usual with GPIs, the D5 uses the common TTL standard whereby each pin is held at a nominal 5V and momentarily shorted to earth (through a switch) to activate the command. In this way virtually anything with switch contacts can be used as a simple remote control —it's one way of getting around the limited remote facilities.

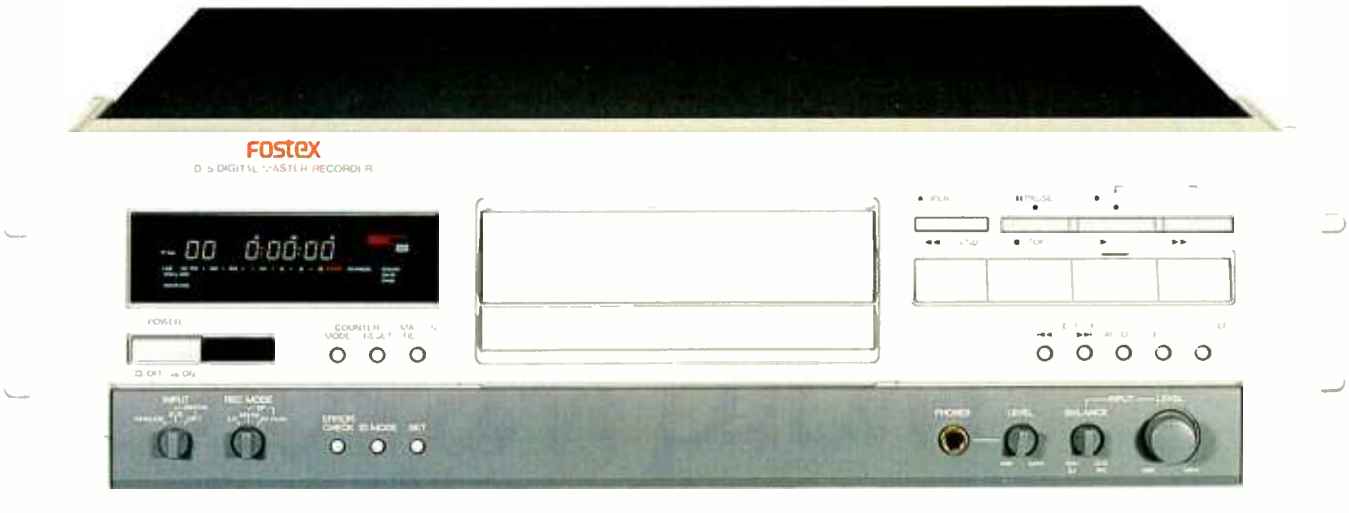
Tascam have also produced a machine—the DA-20—based on the same Pioneer model. For professional applications, the two main differences between this and the D5 are that the latter has balanced switchable -10dB/+4db analogue inputs and outputs plus balanced AES-EBU and optical SPDIF connections, as compared to the unbalanced -10dm analogue and the unbalanced (phono) SPDIF connections on the Tascam. Also, the DA-20 lacks the D5's GPI ports. If these refinements are unimportant to you, the Tascam is about 20% cheaper and otherwise virtually identical.

THE D10 was the first DAT machine to utilise onboard RAM to provide instant start and cue-to-start-of-programme functions —facts reflected in its £2,295 (UK, ex VAT) asking price. These features are clearly very useful for cart-style broadcast applications and, in conjunction with the GPI ports, also allow basic but accurate machine-to-machine editing. It is excellent for voice work or programme assembly and even capable of music edits.

An optional board (the 8333) provides basic time-code facilities, making the D10 the least expensive time-code capable DAT machine on the market. With the 8333 in place you can record code simultaneously with programme. Unlike with the D25 and D30 (and the D20), however, it isn't possible to insert time code—that is to go into record on the time-code track alone. Nor are there any onboard chase synch facilities as found on the D25 and D30 (and D20B) or even any facility to hook-up an external synchroniser—this really is a master-only machine.

Another limitation for working to picture is that there is no video sync input Via its AES-EBU inputs it will, of course, lock to incoming digital clock and so if recording digitally within a synchronised system, proper lock-up would be possible. If working analogue, however, you're stuffed.

The D10's display is quite comprehensive. A button marked DISP TIME allows you to step through three time display formats: A-Time, R-Time (the time element of IEC standard time code) or Date Pack ☞



Fostex D5: in direct competition with Tascam's DA-20

Fostex D25: successor to the D20B

—this gives a continuous record in years, months, days, hours and seconds, of when the recording was made. It's part of the standard DAT format, although not all machines have it implemented. In addition to two large bar graphs (28-segment fluorescent tube), there's an associated numeric display can be switched to show headroom or error rate.

The jog-shuttle wheel on the D10 has two parts: the outer ring which operates in the style of a shuttle wheel (the further you turn it the faster the replay speed) and the inner part which works in the style of what Fostex call a digital jog wheel. A unique Fostex invention, this seeks not to simulate the effect of reel-rocking analogue tape but instead repeatedly loops a single DAT frame around the current position (from RAM), thus producing a rather unpleasant digital chattering. Though initially unsettling, a few minutes of experimentation shows that it is in fact a very quick and accurate method of finding an edit point. For approximate location of a position the wheel can be used in the Search Cue mode to actually move the tape against the heads, producing a low quality, interpolated audio. This allows shuttling from 0.5x to 10x play speed and enables you to find a given point quickly, to an accuracy of a few frames.

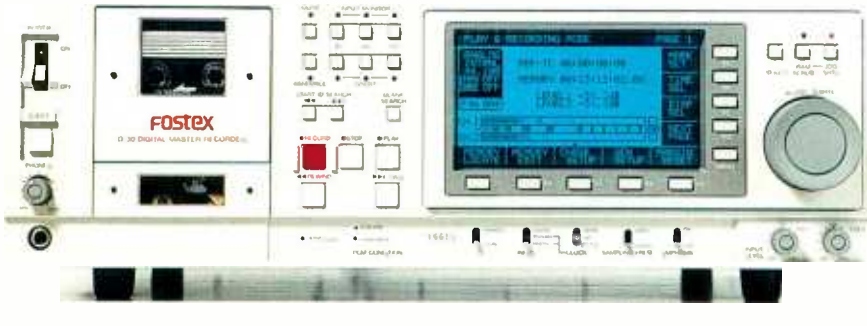
Strangely for a professional machine, there is no wired remote for the D10. It comes with a comprehensive wireless remote switchable between Channels A and B allowing you to independently address two machines (the machines, too, can be set to either A or B). Again, its GPI ports offer an alternative if you fancy a bit of DIY.

The D10 has sold well into broadcast, it's instant-remote start and machine-to-machine editing, time-code and 9-pin facilities make it far more flexible than a standard twin-head DAT workhorse. It is, however, still a 2-head machine and hence is not capable of gapless punch-ins or confidence monitoring in record. If you want such facilities you'll need the four heads of either the D25 of D30...

THE D25 replaces the D20B. At £4,845 (UK, ex VAT), it has everything the D10 has in terms of onboard RAM, instant start and machine-to-machine editing using GPIs, and its display is very similar. The main differences are that it has four heads allowing gapless punch-ins and punch-outs along with confidence monitoring; you can record on any of its three tracks independently, it has an onboard time-code generator-reader and chase synch including a high-speed code reader, more comprehensive 9-pin facilities and an optional wired remote control. There are a couple of oddities in this list: you can only gain access to independent left-right channel record ready control via 9-pin. The same goes, possibly more seriously, for varispeed, so if you're just an humble sound studio, you're confined in those departments. The 'solution' is to buy a stand-alone 422 controller. Another odd thing is that the remote isn't 9-pin based (as the D20's was) but connects to a parallel port on the D25 and hence doesn't have any kind of alphanumeric display so there's no way of knowing where you are unless you can see the front of the machine; nor can you access the varispeed. The official answer is most users work in audio post and so have 9-pin controllers anyway.

THE D30 is the top of the range DAT recorder. Though there are many excellent features to be found in the details of its operation, the main feature that sets it apart from the D25 is its user interface. The machine has been designed for the video-audio postproduction market and hence its look and much of the terminology used is similar to that of a digital VTR. The display is like a cut-down version of a digital VTR and has five soft keys along its bottom edge and down its right-hand side. The idea is not necessarily to offer any greater level of parameter editing on the machine, but rather to make it easier to access and display more information at once. Although this is largely true and effective, once in Record, it isn't possible to access display pages other than the Record page. This can be quite frustrating. Having said that, the amount of information available is generally very helpful and greatly missed on returning to my D20B. The remote controller situation for the D30 is the same as with the D25, except that in addition you are required to buy an 8334 37-pin parallel board to allow it to work with the D25 remote. Suffice it to say, they're not big on remote controls at Fostex.

On a positive note, when looking at the costs of the D25 and D30 it's important to remember that they come more or less fully loaded with time-code facilities, instant start, chase lock, digital inputs and outputs, and so on, whereas on other makes of DAT such fairly basic features can come as optional add-ons which add considerably to what might initially seem a lower price.



Fostex D30: top of the heap and purpose-built for postproduction

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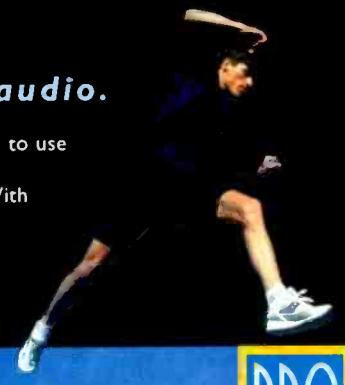

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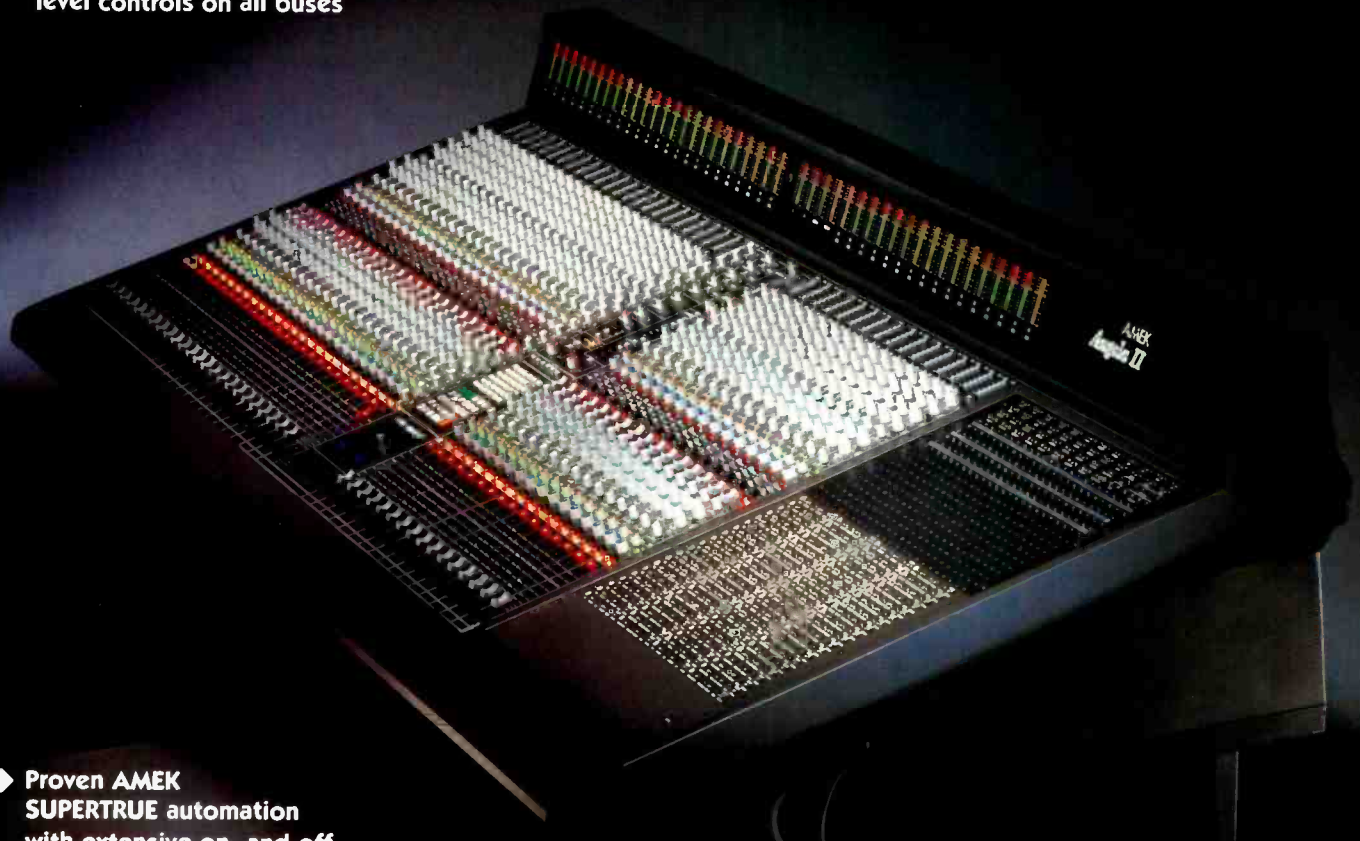

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Crown STUDIO REFERENCE II

Crown's Studio Reference series of amplifiers could be described as another black box for the rack, but as **WADE MCGREGOR** finds out, these amps are a little bit special

CROWN INTERNATIONAL moved into the power-amplifier market many years ago when a Philadelphia doctor (and hi-fi enthusiast) convinced them that their DC power supply, built to solve a noise-floor problem with batteries, could also be used as a audio amplifier. The DC300 went on to become a standard studio—and, to some extent, live—power amplifier in the late 1960s. In the latter part of the 1980s, Crown re-examined the needs of the professional, studio, monitor chain, developing the Macro Reference amplifier introduced in 1992. The Macro Reference was designed to be the ultimate in sound quality available in a professional power amplifier.

The success of this model has led Crown to update the series in the form of the Studio Reference I and Studio Reference II amplifiers, incorporating many of the suggestions made by users of the original. The Reference I is rated for an output of 780W/channel into 8Ω and 2,315W into 2Ω in Parallel-mono mode. The Reference II is rated for an output of 355W/channel into 8Ω and 1,115W into 2Ω in parallel-mono mode.

Parallel-mono mode was developed by Crown as a way to achieve the high-currents demanded by extremely low impedance loads. Connecting the outputs of each channel in parallel side-steps the problem of doubling of output impedance that occurs in bridged-mono mode and maintains the damping capability while reducing the thermal stress on the output devices. This is a product of the grounded-bridge output topology used in Crown amplifiers.

The Studio Reference series of amplifiers could be described as another black box for your rack except the black finish has more in common with a grand piano than typical audio gear. The sound quality of the unit also glistens, providing the level of detail and clarity associated with the very best amplifiers available. Noise is over 117dB (A-weighted) below full rated power and the power supply includes line filtering to reduce mains-borne noise, as well.

The extremely low output-impedance of this design provides a damping factor in excess of 20,000 at low frequencies and remains above 2,500 at 1kHz. This is coupled with excellent control of distortion products (reaching 0.1% THD at full rated power but typically less than 0.02% THD and 0.005% IMD 36W to full power, rising to 0.025% IMD at 36mW).

I COMPARED a Reference II with some other common studio and sound-reinforcement amplifiers driving easy loads like the Tannoy's System 800 close-fields and weird loads like Quad Electrostatics. It was always well behaved and transients were not exaggerated or muffled. The character of the loudspeaker was fully represented; imaging was solid; the bass was very tight and punchy; and the low mid-range sounded slightly warmer than any of the other amps.

High quality build and superb sound combine to create the best amplifier Crown has built

The circuitry includes the same rigorous protection as the Crown sound-reinforcement series, yet the sound quality is superior to the sound reinforcement amps auditioned. The Studio Reference amplifiers include Crown's ODEP (Output Device Emulation Protection) circuitry, which models conditions in the output devices and reduces the drive level if the devices are leaving their safe operating area. This protects the devices by activating a form of limiting. The output begins to pump like a badly set compressor when the amp is driven heavily into ODEP limiting. However, unlike a conventional limiter, ODEP has the advantage of passing short-term transients untouched as they don't threaten the output devices and therefore don't trigger protection limiting. This maintains the sound quality of the amp when used near full power and offers prodigious levels of very short term power. ODEP also protects the amp if other forms of overheating occur, such as a blocked air filter or cooling fan failure.

The temperature of the toroidal power transformer is also monitored and if it gets ☞

too hot a switch will kick the amp into Standby mode. Standby mode occurs for three seconds after startup to avoid amplifying noise of circuitry stabilising, especially that of the mixing console after a power outage. This mode will also be engaged if other faults are detected, such as overvoltage on the mains supply, or amp failure resulting in DC voltage or common-mode current at the output. The standby mode shuts down the high-voltage supply from the output devices and can also be engaged by optional PIP (Programmable Input Processor) modules that plug in the back panel of the unit.

The PIP module supplied with the unit is simply a pair of XLRs and a minimal input filter circuit but the PC board includes pads, traces and jumpers for the user to build input filtering, add transformers, or change the input polarity. More sophisticated PIP modules can be installed to provide anything from crossover functions to full computer control and remote status monitoring. The one that may be most useful in the studio is the Smart Amp module which provides a range of capabilities, including configurable loudspeaker protection limiting (interacting with the ODEP) and automatic switch-over to Standby mode when the unit is not in use. Crown is about to release a valve-based PIP module for those that need even more warmth from the amp.

The rear panel also includes two sets (for biwiring) of dual 5-way binding posts for each channel; a MODE switch (bridged-mono, stereo, or parallel-mono); and two balanced 6.3mm phono jacks with a GROUND-LIFT switch. Ironically, the binding posts do not have large diameter holes for fitting heavy-gauge wire, in spite of the fact that this would be the only way to maintain the damping factor of the amp. You will have to cut away a lot of wire strands, resort to spade lugs or banana plugs.

The black finish has more in common with a grand piano than typical audio gear. The sound quality of the unit also glistens, providing the level of detail and clarity associated with the very best amplifiers available

The front panel sports a pair of knobs for adjusting the input gain, the pots include detents and are well matched with approximately 1dB per step over most of their range. Between the knobs are two rows of little LED indicators with Channel 2 arranged as a mirror image of Channel 1. The upper row indicate ODEP status; IOC (lighted whenever the waveform at the output differs from the input by more than 0.05%); and signal presence. The lower row is a series of five LEDs per channel that indicate the dynamic range of the signal by comparing the peak to average ratio. This is not really a meter related to the amplifier's function but as many amplifiers are sited in the front wall of the studio, it offers another form of metering to view during a session. If you dismantle the front panel, you can switch the meter to a typical output power indicator or even turn off the meter if you find it distracting.

Internal construction is impeccable and Crown backs it up with a 3-year warranty. Many details that plague the user, such as noisy cooling fans, have been considered. The fan is placed in the middle of the interior where it is least audible and only comes on momentarily until the unit returns to the safe operating area, then it shuts off. During my listening tests the fan didn't even turn on. The front panel protrudes 7cm from the rack and includes covers to hide the mounting hardware, giving the unit a very solid appearance.

Clearly, this is the best sounding amplifier Crown has ever built. Reliability is enhanced by protection circuitry which stays out of earshot for all but the most extreme monitoring situations but will provide protection against that slip of the SOLO button. If you can find big enough loudspeaker cables, you can take full advantage of the control this amp can have over the driver motion. Just make sure you also give the unit a stiff mains supply so it can perform to its full potential.

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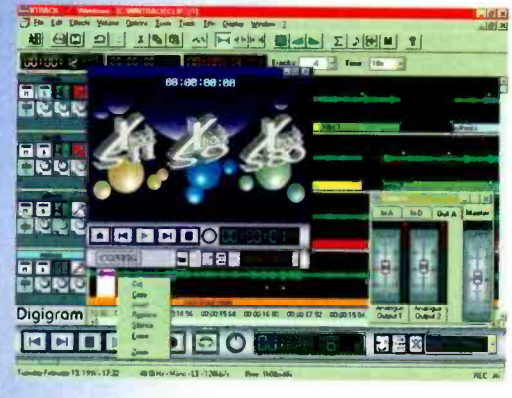

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PARALLELES

Amek REMBRANDT

Comfortably established as a key player in the upper-mid console market, and new desk from Amek will necessarily attract the attention of many modern studios. ZENON SCHOEPE paints a picture of Rembrandt

A WOMAN CALLED Angela has a lot to answer for. When fumbling around for a title to give their, then, newly created console, Amek Chairman Nick Franks suggested they temporarily name it after one of the members of staff. Unable to come out with anything better, the name stuck and the Angela went on to become one of the most popular and well respected medium-priced desks around —and helped establishing Amek as a major player.

Of course, those were less inspired days as far as product naming went and the Angela was followed by a series of alphanumeric titles before the company adopted its current policy of naming its consoles after other famous people.

Rembrandt is a completely new console and the first of a new Amek range. It will be followed by Galileo which uses the same EMC-friendly frame and is aimed more blatantly at post with LCRS panning and a more traditional in-line format. It was the advent of EU EMC legislation that stimulated the development of the new range but the Einstein console's use of two independent signal paths per module with separate EQ sections had proven popular with studios needing to handle large numbers of sources and so played a part in the Rembrandt architecture. The Rembrandt also answers some of the criticisms of earlier designs, offering more EQ, input reverse, more auxes and more LEDs. Rembrandt is not, however, a replacement for the Einstein—that is an honour that falls to the Angela II which has a similar architecture to the Rembrandt but with less EQ, fewer auxes and fewer central control functions.

Rembrandt will be followed by Galileo which uses the same EMC-friendly frame and is aimed more blatantly at post with LCRS panning and a more traditional in-line format

THE NEW CONSOLE comes in 40-channel or 56-channel frame sizes and can carry an integral or remote patchbay. Standard features include SuperTrue automation of long and short faders, mutes and four aux mutes as standard, Recall of all nonautomated controls, machine control (which operates from the function keys on the keyboard), and Virtual Dynamics. There is one dynamics module per strip and this can be assigned to the ☞

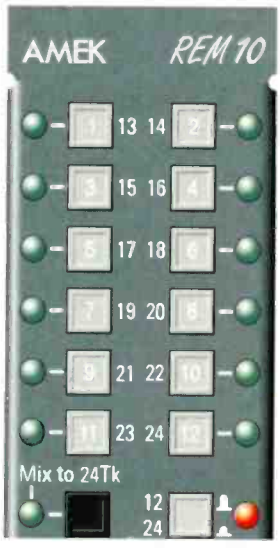
long or short fader signals with either being side-chain sourced from the other or keyed externally.

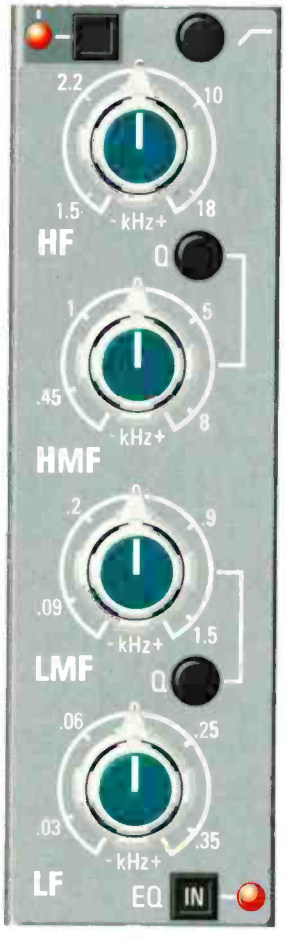
Ten VCA subgroups can be created within SuperTrue and any fader can be allocated master status as there are no dumb VCA master faders.

On the pure desk features front Rembrandt has two fully-featured inputs —the channel and monitor paths—per strip each with identical 4-band EQ and access to 24 buses and the main stereo mix. A total of 16 auxes are accessed from eight sends configured as four dual-concentric pots and the switching to the eight upper auxes is performed from within the computer software. A pair of aux sends can also be routed to the 24 buses for those frequent times when 16 is just not enough.

It's worth pointing out at this point that while the strip looks busy there are hidden functions, and some of the setting is performed from the computer software. For example, strips have a direct out function but activating it is done from the screen there is no physical switch. Similarly with metering—(off channel or the monitor path) and the entire assignment of the Virtual Dynamics—where it sits and how it is triggered. Dynamics LEDs give a visual indication of the selected choice and the section's activity be it gating or compression.

The desk will also run Amek's new Visual Effects package which allows remote control via MIDI of popular outboard gear





Above: REM 10 input module showing detail of routeing arrangements —12 switches give access to Rembrandt's 24 buses. Note LED indicators. Right: EQ on both channel and monitor paths is identical, featuring four bands with switched Q on the two mid bands

with visual on-screen representation of front panels and real-time control of parameters complete with saving routines.

Rembrandt uses the original Angela discrete microphone preamp and a stereo module is planned but EQ is more comprehensive than that on the Angela in being 4-band fully swept with switched Q on the two mids which overlap rather than overlay.

An interesting twist is the use of two Solo systems with PFL, AFL, and solo in place—one live system operated entirely from the desk surface complete with a panic Solo Defeat switch and a fully automated equivalent within SuperTrue which allows solo data to be written as part of a mix on the automation mode Select switches. Both systems allow for user programmable switch responses such as latching and cancelling.

REMBRANDT ANSWERS some of the criticisms levelled at Hendrix and Einstein—both are great boards but lacking in visual feedback. Input reverse is a welcome addition. Virtual Dynamics remains a very handy and fine sounding means of dynamics processing that is certainly quicker than patching in outboard.

I don't consider the software switching of some of the strip functions to be a great disadvantage—presumably it can be attributed to the attainment of the cost-vs-features equation—because the majority of them are the sort of things you wouldn't

be changing that frequently. However, 12 switch access to 24 buses will inevitably catch you out.

The centre section doesn't look terribly busy but it covers most eventualities in terms of monitoring sources and handling and routing talkback. It might have been more convenient to have the automated stereo master fader down in the large fader panel of the desk rather than tucked away above the keyboard.

SuperTrue doesn't need much introduction as it's been around for some time and it's now rock solid having made a quantum leap when Amek ditched the original Atari platform in favour of fast PCs. However, there are still minor irritations such as the fact that long and short fader VCA displays are tagged onto each other in one continuous row with not a lot of differentiation between where on ends and the other starts.

Less trivial is the mix-saving routine as SuperTrue will save a mix every time code stops unless you tell it not to any time before it hears code again. Admittedly this is much better than having to make the decision while the tape is still running but listen back to what you've just done and you've effectively stored it. The solution is obviously to store before attempting anything ambitious but it would be so better if SuperTrue could accommodate two mixes in RAM at the same time—the previous and the current—so you could compare and reserve judgment. Given that this automation package has improved so dramatically and has continued to evolve it is strange that this situation still persists. I still like it though.

Above: Close-up of channel strip showing Virtual Dynamics LED indicators, channel automation mode selector switch, and monitor gain pot.
Right: Aux sends featuring pre-post switching and access to 16 aux buses from eight sends

REMBRANDT OFFERS good all-round stout performance with terrific Amek EQ and plenty of flexibility and resources on a desk that is extremely straightforward in presentation and operation.

The fact that you have to get involved in the computer side of the console as a matter of course, because there are features on the board that you can only get to from the computer, doesn't have to be a limitation. It's just an operating principle.

There is no getting away from the fact that Rembrandt is remarkable value for money but then it does come from the manufacturer that redefined the term in this price band of the market. There is an awful lot of desk here. It outstrips Amek's previous generation of products quite conclusively.

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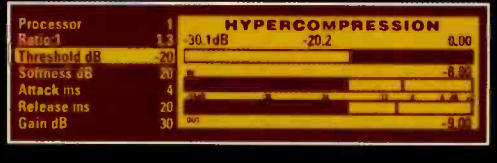

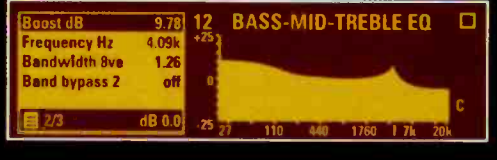

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E-mu DARWIN 4001

Moving from MIDI samplers into nonlinear recorders is a familiar strategy to the likes of Akai, Roland and Korg. **ZENON SCHOEPE** evaluates the latest system to share this heritage—E-mu's Darwin

E-MU'S VENTURE into the world of nonlinear recorder-editors is not unexpected. As one of the major players in the sampler game, the company has been part of the way there with much of its equipment already—as was Akai before it took the plunge. As a result, Darwin tells us a lot about how E-mu views the market and also reinforces a trend that has been taking form from other manufacturers with an MI background who have all adopted hardware-based implementations of hard-disk recording.

The commotion surrounding Darwin is that the 4001, 1Gb-loaded box (being discussed here) and the driveless 4000 were joined recently by the 4002 which had the distinction of being the first system to employ the considerable promise of Iomega's 1Gb removable Jaz disk drive. The Jaz equates to extremely affordable removable media and adds an independence to the Darwin at this level that cannot as yet be matched by anyone else.

All Darwin variants are 8-track capable and come with four balanced inputs expandable to eight with an input expansion card, and eight balanced outputs all on standard jacks. There is also an SPDIF digital I-O, a SCSI port for linking up extra hard drives and MIDI ports for MMC although true synchronisation is restricted to master-only operation via MTC. There is no direct SMPTE input.

In line with other recorders of this nature, Darwin also has a rudimentary level and pan 8:2 mixer for bouncing signals internally and for creating a separate headphones mix on the front-panel headphones output with level control. Sampling frequencies are 44.1kHz or 48kHz.

The front panel presents eight bar-graph vu meters with peak hold, eight track-arming buttons, a full set of traditional transport keys (including zero return) and cursor keys for moving around the largish LCD. The LCD works in conjunction with six soft keys which correspond to different on-screen functions depending on the menu you're in plus an EXIT key for getting out. There's also a cluster of keys for alphanumeric input, increment-decrement, and STORE and GOTO buttons for the units 40 locate points. Audio is scrubbed on a jog-shuttle wheel.

> Tape transport analogies go out the window when you discover that you can't throw Darwin into record from the track-ready switches

Other features include auto punch in and out with rehearse and auditioning of an edit before committing it. It is worth mentioning that this machine has up to 16 layers of Undo and Redo (just how many can be programmed by the user) which unusually is actually easy to keep track of as you're given a list of the editing operations you've performed.

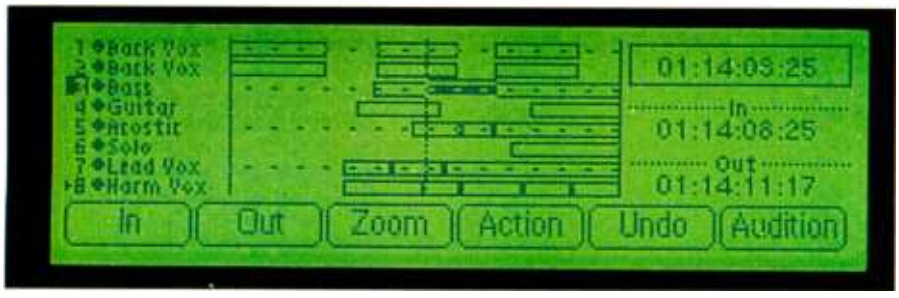
Darwin is a playlist-based system and organises material into Projects—you can have as many of these as you have disk space—and Projects are subdivided into Versions. Versions are different edits or combinations of the Project and are independent and do not effect the original recording.

ROUTEING INPUTS to tracks, and tracks to outputs is performed on a clear matrix-type screen and it's here that you can also control digital internal bounces via the mixer. E-mu should be applauded for going for the tape-machine-transport-style approach, proving again that if a manufacturer has a mind and heart they can offer this level of user friendliness for the price of a hard-disk editor and computer equivalent. However, tape transport analogies go out the window when you discover that you can't throw Darwin into record from the track-ready switches.

The editing process is governed by the setting of In and Out points to which you want to apply copy, cut, insert, replace, erase, extend and move operations. All work well particularly when you adjust the crossfade time to suit the action and material. There's a scrolling bar representation of the audio tracks on screen which can be zoomed in and out which is better than nothing but not a patch on a proper waveform display.

It is a strange fact that affordable hard-disk systems are subjected to appraisals ☞

From sampling to hard-disk recording: E-mu's Darwin



Detail of Darwin's LCD screen

of their sonic performance while the more expensive ones seem to be accepted blindly in this respect. It's strange because you have to consider what they're likely to be replacing. In the case of the Darwin, it's likely to be some form of narrow gauge analogue multitrack or a modular digital 8- track and it's well up to either of these. It sounds fine but then most of them do.

The audio scrub quality is superb and extremely tape like. On the inner finger dial you can hit really slow speed and get to the scraping the oxide off the tape stage. The outer ring speed is distinctly stepped in operation and I might have preferred a more gradual range. When using this shuttle wheel at maximum (2x) play speed Darwin will only let you hear the first four tracks. It's a bit of nuisance going into Scrub mode from Stop although you quickly develop a two-handed Stop and Play key routine to overcome it.

Operational flaws include the fact that changing modes has an irritating way of stopping playback. For example, going into Edit mode from the main menu kills the audio as indeed does accessing the mixer which sort of defeats the object. Unless you ingrain this fact into your operating practices you can very easily stop playback accidentally when fiddling around for something to do while running out to DAT. You really ought to be able to move around the system while the thing is playing.

Backup is to SCSI DDS DAT drive.

You can't enter locates on the fly, you have to be in Stop mode. When storing a point a screen asks under which number you want to save it and pressing a single digit from the keypad causes the screen to disappear so quickly that you don't actually see the desired digit registered. When you first get the unit you might waste time checking the Locate menu to see if it did actually happen. On the other hand, some of the other screen redraws are slow enough to make you wonder if you've hit the button. I find this disparity a little puzzling.

The metering overload LED is green like the rest of the bar graph and indicates its displeasure by winking regularly. It ought to be red because when running signal in as hot as possible you'll barely notice it.

Although it is early days for Darwin, providing MTC as master-only and not catering for SMPTE demonstrates poor understanding the state of the game. MMC, while a useful facility, is only a glorified remote control and the whole point of running a hard-disk system in a MIDI rig—or anywhere else for that matter—is that it should be able to chase as well as act as the master. Omitting SMPTE support means only those with suitably equipped software and sync boxes will be able to incorporate Darwin into their setups, and that's not the entire world yet. It sort of isolates Darwin from all but the screen-tanned MIDI buff and that's a shame.

Additionally, there is currently no contingency for multiple Darwin setups.

E-mu ought to be planning an ADAT or DA-88 digital interface if they want this bird to fly because it would mean you could hook this up to a Yamaha 02R desk

DESPITE THESE RESERVATIONS, on balance it is hard not to take a shine to Darwin. As with all things software driven, you'd kind of expect these things to be sorted out at some stage. Just how quickly and how elegantly should be a good indicator of how serious E-mu are about this particular business.

E-mu ought to be planning an ADAT or DA-88 digital interface if they want this bird to fly—not just because direct modular multitrack interconnection would be useful but because it would mean you could hook this up to a Yamaha 02R desk and that's an important consideration these days.

What is endearing is Darwin's absolute simplicity—how it presents and goes about what it does. The approach is substantially different to something like the Akai DR4 or DR8 which concentrate more on physical switches where the Darwin is based around a softkey-LCD bias that is honed into the specific set of tasks being attempted. Darwin sets you on a route that it knows you want to go down and it's extremely well thought out in this respect.

The results are effectively the same and comes down to a matter of personal preference. Hard controls ought to be best but by the same token Darwin's strength is its simplicity and it does the same. Data arrangement into Projects and Versions is great and I have no problem with it. Editing is fast with the welcome bonus of a depth of Undo.

Maybe I've been a bit hard on Darwin but it is still a very good first attempt. A revision or so down the road and it will be excellent.

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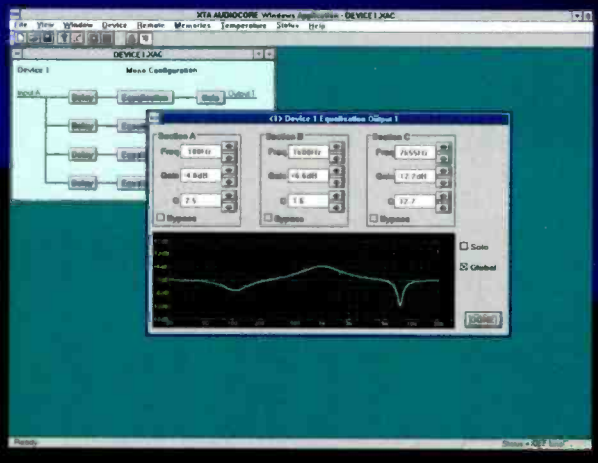

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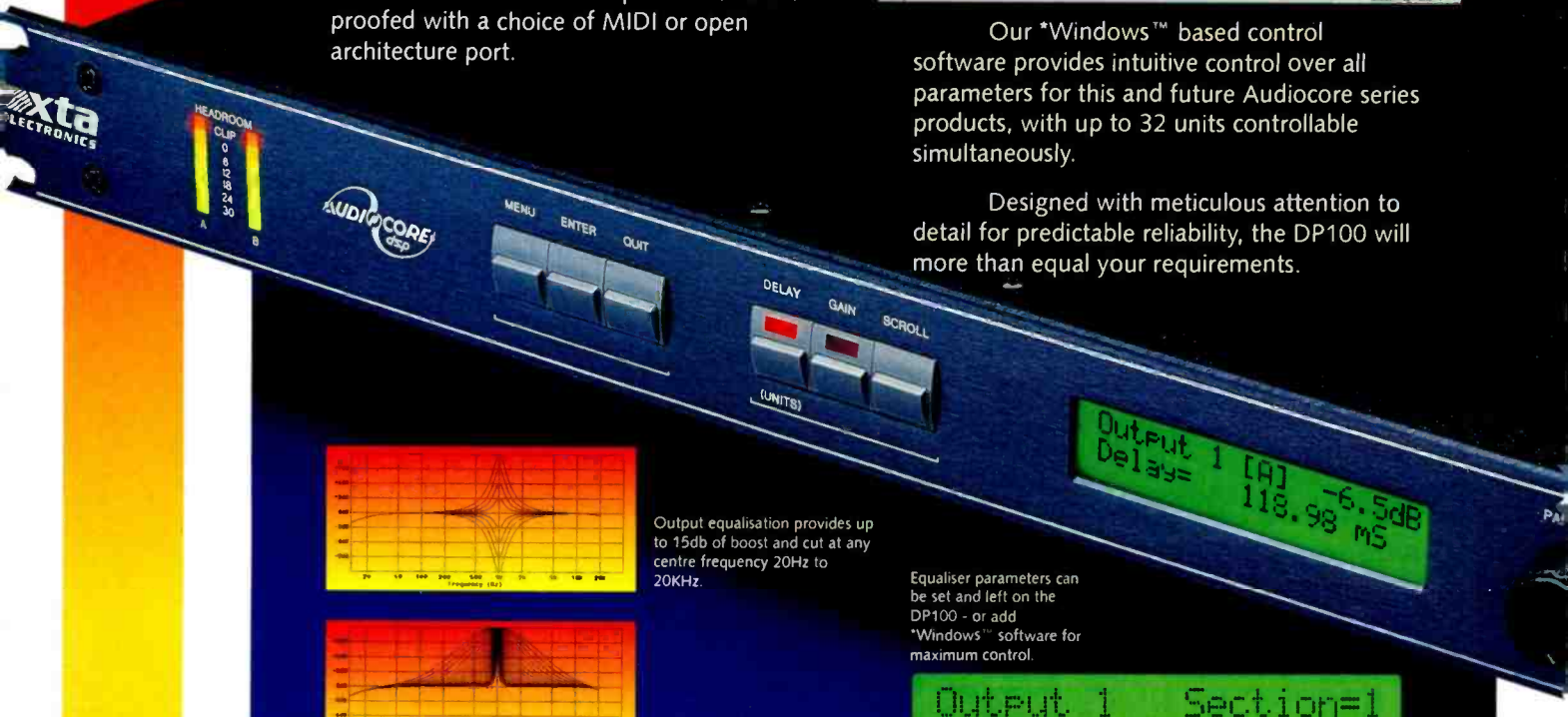
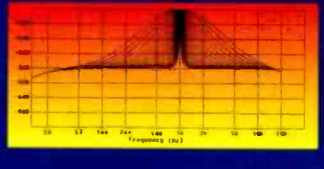


Output equalisation provides up to 15db of boost and cut at any centre frequency 20Hz to 20KHz.



A wide range of filter Q is provided for more flexibility.

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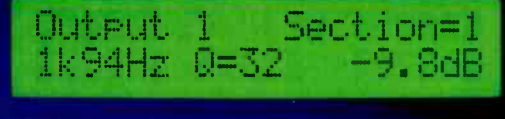

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TL Audio EQ-2

Regarded by many as the ideal audio application for valves, equalisation is also the forte of TL Audio's latest outboard box. DAVE FOISTER warms to the sound of a unit with a voice all of its own

A typical TL Audio design: the knobs are big enough to get hold of, and there is no overcrowding

ONE OF THE many pieces of TL Audio valve equipment to have passed through my hands was the original small stereo mixer, and foremost among its many merits was the sound of its EQ. This, I felt (and so, apparently, do many others), justified its use as a sidecar mixer for access to its equalisers alone. EQ has not featured as strongly as preamps and compressors in TL Audio's outboard range since, but that is surely to be put right by the appearance of this 2-channel, 4-band parametric EQ.

Being a TL Audio unit, the EQ-2's designers could not content themselves with simply building an equaliser. Instead, its phantom-powered microphone preamps give it no less than three inputs on each channel, the third being, as might be expected, an unbalanced front-panel jack for direct connection of instruments. This means it can easily be seen as a complete studio-to-recorder signal path, a prospect enhanced by the presence of TRS insert jacks on the rear panel for connection of compressors or other processors before the EQ.

The EQ itself is substantially more sophisticated than that on the EQ-1, TL Audio's previous valve equaliser. Each channel has four bands of fully-parametric EQ as against the fixed-Q switched-frequency bands on its predecessor, and although the bands are nominally labelled as LF, low mid, high mid and HF they actually comprise two pairs of identical wide-ranging bands. Thus the HF and HM bands both have the same frequency range (1kHz–20kHz), as do the LF and LM bands (30Hz–3kHz), all being wide enough to allow considerable overlap and consequent flexibility. Surprisingly, the outer bands can only be used in parametric mode, neither having a shelving option. Each band on each channel is individually switchable in and out of the path, and each band uses half an ECC83/12AX7 dual triode as its active component. Two further ECC83s form the output drivers, giving a total complement of 12 individual triode stages in the unit; if that doesn't produce a valve sound, nothing will.

Besides the main EQ bands, the two channels have separate individually-adjustable, sweepable high-pass and low-pass filters, again separately switchable in and out of circuit. These have a huge range, meeting in the middle at 1kHz with 12dB-per-octave slopes, making them far more of a powerful tool than most simple filters. The entire EQ chain can be bypassed on either channel, and separate controls for input and output gain allow the multiple valve stages to be driven as hard or as gently as required. Both controls have centre detents which give an accurate unity gain setting for +4dB line level signals; the output in turn is switchable to -10dB on the rear panel if required. A familiar TL Audio addition is a peak LED on each channel which glows more brightly as the valves are driven harder, reaching full brightness 6dB before hard clipping. All of this makes for a well-filled front panel, but while the knobs are big enough to get hold of nothing is too crowded for easy access. TL Audio's usual mesh ventilation grille sits above it all.

> It is the less definable advantages of the valves that lend this equaliser its musical quality

THE REAR PANEL, too, is more densely populated than might be expected, as the microphone inputs have their own separate sockets, unbalanced jacks duplicate the balanced line inputs and outputs, and the aforementioned inserts require another two. This flexible arrangement allows any setup to be accommodated without bodging, and mic-line inputs to be left permanently connected and selected on the front.

An unusual feature on an equaliser of this kind is the facility to gang the two channels together with a single switch, so that the lower (Channel 1) controls take charge of both channels for straightforward stereo setting. The matching accuracy seems very good, and in any event is bound to be better than lining up two sets of knobs by eye, or even by ear.

The all-important consideration of sound demonstrates the EQ-2 to be every bit as flexible and musical as I remember the EQ of the mixer being. Each band has up to 15dB of boost and cut available, coupled with a Q variable from 0.5 to 5; these sensible ranges, together with the broad sweep of centre frequencies, deal comfortably with most EQ requirements from the subtle tweak to the special effect. Noise performance is good, and the detail provided by the valve circuitry means that the EQ only ever seems to enhance. It is, of course, possible to drive it hard and introduce the expected valve distortion if needed, but this must be done deliberately; otherwise, it is the less readily definable advantages of the valves that lend this equaliser its distinctive musical quality.

EQ has often been regarded as the valve's strong suit, comprising a large proportion of the most treasured vintage equipment. It would be all too easy to jump on to that bandwagon, as some have done with microphones, and sell mediocre EQ to the gullible simply because it glows. This is decidedly not what TL Audio has done; the EQ-2 has a voice of its own which deserves to be heard.

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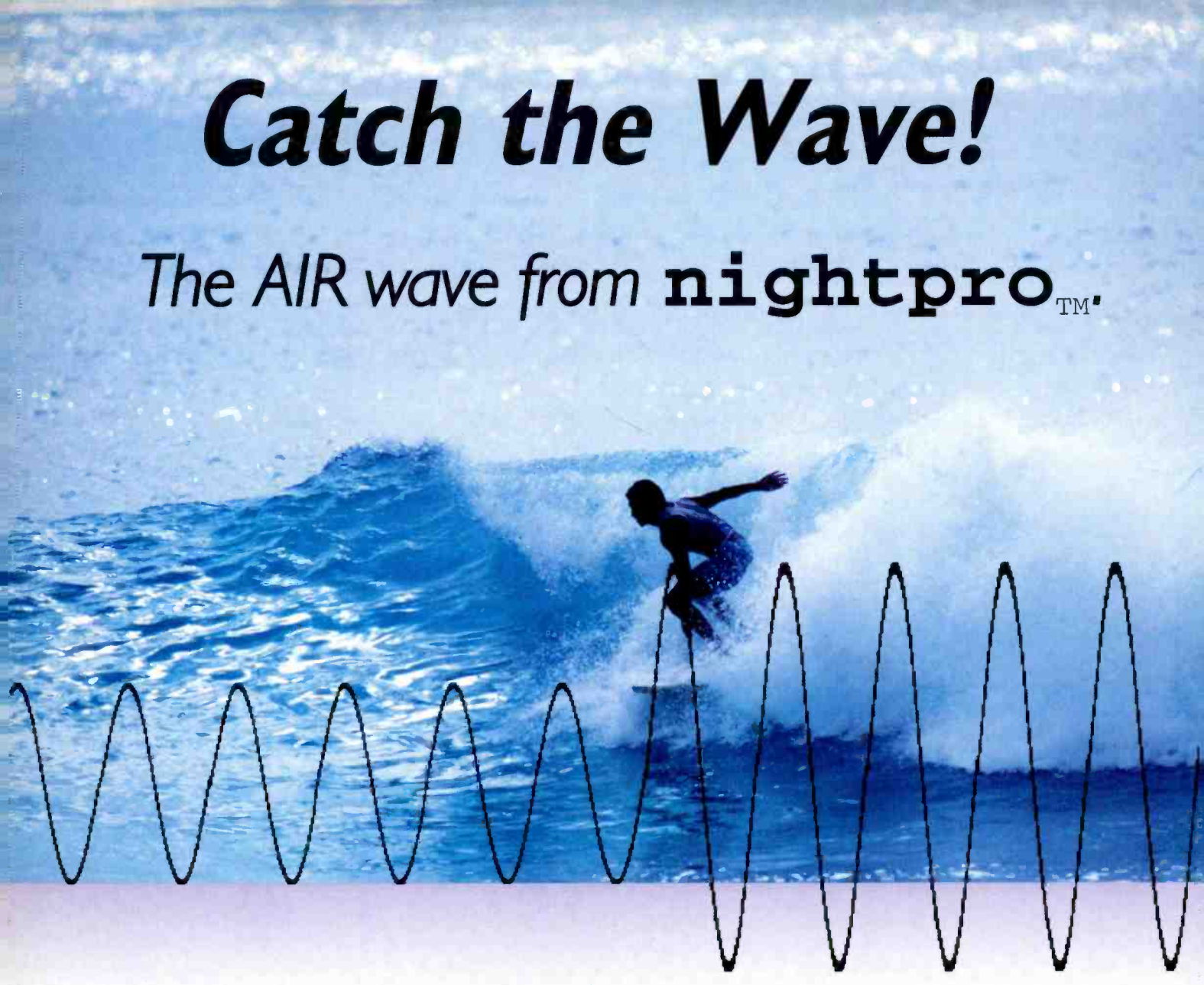

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tc electronic M2000

Moving down market from the mighty M5000, tc electronic has the project studio in its sights with its M2000, 20-bit, dual multieffects unit. GEORGE SHILLING tests its targeting systems

THE M2000 is a 20-bit, dual multieffects unit aimed at the project sector of the studio market. As such it is in direct competition with units such as Lexicon's PCM80. Previous units from this Danish company—such as the 2290 delay and M5000 audio mainframe—have proven popular with high-end studios but have been criticised for their somewhat unfriendly operation. This new unit is clearly designed to overcome these criticisms, as you might tell from the prominence of the computing term 'Wizard' on the packaging (a feature we'll investigate later).

The cheerful, clearly laid-out manual begins by acknowledging that most people don't read manuals. And if you've got that far, then it's a strategy that's likely to hold your attention. The 'ease of use' marketing concept of Windows 95 has been received and understood: 'Plug and play' says tc (I'm not sure what Bill Gates' lawyers might say). For those determined to ignore the manual there is a Help section built into the unit itself that displays a summary of functions—a great idea that saves time spent hunting for the manual when you are stuck. For freelancers, this means that you can pretend you are adjusting the settings, when really you are trying to find out how to.

The M2000 has two effects 'engines', which can be configured in six different routeing options. With these you can combine two independent processes in series or parallel to give two separate effects units with mono inputs and a shared stereo output; a 'true stereo' parallel mode which links the edit pages of the two engines; dual mono; and Preset Glide which crossfades from one engine to the other. Each engine comes with 128 programs, and there are also 128 'combi' programs combining different effects from the two engines.

The front panel has a large back lit LCD; a PCMCIA card slot for storing and loading programs; six logically arranged rows each of four John-Major-grey coloured switches, each with tiny legending beside it, and infinitesimally small legending in blue underneath describing its 'shifted' function; a TEMPO TAP button (sets delay-modulation times between 20bpm and 200bpm depending on tap rate); and a smoothly detented ADJUST knob. The back panel includes stereo balanced XLR inputs and outputs; AES-EBU and SPDIF digital in and out; MIDI In, Thru and Out, (disappointingly only program change implemented: no MIDI clock à la PCM80); and a jack socket for foot pedal, which disappointingly only operates as a bypass switch (no fast-slow switch for Leslie speaker simulation program for example).

Setting up is a piece of Danish. Press the I-O button to select inputs and outputs, mix or 100% effects, digital clock rate and dithering. You can mix digital and analogue inputs, and pressing the LEVELS button allows selection of -10dB or +4dB operation, and trim the levels. Unfortunately, the level trims have a very narrow range (in +4 mode you can only turn the inputs between -6dB and +16dB). This is, to my mind, unforgivable, as there is no level knob on the front—in times of trouble you might need to turn the input right down on the unit, perhaps to locate a source of unwanted noise.

There is a wide range of algorithms: Reverb, (Hall, Room, Plate, Ambience or Gated, all of which can be switched to Expert mode which has a full set of parameters covering all reverb types); Chorus; Flanger; Delay; Phaser; Multi Pitch-Shift (six independent harmonisers); EQ; Tremolo; Stereo (Spatial expansion and Hi-Cut); Dynamics (Compressor, Limiter, Gate and De-Esser).

THE WIZARD is a feature to help you locate a program appropriate to your needs. You select the type of effect you want, type of instrument and intensity of effect and the M2000 presents you with a selection of suitable programs for quick comparison. This is quite a useful facility when looking for similar programs, although you might not agree with the Wizard's choice. Four SNAPSHOT buttons helpfully store edited settings for instant recall. Internal RAM gives you a plentiful 128 store locations.

A help section is built into the unit— a great idea allowing you to bluff your way out of any problems

For me, the best of the M2000's unusual functions is Dynamic Morphing. This takes full advantage of the two effects engines and morphs (executes an 'intelligent' crossfade) between the two selected effects at a selected input level threshold. This gives you the ability, for example, to send quiet sounds to a long wet reverb and loud stabs to an in-yer-face phaser with a smooth join. Unfortunately, most presets don't have anything as exciting as my example: generally, the programs are somewhat bland, lacking some of the inventiveness of, say Eventide's H3000D/SE's mod factory programs or Lexicon's PCM80's more wacky offerings. Although most presets are useful, I found myself twiddling about to get really juicy and inspiring stuff. This is simply unimaginative software programming. Just my opinion of course—others may disagree—and I am sure that before too long a PCMCIA will appear on the market with some better programs.

I cannot fault the M2000 on a technical level: the sound quality is fantastic, the reverbs smooth, and the noise levels very low. Digital compression and limiting is a long way off a valve Fairchild 660, but it is not unusable, and the pitch-changer is remarkably glitch-free.

This is a great piece of kit and great value if priced similarly to the Lexicon PCM80—far more user-friendly too. I'll be looking forward to v2 with MIDI Clock tempo setting, patchable footswitch parameter and some more exciting and juicy presets. Just a hint, tc... S

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Sonic Solutions MEDIANET

The digital revolution is increasingly making interfacing and networking the essential points of equipment development. ZENON SCHOEPE reports on the form, function and philosophy behind Sonic Solutions' Medianet

THE UNNERVING THING about Sonic Solutions as a company—if you look at what it has done and is doing—is that it has a particularly tolerant attitude towards other technologies and manufacturers. Compared to some of the Bible-bashing DAW manufacturers, Sonic's approach is relatively accommodating and prepared to go with the flow.

In no way is this accommodation better illustrated than in the announcement at the IBC 95 convention that it was tying up its technology with picture specialist Discreet Logic. Under the OEM partnership Sonic will supply its digital audio technology for use with Discreet's systems for special visual effects, editing and postproduction.

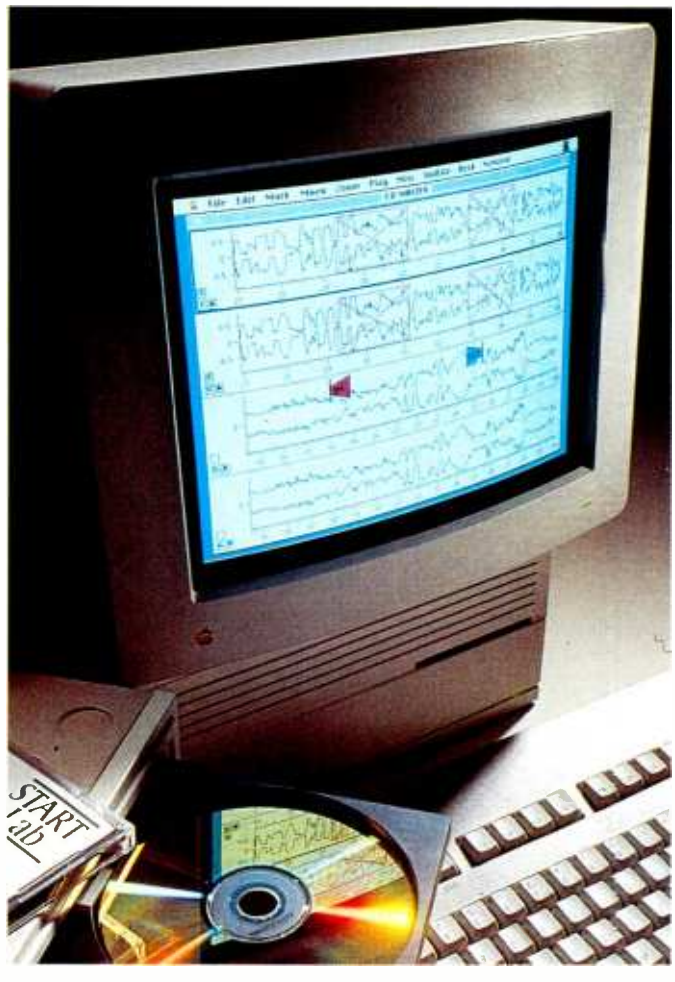
Discreet is employing Sonic workstation technology into its Stream, River and Rain systems for on-line sound design. By integrating sound effects, dialogue and music with visual elements it creates a new way of working in which audio and images are manipulated simultaneously. This sort of co-op is still a rare occurrence in these supposedly liberal times but it's interesting to note also that Opcode demonstrated StudioVision Pro running on Sonic engine boards at the NAMM show. Kirk Paulson, Vice President of Marketing at Sonic Solutions, sees it all as a natural extension of what they do.

'We believe there is a segment of the market that would like to use and have access to StudioVision Pro on what they perceive as a more prestigious platform,' he says. 'It's conceivable that the users could shut down the Opcode programme after they've finished and then boot up a Sonics application and master a CD. It allows us to increase our market share and undoubtedly there will be several other companies who will make use of this as well.'

With the release of the USP, Sonics is now effectively on its third generation of processing power with more RAM and a faster SCSI bus among other things. Most will also be glad that the seemingly enormous and intertangled product range has been clarified to give prospective buyers a better chance of identifying what they need—mix-and-match to the precise application, choosing from categories of software, engines and I-O options. However, Medianet—Sonic's networking system—remains one of its major strengths and attention grabbers. Abbey Road Studios uses it to share the resources of its NoNoise and CD mastering systems and the latest Medianet software adds the ability to record to remote volumes and to restore and archive across network.

The interesting point about Medianet is that it has not been restricted purely to audio and in typical Sonic fashion spreads across applications and disciplines including such non-audio, high-power exercises as desktop publishing.

'What we've done with our network technology is to address the bottlenecks that you find when you're moving around large bandwidth data,' explains Paulson. 'The bottlenecks appear at the hard disk because the file system deals with small data blocks and if it becomes very fragmented, then the disk has to work overtime. The other bottleneck is on the bus itself so we've added SCSI controllers so that there is direct connection to the hard drive itself. Most important is the host CPU; in most network installations you're sharing CPU cycle between what you're doing in terms of editing manipulation and what's happening in network traffic. We've put a CPU on the node to act in essence as a server on the node. This handles all network traffic sending data and receiving without taxing the host computer at all.'



Paulson believes that users are beginning to realise that the ability to integrate a network and to open it up so that it's compatible with video and other applications was there from the word go.

'The ultimate goal for the network is to allow you to access DSP power in a facility with any number of audio processing cards and assign them accordingly,' he continues. 'We're fairly close to that already, the integration of DSP power in a network is the most complex but we're pretty confident that we'll be the first to market with it.'

NETWORKING necessarily involves discussion of network type—Sonics opts for FDDI at present for what it considers to be very good reasons.

'Some of our competitors have suggested that because our network is only capable of 100Mbits, which is approximately 100 channels of audio, that is not enough for a really large facility,' Paulson reveals. 'So they suggest everyone should take a wait-and-see attitude until the next network technology comes along. That's like saying let's not develop a DAW until we can have 48 channels of playback. It's absurd —technology moves along. Medianet is an architecture that can be ported over to other networking technologies.

'The networking technology we use today happens to be FDDI but we can just as easily port it over to fast ethernet or to ATM,' he continues. 'The requirements are that the network we use is very fast —100Mbits or more—and that it permits bandwidth reservation.'

Medianet is compatible with standard Macintosh applications and the file systems used by Sonics are compatible with standard Mac applications and ☞

products. Paulson adds that ATM, while undoubtedly a focus of interest, is currently costly for the performance it gives.

'People tend to read the first bit of information that crosses their desk and they see FDDI—100Mbits, ATM in theory 1Gb—and they go for ATM,' claims Paulson. 'If they researched it further they would realise that ATM today is maxing out at 30Mbits–40Mbits which is, in fact, lower performance than FDDI. The reason for this is the way ATM operates; it operates in very small packets. If you take a block of audio data that you want to send somewhere else, based on the ATM protocols that would require that you take that large file and fragment it and then shoot it out over the network. At the other end you have to defragment it into the original file. The process of fragmenting and defragmenting requires a lot of horsepower. In laboratory environments where you're running SGI machines and huge workstations it's very easy to say "look at the throughput" but getting that sort of power in silicon down to the board level is quite costly and not yet there.

'We're backing ATM as a backbone,' he continues. 'Say we have a FDDI token ring set up between a group of workstations and then a second group of workstations in another zone, like sound effects and sound design, and we want to communicate shared data between them. I would use ATM as the backbone between them.

'It's very difficult to run FDDI cable from here to another facility across town because it would cost a fortune whereas ATM clearly is being set as a standard to transfer data between different facilities,' he concludes.

Although Sonic Solutions do not have an ATM board solution available as yet the company has added to Medianet what it calls Internet Protocol Encapsulation. This provides a means of taking data onto the network and at some point encapsulating it into the small data packets for ATM. According to Paulson, this is critical to and yet absent from some ATM network implementations. He describes these as proprietary implementations of ATM.

'What we've done with our network technology is to address the bottlenecks that you find when you're moving around large bandwidth data. The bottlenecks appear at the hard disk because the file system deals with small data blocks and if it becomes very fragmented, then the disk has to work overtime' – Kirk Paulson

'Some companies are saying we want ATM to the desktop,' Paulson explains. 'They want to take an ATM board and put it in this Macintosh or SGI and that's where we disagree. We would suggest that FDDI or even fast ethernet or some of the other networking technologies that are coming about are better suited to the desktop because of cost, performance and the fact you can deal with large data files. We'd prefer to use ATM between workgroups.'

Paulson is adamant that Sonic Solutions is not alone in this belief, and he draws attention to the fact that the company has been practicing what it preaches for around five years. But then networking has always been pivotal to its philosophy.

'Certain systems are best used to get data on to the network and off the network,' he explains. 'This is the most efficient use of the network today—to be able to declare one room the Transfer Room and another the Editing Suite where nothing is done but editing.'

By the words 'certain systems' Paulson is acknowledging the worth of other DAW types.

'We have to,' he acknowledges, 'otherwise we're suggesting that we have the answer to all the different production needs and that's far from the case. The most recent example of this sort of collaboration is the association between Discreet Logic and Sonic Solutions. Discreet's Flame is one of the premiere computer graphics image compositing solutions for the Silicon Graphics environment and our audio substation is used with that system.'

Point taken.

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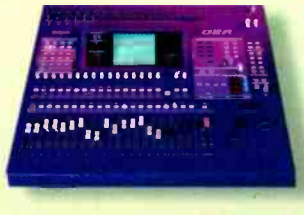

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Milab EMBLA

The introduction of Milab's Embla microphone may well see the Swedish manufacturer emerge from the sidelines to become a significant player in the professional condenser microphone field writes DAVE FOISTER

I SUSPECT THAT for most of us the Swedish microphone manufacturer Milab will be more familiar from adverts than from direct contact. Milab microphones are not generally to be found in many people's armoury, which seems strange, as the very small product range has always looked interestingly different and impressive in specification. Particularly intriguing is the VIP50, surely one of the oddest-looking microphones ever produced and boasting an unusual number of switchable adjustments.

The most recent addition to the range is the Embla, a substantial side-firing condenser clearly related (internally at any rate) to the VIP50 and also with its sights set on professional studio use. Like the VIP50, its remarkable dual diaphragm capsule offers slightly more than the expected choice of polar patterns, with the usual omni, figure-of-eight, cardioid and hypercardioid augmented by a wide cardioid or subcardioid setting. The five patterns are selected by a large rotary switch on the front, suitably recessed against accidental movement but not so much as to become fiddly. This is the only control the microphone possesses—there is no pad and no filter provision.

The cylindrical body can be attached to a stand using a conventional stand mount, and an elastic suspension is also available. This is thoughtfully designed, with a sprung clamp around the body and a ball-and-socket joint for angle adjustment. This form of swivel removes the constraints placed on the aiming of many cat's-cradle mounted microphones and also makes the whole assembly slimmer and less obtrusive. The microphone body itself is made of solid brass, as the weight attests, and the matt black chrome finish with its white engraving is immaculate.

The grille surrounding the capsule itself is almost diaphanous; it does little to shield the diaphragms from blasts, but it does allow a clear view of the Embla's most unusual feature - the capsule itself. The design of the capsule assembly flies in the face of convention in two respects, one immediately apparent and one less so. The first and obvious one is its shape, which is rectangular where everybody else's is round. The capsule is Milab's 2700, first developed in the fifties and sixties and in continuous production, with minor improvements, ever since. It appears in Milab's top three models —the VIP50, the DC96B and the Embla —and like the circular capsule used in the VM44 it is hand-made in Milab's own Swedish manufacturing plant. Precise reasons for the rectangular shape were not forthcoming, but clearly, as Milab has pointed out, the resonance phenomena of the diaphragms are bound to be significantly different from those of a circular membrane. The ratio of length to width is, of course, chosen so that resonances in the two dimensions across the diaphragm will occur at different frequencies, and Milab also point out that the acoustic properties of the shape —its effect on the sound field it is placed in—will differ from a conventional circular capsule of the same surface area.

The other deviation from the norm is the material chosen for the membranes, which are made of aluminium rather than the usual metal-coated polyester. Again, this will have a significant effect on the stiffness, the mass and other important acoustic properties, and Milab also points to electrical advantages.

The preamp electronics are a hybrid of surface mount and conventional design techniques, with matched JFETs for the head stage and a transformer balanced output.

> The resonance phenomena of the diaphragms are bound to be significantly different from those of a circular membrane

Whatever the ins and outs of the capsule design's departures from normality, the published curves for frequency response and polar pattern accuracy are among the best you will find. Each microphone comes with a plot of its own personal frequency response in cardioid mode, and the one with the review sample differed little from the published curve. The omni response —inevitably the best—is almost completely flat, with a 2dB lift at 12-13kHz, and the cardioid and wide cardioid responses are little different. Hyper cardioid and figure-of-eight deviate only in having a slight rise in the upper mid, and in all cases the off-axis response is remarkably flat, a fact borne out in use by the lack of coloration of the spill.

In fact lack of coloration of any kind and from any direction characterises the Embla. This is a really neutral, detailed microphone, with convincing depth at one end and completely open clarity at the other. It can handle any of the jobs normally dealt with by one of our more familiar side-fire condensers, and I particularly enjoyed its vocal sound, which was utterly natural and unstrained, with remarkably little tendency to pop—even without a shield and with its insubstantial grille—unless approached too close.

In fact I enjoyed the Embla every time I used it. I have come late to Milab microphones, which is a shame; if this is typical of what Milab can do, they deserve to be gracing the best microphone cupboards in the world alongside the established stars.

Lack of coloration of any kind and from any direction characterise the Embla microphone

CONTACTS

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Focusrite RED 4

Applying the standards of quality and insight that have helped make Focusrite internationally popular, the company has produced a comprehensive level-matching input unit to reconcile pro and consumer sources. DAVE FOISTER reports

IT DOESN'T MATTER how grand the facility is, somewhere or other it uses some domestic equipment—the less grand, the more of it there will be. Cassette decks, CD players, video recorders, tuners, turntables, the odd low-end DAT machine, and often more obscure things like old F1s—all sporting phono output sockets delivering consumer levels around -10dBu unbalanced, and all needing to be hooked up to professional equipment. The need is obvious and simple: a high-quality gain stage to bring all these signals up to line level and balance them so that they can be connected directly to recorder inputs, desk tape returns and channel line inputs without either constant realignment or irritating level jumps on the monitors.

The requirement is, in fact, simpler than it was some years back because domestic kit is far more consistent about its -10dBu than it used to be, yet very few manufacturers have addressed the problem. Other than simple 2-channel preamps, build-it-yourself bodge boxes and a few other oddities, Totalsystems' Studio Preamplifiers and the Omniphonics models are the only dedicated units that spring to mind. Except, of course, the Focusrite Red 4, an almost inevitable addition to the familiar eye-catching range of high-end analogue signal processors.

The Red 4 makes no pretence about doing anything other than the simple job outlined above, with a few frills for delicate adjustments along the way. EQ, historically Focusrite's forte, is not included—although unusually elaborate filter controls are fitted. Seven stereo inputs are provided, each individually switchable between consumer and professional level sensitivities, and selectable by means of a large rotary switch. Sensitivity settings are on a set of DIP switches on the rear panel, and, surprisingly, all the associated input connectors are XLRs, although Focusrite can supply ready-made suitably terminated cables. There is no dedicated turntable input, neither is there an optional phono preamp of the type so beloved of the hi-fi fraternity.

Red 4 offers 7 stereo inputs with level matching to bridge the gap between consumer and pro sources

FOCUSRITE NORMALLY fights shy of any displays more elaborate than a vu meter, a couple of illuminated switches and some discreet panel printing, but on the Red 4 the rule is spectacularly broken. The selected source is shown on an alphanumeric red LED window, and another set of back-panel DIP switches determines which of the factory sets of labels is to be displayed. These range from simple sets of numbers (in a choice of languages, naturally) to a couple of sets intended for the kind of choices a domestic user will have available, and also include some selections for the professional user, mentioning words like cues, monitor and cans. The last, being Focusrite, says 'Buy more Focusrite Red modules dude!'

Level matching is fixed, with no trim adjustment available for individual inputs, and the difference in gain between the two settings is, of course, about 12.8dB—the difference between -10dBu and the standard voltage equivalent of +4dBm. As expected, this gain makeup is very precise indeed, and in case this is not matched by the precision of the source, a balance control (with a bypass switch) adjusts for interchannel differences while a P&G rotary control acts as an overall output fader with a small amount of additional gain available if required. Levels pre the output control are shown on a pair of the now familiar round Focusrite vu meters.

Filtering comprises a switched 50Hz high-pass filter and a switched variable frequency low-pass, with turnover frequencies selectable from 8.2kHz to 22kHz, all calibrated to the expected Focusrite accuracy. Two further controls allow for the use of the Red 4 as a straightforward preamp to act as a front end for a power amp (strangely, the Red 5 is suggested as an ideal partner). A DIM switch reduces the level by about 12dB while a MUTE switch does exactly what it says. None of the switching produces clicks, particularly as the input selector switch operates muting relays while a change is made. These few details suggest that the Red 4-5 combination could easily cross over into the consumer market as a very high pose-value hi-fi, and that is indeed what is happening. Initially, a few home audio enthusiasts bought Focusrite systems (including, occasionally, studio EQ modules to act as tone controls) until Focusrite decided that an all-out attack on the top-end hi-fi market was in order—an attack which has already begun and whose outcome we await with interest.

In the meantime, those of us in professional circles with consumer sources needing appropriate level matching have got little enough to choose from; Focusrite design quality ensures that the Red 4 will deliver every ounce (should that be gram?) of performance from those sources and provide a few highly-accurate corrective treatments as well. Cheap it isn't (wait for the Green equivalent, if there is one) but many of the sources it is intended to deal with, domestic as they are, are actually capable of very high quality themselves, and deserve better than to be let down by a cheap preamp jacking their levels up.

CONTACTS

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Night Technologies PREQ3

Already recognised for its 'alternative' approach to both audio and the audio business, Night Technologies is enjoying a wide acceptance of its Air band circuitry. ZENON SCHOEPE reports on a mic preamp with extra puff

NIGHT TECHNOLOGIES is a small American company based in Provo, Utah. Most of us recognise the company as the manufacturer of the not inexpensive, but quite remarkable, EQ3 equaliser. It is unusual (or 'unique' as Night Technologies prefer) in incorporating a so-called Air Band EQ. What is unusual (or unique as I also prefer) about Night Technologies is that they recently actually invited former British Prime Minister Margaret Thatcher to come to Provo to take part in its British Week. Anyone responsible enough to take their turn to get her out of the UK for a few days can't be all bad.

A refresher and recap on the EQ3 will be useful as there is some shared lineage with the PreQ3. The EQ3 is a dual-channel 6-band EQ with four extremely broad peaking bands, a reciprocal shelf and the aforementioned Air band centred at around 10kHz. This offers boost only and effectively adds high frequency shelving boost on top of whatever else has been done earlier in the chain.

The PreQ3 takes this Air band thing a little further in offering switchable Air band frequencies of 2.5kHz, 5kHz, 10kHz, 20kHz and 40kHz frequencies and a defeat option in addition to a continuously variable pot for maximum boost at these turnover frequencies of 12.5dB. Things are taken further still by the inclusion of a mic-preamp section and the PreQ3 is modular in that the front panel is drilled and legended up for four channels and can be partially loaded and expanded at a later date. Where things get really clever is that the incoming mic signal hits the Air band equaliser before it hits the unit's gain circuitry. What it reveals is that having equalisation before the gain stage allows mic characters to be altered in a way that is different to doing it after. Part of the reason they can get away with this is that the Air EQ is extremely gentle and smooth and the unit is remarkably quiet. Operation is via a GAIN pot working with switches for a 20dB pad and 20dB boost plus phantom power, phase reverse, a high-pass filter and a mic-line selector permitting the Air band to be applied to mix sources. Connectors are balanced XLRs and all the switches including the Air band's bypass have LEDs.

> Of course, the PreQ3 doesn't cure anything but generally passing a mic through the Air band only really improves matters

SO WHAT does the PreQ3 actually sound like? Well, it is curious that such a small change in the order of the circuitry should make any difference—but it does. Basically, the better the mic the more you can do with it. What the PreQ3 does, and its effect is extremely variable due to the gentle slope of the shelves, is to open up the sound of a microphone to make it brighter, give it more presence, or on the 20kHz and 40kHz settings to give very fine, ultra-high-end lift. The point is that because of the sort of slope involved, at maximum tweak on 40kHz you are still getting more than a handful of dB lift in the mid-teen frequencies—it's not profound but I'm still young enough to hear it.

The 20kHz setting is much more obvious and is probably the highest you'd actually go in practice without irritating your dog or drawing the interest of every love-happy bat in the neighbourhood to come and mess on your window ledge.

You can add broad presence that transforms the mid range to even a rudimentary condenser mic, and can even give the complexion of a condenser to a decent dynamic simply because it gives you more of what is there already. Microphones tend to sound bigger and fuller.

Of course, the PreQ3 doesn't cure anything—flaccid low end responses remain as such—but generally passing a mic through the Air band only really improves matters. The lower frequencies are particularly good on vocals for enhancing a voice's broad character and excellent on acoustic instruments where you can really wind it on and use traditional EQ afterwards as well. The higher settings open up detail in room mics and makes the room sound a little more special and the mic a little more expensive. If anything, I found the lack of any sort of LED indication—not even a power indicator—a little disconcerting when all the switches are deselected for line input operation.

The Air band is a fair sweetening tool for mix processing but if you want to get serious then you ought to be looking to tie it together with an EQ3 to benefit from the extra band control. An EQ3 with switchable frequency Air bands and mic preamps wouldn't be a bad idea.

In conclusion, the PreQ3 is a well-made unit with a good solid preamp stage and the undoubted bonus of some unusual EQ shaping. The effect is difficult to describe. You should try one.

Now widely accepted in the US, Night Technologies' Air band processing is ready for international recognition

CONTACTS

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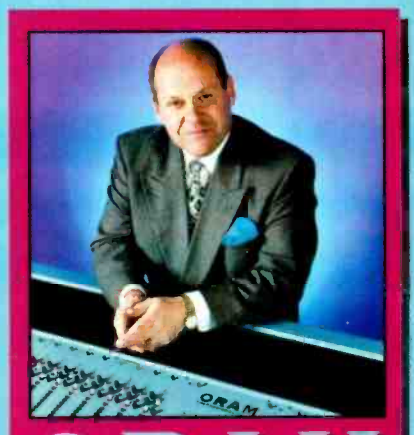
* Source: EQ magazine March 1996, "Console 96" round-up.



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AD Systems

Aimed at 8-bus mixers, the Drax automation package is VCA-based and comes with a choice of dedicated mounting kits for a wide variety of 8-bus consoles and employs existing faders and mutes. The system is time-code based and has a built-in reader, and an RS232 cable for direct connection to a PC running Windows 95.

Akai

Version 2.00 software for the DD1500 adds five DSP functions, an autoconforming package and SCSI-based backup options that support the new generation of 4mm and 8mm tape drives. DSP features are timestretch, pitch shift, varispeed, EQ and reverse. Timestretch and pitch shift ranges are -50% to +200% and employ preprogrammed algorithms that have been optimised for different types of audio signal. The autoconform features require no external PC for list editing and support most common EDLs which can be loaded straight in to the graphical display of the machine.

AKG

The WMS 300 is low-priced contender in the UHF radio-microphone market and offers ten different system configurations all based around the SR300 dual-channel UHF receiver and covering hand-held, beltpack and head-worn models.

The upgraded MicroMic II series features seven new models designed for miking a wide range of instruments including brass (C419), drums and percussion (C418) and acoustic stringed instruments (C411). Guitar cabs, accordions and pianos are covered by the C416. Also included in the range are specific microphones for lapel (C417) and head-set use (C420), plus a dedicated acoustic double-bass mic—the DB 1—which has an acoustic double-bass pick-up system with a piezoelectric capsule which is incorporated into a conventional wooden bass bridge. Four of these models, the C417, C416, C419 and C420 are available as part of the new radio-microphone range combined with the new PT300 body-pack transmitter.

Allen & Heath

The GL2000 live desk combines FOH and on-stage monitoring capabilities in a single package. Featuring four groups and six aux sends it is available in 12, 16 and 24-channel frame sizes, and also includes a stereo input section complete with mic preamps, two stereo effects returns on faders with EQ, individual pan control on all subgroups, input meters on every channel plus 4-band EQ with two mid sweeps.

The models are expandable via Allen & Heath's proprietary SYS-LINK system, allowing daisy-chaining. In FOH mode the GL2000 works as a straight 4-bus console, while in monitor mode it acts as a six mix console but the desk can also be configured for both simultaneously.

Alesis

The NanoVerb 18-bit digital effects processor is based on 16 preset effects algorithms taken from the MidiVerb and MicroVerb series, which have been combined with 18-bit digital convertors.

Amek

Amek's Digital Mixing System consists of four elements: the control surface with host Pentium computer, the 32-bit floating-point DSP engine, I-O racks and an automated crosspoint matrix. These are configured to provide the advantages of virtual and hardware solutions as the mixer can be reconfigured on a task-by-task basis to suit an application within the constraints of the available signal processing and the number of I-Os. Dynamic Resource Allocation is part of this principle and will eventually extend to shared configurations in which the DSP may be divided between two or more control surfaces. Automation is based on the SuperMove moving fader system and combines full dynamic automation with snapshots and on and off-line editing with graphic displays and libraries of popular settings.

Amptec

The Stone-D001 digital desk has a traditional desk user-interface with mono and stereo input and group and master output modules working on a TDM-style bus that can process 64 stereo input channels, four aux sends, four stereo groups, one stereo master and monitor outputs. Every input has access to a high-quality mic preamp, 24-bit A–D, 24-bit digital input, 32-point floating-point DSP and 4-band EQ. All outputs have Ultimate digital limiters and analogue outputs exit via 20-bit D–As. The desk is targeted at radio and TV on-air, postproduction, mastering and sound recording.

AMS Neve

The long-serving Logic 1 post system enjoys the first European showing of its v1.7 software, including feet-frame timing, automated insert-point switching, global convertor overload detection and time-stamping of setup and automation

dBx 1034 2-channel crossover-limiter incorporating PeakStopPlus limiting

Audio Follow

The Changeover Concept is designed to keep stations on-air even if there is a technical failure. Based around a synchronising unit for two DDO machines, the Concept creates two 'mirrors'. Although only one device may be being used, the mirror updates every move on the system, with the second unit always in operation, even when not on-air.

Audiomation Systems

Audio Station is a control surface for DAWs. Its main features are eight assignable motor faders with mute keys, six assignable rotary controls, 12 function keys, transport control keys, an assignable shuttle or scroll wheel, control matrix software that can be assigned to any function and interfaces that allow connection to most of the leading DAWs on the market, plus MIDI, GPI and ADB interfaces.

Audioscope

The Model 3000 operates as a third-octave spectrum analyser and an RT60 reverberation time analyser. The third-octave analyser is specified with 30, 4-pole, enhanced filters, which conform to ANSI S1 class II. Exponential averaging is used on all third-octave bands and SPL levels, while integration time constants move from an 1/8th of a second to eight seconds in eight steps. Offering inputs for a microphone and stereo line, the microphone can be powered from phantom, while phase between line inputs are shown as spectrum difference. Also provided is a built-in, digital, pink-noise generator.

Augan

The OMX Series III features selectable multiformat 16-bit, 20-bit and 24-bit recording capability, conforming to OMF requirements. With the option to expand to a full 24 I-O system, the OMX Series III combines nonlinear editing with the flexibility of portable media. Features include a 4-input/8-output package in a 24-track editing configuration. The RC24 DiaMOnd remote control unit is aimed at sound-to-picture applications and gives integral 10½-inch TFT full colour 24-track display, QWERTY keyboard, dedicated tape transport keys and a jog-shuttle wheel. A standard SVGA colour monitor can be added to this device if desired.

beyerdynamic

The MCD100 condenser mic features preamplification and A–D conversion directly behind its microphone capsule to produce an AES-EBU digital signal at its output.

Broadcasting Control & Communication

Broadcast Media Server (BMS) software for Windows can deal with all radio station requirements, including remote-controlled broadcasting. Other functions covered are the editing of programme features, creating play lists, keeping disc archives (including music selection and find functions), jingle insertion and reporting for invoice purposes. BMS has been designed for local, regional and national operations.

Broadcast Software

Masterlog is an on-air control system for digital-audio hard-disk stores that can work either in live assist or completely automated modes. The package is able to control multiple playback devices, enabling the synchronisation of split commercial breaks for services with opt-outs. This is all handled over a Novell Network for centralised audio storage. Presenters can access any audio clips stored on a central audio file-server through the Mastercart digital-audio playback system, which is controlled by a dedicated Cartkey DJ console to give instant triggering of the desired cut. Masterlog and Mastercart can be run from CartEdit record and playback software.

Cadac

The F-Type FOH board combines compactness and flexibility with a range of pricing options and facilities and is claimed to be equally suited to cost-conscious repertory theatres or the demands of rock 'n' roll touring. Modules can be moved around to suit the application and the desk can also be expanded easily. The F-Type can be configured to provide sophisticated routeing and automation functionality for projects involving multiple acts and fast changeovers. Up to 12 subgroups, 24 matrix groups and 16 auxes, plus an optional VCA system with 12 VCA masters can be specified.

The Cadac Monitor Board is available in customer-specified frame configurations of up to 112 dual input channels, 48 outputs and 12 VCA masters. Cadac claims that the desk offers more than double the number of mix buses currently available from any competing design and VCA assignment is provided to 12 master and two grand master faders. Additional features include full MIDI control of external equipment and event controllers plus mute and VCA assignment recall.

The desk's arrangement of three independent Monitor outputs allows listening flexibility —engineers can monitor in-ear outputs via an in-ear system, speaker outputs via their own wedges and any inputs using either system.

D&R

The CineMix Plus surround console uses Dual Line input strips along with two 4-band semi-parametric equalisers, ten aux sends, 100mm automated faders for the lower section and 60mm faders on the upper section. D&R motorised faders are an option. Much of the desk's switching is digitally controlled, and it also including dynamics modules on each strip. A Central Processing Module houses fully automated joy-sticks and V.v (Virtual Vision), the company's visual interface for sound in the listening environment.

The Control Room Monitor has been designed to accept surround-sound encoding and decoding cards, with presets under software control.

DK Audio

The 4-channel MSD600C master audio meter is said by the company to be ideal for surround sound applications and employs a 7.8-inch colour screen, switchable digital and analogue I-Os, PPM with seven international scales, an audio vector oscilloscope, a phase correlation meter and an optional FFT-spectrum analyser, all in one compact box that can be installed into any mixing console.

Dalet

Dalet is showing four different control panels for

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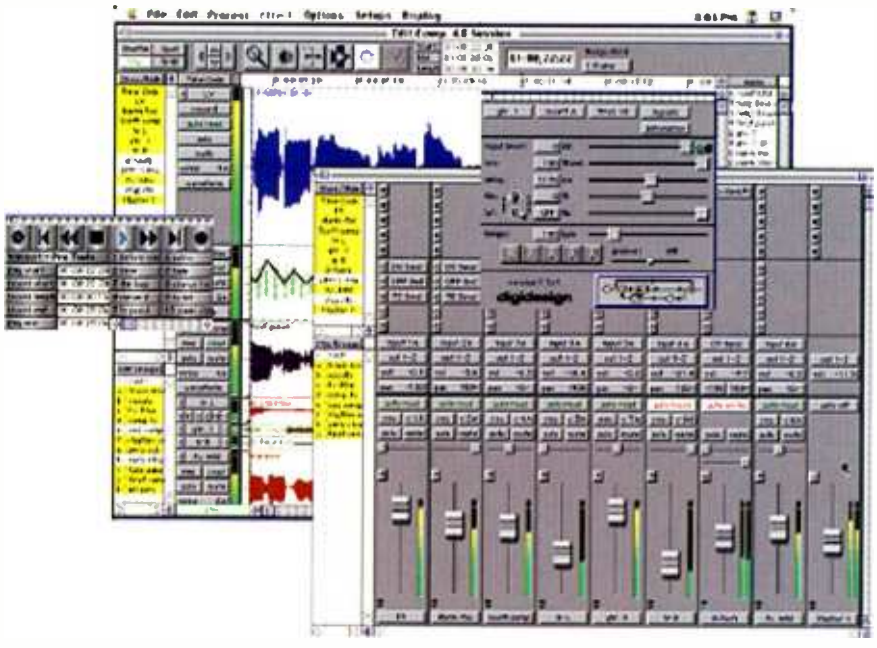


Version 4 of Digidesign's Pro Tools will be previewed in Copenhagen

specific broadcast applications. The modules are: scrub wheel and keys for editing; hotkeys for triggering carts where traditional NAB units have been replaced by computer-based alternatives; keys to scroll and locate items on databases; and fader controls. The products rely on nonproprietary hardware and run in Windows.

Danish Pro Audio

Three Brüel & Kjær instrument microphones, cardioid and omnidirectional, are aimed at encouraging engineers to close-mic in live and recording situations. The 4021 is a dedicated instrument mic that uses a thick-film preamp with SMD transistors, enabling the use of the prepolarised condenser capsule of 4011 and 4012 mics. The 4037 is part of the B&K compact omnidirectional series of microphones and a repolarised condenser cartridge, 12mm in diameter, provides the headroom and extended frequency range, and is coupled to a tiny FET-amplifier. Also new in the series of compact omnis is the 4051 which is suited to live work and film applications and are insensitive to handling noise, pop noise and humidity.

DAR

The company's D-Net Open Media system is designed for rapid and efficient audio auditioning and project transfers and allows the interchange of audio data and EDLs with other workstations and nonlinear video-editing machines. Audio Reels can be sourced from non-DAR machines such as Lightworks. Remote machine audio can be displayed on a local workstation and stereo segments can be auditioned over the network prior to copying across. D-Net can be configured to include DAR's AXIS AudioServer with large hard-disk stores for on-line access to commonly used sound effects and music.

The OMR8 Open Media Recorder is capable of 24-bit resolution, has standard 18-bit A–D and D–A convertors and replays direct from M-O or hard disk. Recording time can be optimised by employing a 2.6Gb optical disc for 30 minutes per side using all eight tracks (two hours per side for two tracks) or an 8Gb hard disk which increases the available time to three hours for eight tracks and 12 hours for two. The 3U-high 19-inch rackmounting OMR8 has full overlay and insert functions via simple Cut and Paste style editing. Takes can be slipped along a track or between tracks after recording, and the device includes record Undo.

DAS Audio

The Sound Touring (ST) 2000 Series is the direct result of the company's development of a new neodymium magnet structure compression driver—the ND-8—which has resulted in a dramatic reduction in size and weight. The range features a ST-215 high-mid pack and ST-218 bass unit. The hi-mid pack combines two B-30 15-inch drivers together with a centrally-placed horn-loaded ND-8 driver, with the bass bin loaded with two G-45 18-inch long excursion drivers in a folded horn design.

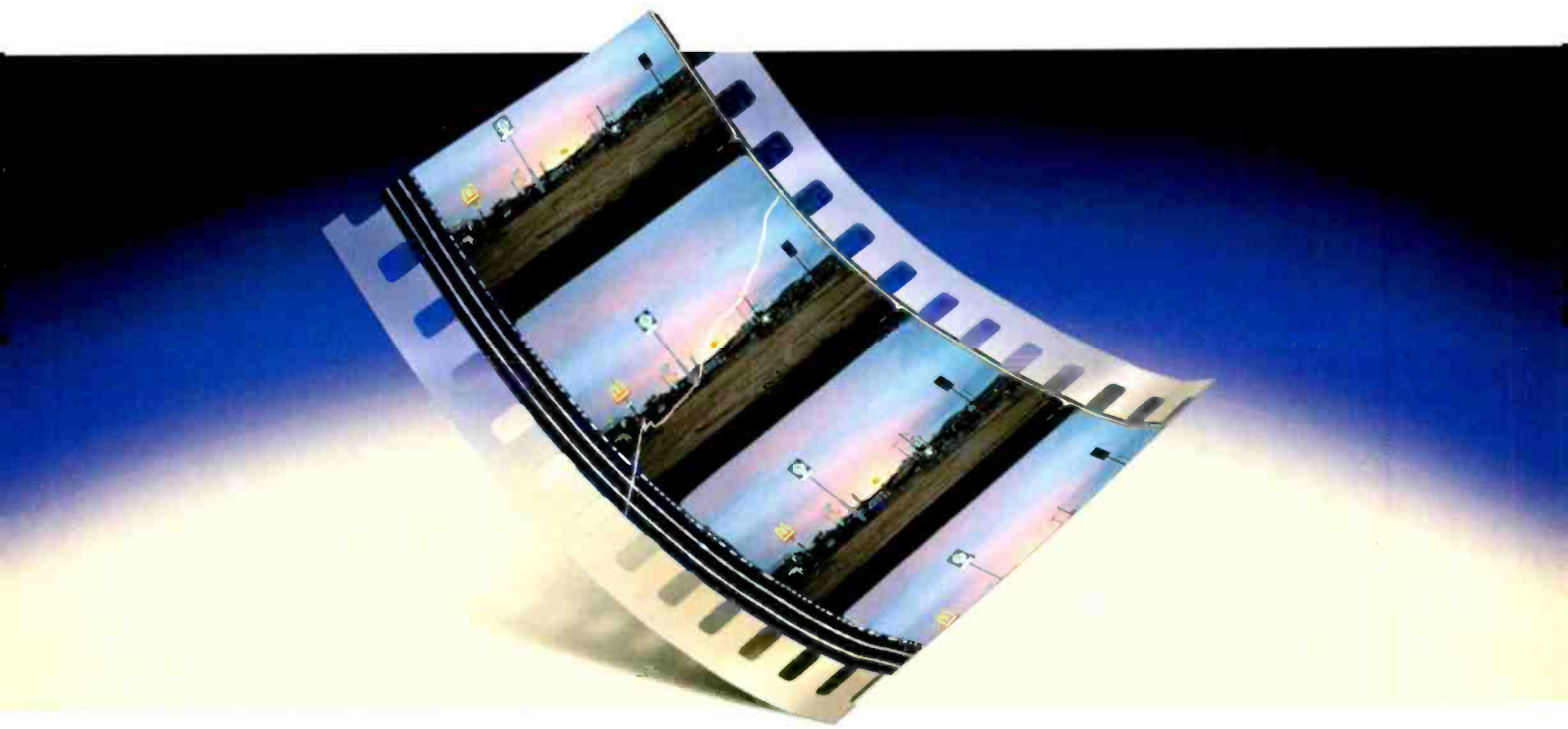
Dateq

The BCS70 desk features input modules with microcontrollers for audio switching, routeing and control; Dateq's own QDMsquared (Quad Dynamic Mix Minus) telephone system; advanced remote start, stop and tally connections; ergonomically positioned START-STOP buttons; balanced ins and outs; a special bus for extensive communication capability; and a setup button for easy changing of presets. The console frame can take up to 18 input channels and comes with VU, 40-segment LED bar or 200-segment Neontrack meter bridge. Mic-line channels come in a variety of designs: with or without EQ and with or without gain control.

dbx

The 1034 electronic crossover-limiter incorporates the company's proprietary PeakStopPlus limiting on each band of its two channels. The 1U-high rack unit gives 3-way stereo or 4-way mono, both of which can be selected on the back panel. A feature of the 1034 is the option of a summed low frequency output (mono subwoofer out), in addition to a switchable circuit that adds pre-emphasis

IN THE TIME IT TOOK TO WATCH THIS FRAME, CEDAR RESTORED THE AUDIO.



CEDAR 20 PC-based system

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HHB

One of Fostex' range of professional DAT machines—the D25

for constant directivity horns. Monitoring is carried out through three 8-stage lightpipe ladders on each channel.

Deltron Components

The company's colour-coding system for XLR multipole units has been upgraded and titled the Channel Identification System. The internationally-recognised, resistor colour-coding system has been adopted, which identifies channels by colour and number and can cope with up to 99 sends.

Digidesign

Pro Tools III v3.21 (PCI) is the first release of Pro Tools software with support for the new Pro Tools PCI hardware and provides an increase in processing speed and compatibility with the new generation of Power PC-based Macintosh platforms.

ProControl is an assignable and modular hardware control surface that adds tactile mixing and editing capability to the Pro Tools III system. The base system consists of a Master Unit featuring a touchscreen display, dedicated edit controls, scrub-shuttle wheel and transport controls coupled to a Fader Pak of eight moving faders with dynamic automation and LED-ringed rotary data encoders. The interface supports up to four Fader Paks for a total of 32 moving faders.

Digitech

The VTP-1 outboard processor features a valve, mic preamp, tube line amp-DI, EQ section and 18-bit A—D convertor. Each preamp control division has front-panel switches for mic-line input selection, phase invert, 20dB pad and 48V phantom power. Visual display of signal levels comes via twin illuminated analogue vu meters with clip LEDs. Equalisation is handled through a 4-band EQ section, which features two fixed and two sweepable bands.

The Studio Quad, digital, signal processor has four independent inputs and outputs and uses proprietary S-DISC technology. It has 128 preset factory programs and 128 user settings. The company is also showing the RPM-1 valve rotary speaker emulator.

DOD

The 512 is a stereo multieffects unit that offers 32 effects combinations for a total of 480 presets. The unit gives a choice of multiple reverbs, delays, choruses, flanges and pitch-shifting. Two independent sources can be plugged in and processed through two effects separately, which can then be combined with two real-time parameter editing controls.

Dorrough Electronics

The 40, 20, 12 and 400-2 series loudness meters feature the Dorrough Practical Standard audio metering system which is based on the relationship between RMS metering, integration time and peak levels. Using a continuous row of LEDs, the new meters show the actual energy contained in the programme material. A separate circuit handles a peak-reading LED. Meters can be configured to monitor either individual channels or the combination (sum) of the left and right channels. Another feature is an alarm, which is triggered when undesirable operating parameters, such as total audio drop-outs or over-driven levels, are detected.

Drawmer

The 1962 digital valve preamp has a high quality front-end using valve technology with two low-noise preamps plus two integral 24-bit A–Ds to act a means of committing audio straight to digital. This is aided by a zero overshoot limiter to prevent digital overload. Other features include variable tube drive, fine tune EQ, dynamic enhancement and variable high and low-pass filters. The unit can additionally be supplied as analogue only with the digital slot-in module available as a retrofit.

Euphonix

Mixview 2.6 software completes the total automation for the CS2000 desk and includes dynamic automation of EQ and automation features such as absolute takeover, move record, absolute offset and relative update modes. Other features include MIDI remote control and off-line automation file editing. Penny & Giles moving faders are now available on all systems.

The Time Transporter TT007 is an optional remote-control interface and translation module that integrates various time references and remote control types into a central network and adds to the CS2000's existing machine control functions. It can translate between MTC, MMC, Sony 9-pin and SMPTE-EBU LTC and is configured to store setups and patches of system configurations.

Fairlight

FAME is the result of Fairlight's collaboration with console manufacturer Amek in which it has adopted Amek's digital desk-controller surface, complete with SuperMove motor fader automation, as a front-end to its 40-bit floating-point DSP mixing engine for the MFX3 DAW. FAME combines a 24:24 MFX3 Mainframe with a 36-input digital mixer configured for multiformat work. Features include 24

Two of the most respected sound engineers on either side of the Atlantic agree on one thing...

if it's wireless, it has to be beyerdynamic UHF!

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The artists that perform on it and the engineers behind them will be different.
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Rob Colby and Yves Jaget cannot afford to make any errors with unreliable equipment. The audio quality, the transmission and simplicity of use of the product must be top notch.

It is for these reasons that Colby and Jaget prefer to use products manufactured by beyerdynamic.

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One of two new showings from Joemeek, the Tube Channel

faders, assignable EQ control, assignable dynamics control, eight auxes, 16 x 16 matrix, event-based EQ, machine control, OMF capability and the integration of audio edit list and mix automation data.

Dali-2T is designed for 2-track sweetening, editing and mastering applications and was conceived originally as an affordable platform for the company's Timefx time compression, varipitch and varispeed. It also includes 4-band EQ, digital dynamics and PQ encoding.

DaD is aimed at replacing multitrack and magnetic film dubbers for mixdown in film and TV and is totally modular. A remote control can address more than 300 tracks individually each of which can be slipped separately and reconfirmed working from M-O or hard disk.

Fostex

The D80 8-track hard-disk recorder has a front panel that can be lifted off to act as a remote control and reveals removable hard-drive bays. The unit comes with an 850Mb drive as standard for 18-minutes of 8-track recording at 44.1kHz.

The D80 can record on all tracks simultaneously, projects can be arranged in five 'virtual reels', and D80s and DMT8s can be linked for larger configurations. Audio can be edited with cut, copy, paste and move functions and the device syncs to MTC. Optical and coaxial SPDIF I-Os are provided.

Genelec

Genelec's Model 1039A studio monitors feature dual 385mm woofers, one 120mm-cone midrange driver and one 25mm metal-dome tweeter. The mid and high drivers are loaded by Genelec's proprietary directivity control waveguide (DCW). The 1039A is a fully active system and sits on a total of 1270W of amplifier power per channel.

Harrison

GLW Harrison and Klotz Digital Audio Communications have entered into a joint venture and launched a digital version of the Harrison Series 12 console (see feature in this issue). All digital-audio processing functions are performed by a digital system from Klotz Digital that is based on the technology used in the VADIS mixing matrix.

HHB Communications

New developments in the Advanced Media Products range include upgraded HHB DAT tape

AKG

The new WMS 300 from AKG is a 16 channel switchable and highly flexible UHF radio microphone system that delivers spectacular price benefits.

Providing ten different configurations in one affordable system, no other UHF radio mic system can match its flexibility.

There's a choice of handheld or beltpack transmitters and capsules for vocal, instrumental or lavaller systems. Three interchangeable dynamic and condenser heads are available to suit any type of vocalist or speaker – allowing the microphone to be matched perfectly to every show.

It's switchable to 16 spot frequencies within a UHF TV channel, with the ability to run up to eight systems simultaneously without intermodulation.

A total system solution that includes antenna splitters and boosters and a receiver that can be run on either AC or DC voltage.

Other exceptional features include up to 12 hours' battery life from three AA cells (7 hours with rechargeables), a compact ½U true diversity receiver unit and, of course, AKG's precision audio quality, rugged durability and backup as standard.

Based on the well-proven WMS 900 system (tours include Peter Gabriel, Rod Stewart, Wet Wet Wet and 1996's Simply Red dates), the WMS 300 delivers a total solution at an exceptional price.

FOR FLEXIBLE UHF PERFORMANCE, THREE HEADS ARE BETTER THAN ONE.

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WMS 300 FEATURES

- SR 300 RECEIVER: Switchable to 16 UHF frequencies for multichannel capability • Half 19" rack width • True diversity operation • Removable antennas.
- PT 300 HANDHELD TRANSMITTER: Interchangeable microphone heads • Highly efficient helix antenna for wide range • Extremely rugged construction • Special capsule suspension minimising handling noise.
- PT 300 BODY-PACK: Accepts dynamic and condenser microphones • Mic, mute or line selector for guitar, sax or lavalier options • Locking microphone input • Exceptional operating time.

WMS 300

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which has a new resin-compound cassette shell, antistatic lid, redesigned J-card and a freshness seal replacing shrink wrapping. The tape has been improved for lower block-error rates and claims secure archival for more than 30 years. Tape lengths have also been increased at no extra cost making the DAT125 the longest pro DAT tape available.

Huron

Programming Tools is a package of development software for the company's range of audio DSP systems. The Huron Digital Audio Convolution Workstation is an industrial, rackmountable, IBM-compatible PC with a dedicated, 256-channel, 24-bit audio bus running alongside the standard AT bus. Up to 20 DSP processor boards and I-O boards fit into the system, allowing large multiprocessor audio systems to be constructed. The Tools package is a rapid development framework consisting of an Object-Oriented C class library, an integrated DSP debugger-monitor, a DSP Assembler Template, as well as a number of support and utility libraries.

Joemeek

New units from the Joemeek stable include the Tube Channel, a 2U-high valve-based preamp with compression and enhancement. Also new for the show is the Pro Channel, a 1U-high, half-rack version of the Joemeek preamp, compressor and enhancer.

Jünger

Jünger's e07 4-band parametric EQ makes its debut in Copenhagen. The unit includes digital limiting, high sampling option and storage of user-preferred settings.

Klotz Digital

SPARK is a digital signal-processing and control system designed to perform all the distribution, regulation and control functions normally assigned to several different devices within a commercial PA system. It controls all regulating and switching functions for microphones and speakers, controlling inputs, outputs, routeing, tone regulation, power amplification and system monitoring. The system features four analogue inputs and eight analogue outputs and up to 64 paging stations, each with eight freely configurable paging buttons, can be connected and managed, with each of the outputs featuring a 10-band graphic EQ. Four plug-in slots add announcement and signal tone memory modules, each of which can store 128 items which can be played back to a schedule via an internal CMOS clock.

Lawo

The stand-alone MADI Tester allows users to plug into MADI and optical multichannel links and extract digital signals on AES-EBU or insert them for testing purposes.

Mackie

The SR40.8 is a 40+4:8 large format desk with a centre master section for FOH use. Features include a built-in 11x4 matrix mixer, left-right and centre outputs and group muting.

The compact 12-channel MicroSeries 1202VLZ desk has been updated and now offers 3-band EQ, separate stereo bus, solo and balanced XLR outputs.

Marenius

The upgraded version of the original ProDAT portable DAT machine—the ProDAT II—uses the same Sony TCD-D7/8 DAT transport, but has been toughened up still further. Time code is an option and the new machine uses upgraded and improved A–D and D–As.

Maycom

The Digicorder multifunction recording, editing, playback and communications unit is designed to allow reporters on location to record and edit a piece on site and then file it to headquarters over ISDN or modem connections. By using Digitrans receiving software at the other end it is possible to acquire the audio in the same digital format without cascading. Based on standard transport buttons the Digicorder also uses a jog-shuttle for adjusting headphone and loudspeaker volume and microphone gain and threshold. All levels and status information are displayed on a large LED, which indicates track data, recording level, length of audio, time remaining on disk, battery level and ISDN-telephone numbers.

Media Engineering

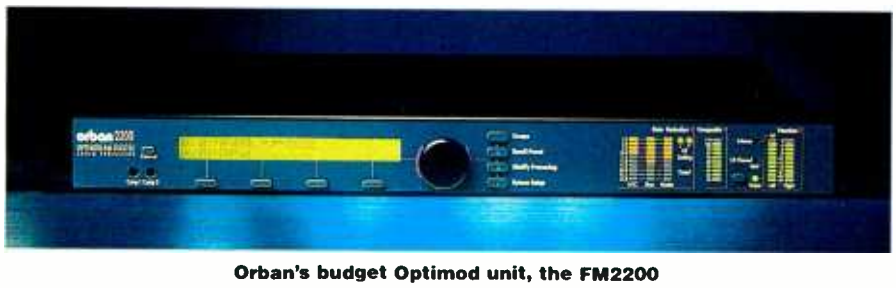
Media Engineering is in the last stages of developing an analogue radio mixing console called the ME-Mix which is minimalist in design, without EQ, which is made up of master and input channel module blocks containing only faders, push buttons and LEDs.

Micron

The SDR ultra-compact diversity receiver is supplied as a single-frequency unit, or with a choice of three switchable frequencies, the VHF receiver uses the same electronics as the company's modular multichannel studio and theatre systems, including the Micron Complementary Noise Suppression (CNS) System that offers a low noise floor with wide dynamic range and tight tracking of fast, high frequency transients.

Microtech Gefell

The KEM range of microphones use beam forming techniques to attain a frequency independent directional characteristic in



Orban's budget Optimod unit, the FM2200

8•BUS CONSOLE SERIES

32•8 24•8 16•8

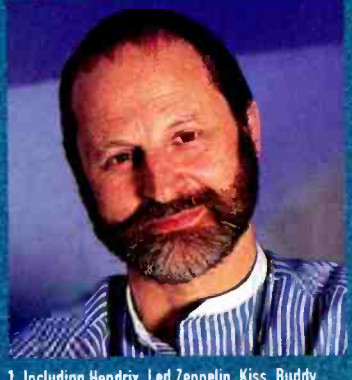
INSTEAD OF BRAGGING ABOUT OUR 8•BUS CONSOLE AGAIN, WE THOUGHT WE'D BRAG ABOUT EDDIE KRAMER BRAGGING ABOUT OUR 8•BUS.

Okay, bragging is too strong a word. But we are very proud when one of the most important, rule-breaking, producers in recording history has become a Mackie 8•Bus fan.

After all, Eddie Kramer's role in the making of popular music has changed its sound forever[1]. His recipe? *"Make a record unlike anything that's ever been heard."* So, while other engineers in London were churning out England's formula Pop of the Day, Eddie Kramer was across the console from a strangely-dressed young man from Seattle named Jimi Hendrix. Together, they broke practically every sonic and musical rule in sight. The result was an aural legacy of such originality that it still sounds amazing -- even revolutionary -- a quarter century later.

Sample Eddie's latest work on *"In From The Storm."* RCA VICTOR a brilliant orchestral homage to Hendrix with an astonishing array of some of the best players in the world. Leave it to Eddie to break more rules. ('Net surfers should check out the RCA VICTOR® Web Page @ http://rcavictor.com.) For a great read, pick up the *Jimi Hendrix Sessions* book by John McDermott with Billy Cox and Eddie Kramer (Little Brown), and on video, *Adventures In Modern Recording* (available from Mix Bookshelf, 510-653-3307).

Eddie hasn't gotten any more conservative over the years. So it's not surprising that a man with Kramer's receptiveness to change would add a 32•8 to his creative arsenal. A mixing console that costs hundreds of thousands less than those he's worked on for most of his awe-inspiring career. A console he says he likes for its *"...sweet EQ, dynamic range, and cleanness."*



Eddie wanted to do more than just take advantage of the creative and lifestyle options afforded by the project studio revolution. He also wanted to help DRIVE it. So a year ago, we agreed to lend Eddie a 32•8 in return for his feedback. Since then, we've learned Eddie is not shy about expressing his opinions. Luckily they're mostly good[2].

And Eddie Kramer recommends Mackie consoles to his associates, too[3]. In these cynical times (when pop stars accept millions to "endorse" products they admit later to having never tried), we at Mackie Designs think that's the only kind of "endorsement" worth having.

If you're in the market for a serious but affordable mixer, we hope you'll take a close look at the only 8-bus console Eddie Kramer says is worth having.

1. Including Hendrix, Led Zeppelin, Kiss, Buddy Guy, and more recently, his work with other Mackie mixer owners: Sting, David Abbruzzese, Vinnie Colaiuta, Stanley Clarke, Tony Williams, Steve Vai, and Carlos Santana.

2. He hates the location of the 8•Bus' talkback button.

3. According to Eddie, Eric Shenkman (Spin Doctors), Little Red Wagon Mobile Recording Studio, Bootsy Collins and John McEnroe have purchased 8•Bus consoles at his urging.

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which the pattern is super-cardiod in the horizontal plane but a beam pattern with an aperture of about 20° in the vertical plane.

The M100 dynamic cardioid is manufactured together with Mikrofontechnik Leipzig, a company with a long history in dynamic mic production. The dark-bronze finished mic's frequency response has been optimised for intelligibility, it has an internal elastic suspension to reduce handling noise and the wire head grille incorporates a pop filter.

MusicTronic

An MPA8800 power amp 3U-high housing can replace a traditional 12U–16U amp rack and includes a controller, line I-O connector panel and a speaker output connector panel wired for a 4-way stereo PA or a 4-send biamp monitor system.

The CU8240 controller contains limiters, EQs and crossovers while high power amplifier modules are designed to work continuously into 2Ω loads to yield a total of 6400W. The company's speaker range has two models which combine with the MPA8800 to construct a high power FOH system.

Nagra

Nagra has entered into a partnership with dCS Ltd from the UK to realise 24- bit, 96kHz recording and playback capability on the Nagra-D open reel digital machine.

The process uses special versions of the dCS 902D A–D and 952 D–A which can operate at 24-bit word lengths and at sampling frequencies of 88.2kHz and 96kHz sampling frequencies.

Nagra Lysis

The Integrated Broadcast System can handle sound-news editing, scheduling, broadcasting, statistics and administration functions. Also included is an object-orientated information system that can manage all multimedia documents, allowing the creation, processing, planning, sharing and archiving of all such files. The system has an open architecture, founded on distributed computing, client-server applications and high-speed networks, high capacity storage and multitask-multiuser operation.

Neumann

Showcased in *Studio Sound* last month, the valve M149 employs the K49 capsule—a hand selected version of the high tolerance K47 capsule—introduced after 1960 on the U47 to gives it an instantly recognisable head grill. A sensor circuit regulates and stabilises the heater voltage of the valve and compensates for any loss of output level due to long cable runs. Unusually for a valve mic, the M149 is transformerless and uses Neumann FET100 circuitry in the output stage to give self-noise performance on a par with modern studio capacitors. Nine polar patterns are offered along with a 7-step high-pass filter.

On Air

MusicMaster 1.2, is the latest version of A-Ware's on-air automation package and offers a range of features that can be used by a radio station. DOS memory management has been optimised to run MusicMaster under Windows 95.

Orban

Upgrades to the DSE7000 RAM-based, 8-track editor include a replacement DSP board with 24-bit internal processing, plus new v6.0 software. Among the effects now offered are parametric equalisation, Optimod compression, digital delay and Lexicon digital reverberation. The new effects will be standard on all new units and the multieffects package can be retrofitted into existing units.

Optimod-FM2200 is a low-cost digital unit eight programmable presets; 2-band processing with HF enhancement; protection processing; prevention of peak over-modulation; a standard all-digital stereo encoder-generator; a standard analogue I-O; optional digital AES-EBU I-O; remote control; a nonvolatile memory; alignment tone generator; and LED I-O gain-reduction meters.

Penny & Giles

Exclusively introduced in last month's *Studio Sound*, the multichannel 24-bit PP10 Audio Multiprocessor will be shown together with an associated range of Pythagoras Audio Software. This employs an unusual approach to DSP which involves a multitasking operating system. The system uses 32-bit floating-point architecture for simultaneous processing of up to 16 digital-audio channels, with further expansion possible. Processing is software-based, as is internal routeing for soft-wired patching, linking and cloning of multiple processors. Pythagoras Audio Software includes suites of processors, such as dynamics and EQ, and allows processors to be inserted at any position in a signal path, with the same processor type capable of being used many times in one patch.

Pro-Bel

The XD digital-audio router can switch signals synchronously at one sample frequency, while also dealing with signals at different sample frequencies asynchronously. The unit features two outputs per destination.

Re Technology

Designed for digital networks the 660-661 codec comes in two versions: one for analogue and digital audio, the other for analogue only. Intended as a cost-effective unit, the 660-661 can encode and decode either a stereo, mono or 2-channel signal, based on the ISO-IEC Layer II standard, or one or two independent mono ITU-T Rec. G722 signals. The unit's transmission system is based on a bit-stream using Layer II or G722 coding, with the first of these treated in single channel, stereo, joint stereo or dual mono modes. Bit rates range between 56kbit/s and 384kbit/s, while the sample frequencies range from 16kHz–48kHz.

A Layer II codec intended for cost-effective ISDN use, the 662-663 comes in two version (analogue and digital; analogue only) and uses the same configurations and transmission specifications as the digital 660-661. The 662-663 can transmit additional information (for example in-house data or RDS/RBDS) combined with the Layer II audio signal or along an RS232 channel, which has a capacity of up to 9600 baud.

Roister

The Acoustics Compensator digital speaker and listening room correcting equaliser adjusts frequency and phase nonlinearities and is

CONDENSER
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MICROPHONES

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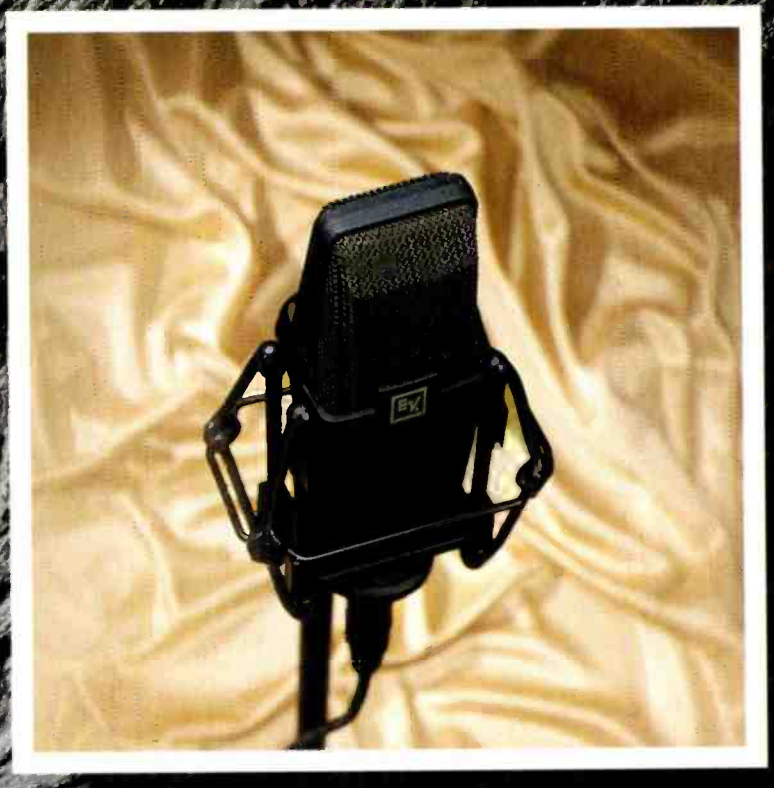

DON'T LET AN ORDINARY MICROPHONE INTERRUPT

Just when you've got a great take from an extraordinary talent you come up against noise, coloration and distortion. The ordinary condenser mic has to have its say. Cue the Electro-Voice RE2000. Plus a discrete (not to mention discreet) computer-grade power supply; plus a regulated operating environment that ignores the real world conditions outside its shock-proof casing; plus a best-of- all-worlds ultrathin gold-laminated diaphragm combining uniformity, wide dynamic range and exceptional transient response. Minus noise and interference.

Enough of the theory. This mic's been tested in practice by some extraordinary engineers, and they've got plenty to say. Like "the perfect mic for recording any acoustic string instrument" (John Beland of the Flying Burrito Brothers), and "the warmth of a tube mic — extremely quiet and sensitive, allowing me to pick up low-level material without adding noise" from Scott Weber of Buena Vista Sound, Walt Disney Studios. Tom Cusic of TM Century, Dallas "used less EQ to achieve what I look for. What goes in...comes out! It's also extremely versatile...from vocals to acoustic guitars to trumpets and violins", while Roy Thomas Baker (Producer of Queen et al) thinks "it's one of the most versatile I've ever used."

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Electro-Voice

Soundcraft's new Ghost project console continues its world tour of trade shows

available in three different configurations with multiple digital I-Os, volume and balance, polarity selection, bypass, optional analogue inputs, unbalanced and balanced analogue outputs and a PCMCIA Type II card driver which can address 512 filter sets.

Roland

The VS-880 Virtual Studio is an 8-track hard-disk workstation (with eight virtual tracks per track giving a total of 64 tracks) with recording-editing functions and a 14-channel digital mixer. Studio effects processors can be added via the onboard expansion slot which also accepts a dedicated VS8F-1 effect expansion board, and the unit is compatible with storage media via internal IDE and external SCSI interfaces. The unit is equipped with MIDI ports plus eight inputs, a pair of aux sends, a pair of stereo master outs, and digital I-O.

RTW

The 11529EBU DSP-based panel mount AES-EBU meter provides stereo metering at 32kHz, 44.1kHz or 48kHz sampling rates with DC filter, overload indication and various configuration modes. It is also fitted with a phase correlation meter and a volume display. Operation is also possible via a table-top housing which features a power-pack.

Sennheiser

The 3000 Series radio system is intended to be a more affordable replacement for the company's current EM4002-based systems and comprises an entirely new 16-channel switchable frequency receiver, and hand-held and belt pack transmitter units. The system is available in single or dual-channel receiver configurations. It features HiDyn Plus noise-reduction circuitry and the new transmitter models are compatible with the current range of Sennheiser receiver systems.

Shure

Shure's long-awaited UHF radio microphone system is available in single and dual UHF channel models. Microphone options include Beta 87 condenser, Beta 58 and SM58 hand-held, plus head-worn, lapel and instrument beltpack models.

SSL

New features for the Axiom Digital Production System include remote mic amps which offer a switchable limiter and selectable high pass filter in addition to gain, phantom power, impedance and pad functions. The system now includes a central channel control which allows adjustment of any channel from the central area of the Axiom. It also incorporates a bilevel capability for the control surface in which each physical Axiom channel can switch control to another processing channel allowing a smaller control surface to control a larger number of mix channels. Axiom now incorporated FreeWay and SDIF2 features from the SL9000j. FreeWay is a HiWay data stream on fibre-optic cables which allows interconnection over long distances.

Sonifex

The uprated version of the Sovereign series of broadcast mixers feature dual stereo-balanced inputs and an RIAA gram amplifier option. Two frame sizes are available, the MX10 with up to ten input modules and the MX14S, which offers up to 14 input modules with a seven module script space. There are controlled VCAs on each input channel, with stereo channels featuring configurable remotes for control of CD players, cart machines and turn-tables. Mic-line channels feature external inputs for cough muting and reverse talkback and come with 2-band EQ. A standard feature on all Sovereigns is a in-built 4-way stereo distribution amplifier that is additional to the main and monitor outputs.

Sonosax

Among the features on the new SX-S mixer are upgraded ergonomics, improved gain structure, quieter operation and switchable input modules. An accessory is the Film-Intercom Module, which takes the place of two input modules and gives ultra-high quality metering, remote tape machine start-stop, integral private intercom and advanced slate functions.

The StellaDAT II is cast in an aluminium frame and features a 4-channel mic-line mixer, a built-in monitor loudspeaker, LCD screen and new A–D and D–A convertors.

Sony

This PCM-3348HR high-resolution machine offers 48-tracks of 24-bit audio with a recording time of 45 minutes per reel and offering playback and record compatibility with existing 24-track and 48-track machines. Interfaces are provided for MADI and a range of parallel and serial

When it comes to monitors, honesty is the best policy.

Boyz II Men, Stonecreek, Philadelphia – HQ410u

Hans Zimmer & Jay Rifkin, Media Ventures, Los Angeles – Q210/QSB121

Gloria & Emilio Estefan, Crescent Moon, Miami – Studio A – Q412C; Studio B (inset) – Q212C

THE QUESTED CLIENT LIST includes some of the world's top names in virtually every field of professional audio. They choose Quested studio monitors for one, simple reason: they tell the truth. Whatever your application, Questeds provide an *honest, uncolored sound* – sound that you can rely on.

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Soundtracs Virtua console sets its sights on Copenhagen

ports for machine and system control. Sony also has a 24-bit DAT recorder which is portable and offers time-code recording and playback plus 16-bit DAT compatibility. Maximum recording time at 24-bit is 90 minutes and Lithium Ion batteries give three hours' operation per battery.

The PCM-600 M-O disc recorder uses 650Mb 3½-inch data M-O discs for an hour of 2-channel record and playback at 48kHz. This base unit's features are enhanced in the PCM-S610 M-O device which adds editing based around a waveform display working under Windows. Multiple machines can be networked together and controlled by a PC-based control and scheduling system, This system is additionally capable of operator assist or fully automated control and can run a complete digital on-air station with CD, DAT and MD machines as well as the company's DMXB4000 digital radio console.

UHF spectrum divider WD880A permits 60% more audio channels to be carried in the same frequency band when using the WL800 system. It achieves 24 channels between 774MHz –798MHz and 24 between 800MHz–820MHz and works in conjunction with the WRR850A twin broadcast diversity receiver, WRT860A miniature beltpack transmitter and the WRT867A stage vocal mic transmitter. All current Sony wireless systems can be upgraded to the new multichannel system by the addition of a WD880A.

Soundcraft

Ghost is aimed predominantly at the project recording sector and features extensive EQ, individually switched phantom power, a new mic amp, two stereo auxes in ten aux buses, four stereo returns, MIDI muting, time-code sync, MIDI machine and Sony 9-pin transport control and record track enabling from its computerised master section. MIDI continuous controller adjustments can be made from the desk's group faders.

The RM105 is a variant of the RM100 on-air console but where the latter was designed for the US market, the new desk has input modules designed specifically for UK and European broadcasters. Targeted at local radio and available in 8, 12 and 20-input configurations, modularity permits inputs to be selected from mono, stereo and Telco modules and also a script tray. Main features include two stereo outputs, VCA faders on all inputs, remote start-stop machine controls and standard or deluxe meter bridges.

Soundfield

The competitively priced SPS422 studio mic system is based on the same design principles as the more expensive products in the range. Developed for main microphone recording, the system consists of a multicapsule mic and a 1U-high rackmount processor that allows parameters to be adjusted in the control room.

Soundscape

New for the Soundscape DAW system is the AVI file player, EDL file player, RS422 machine control and DSP plug-ins for the SSHDR1. A prerelease version of v1.18 software supports all of the above functions and will be exhibited for the first time in Copenhagen.

Soundtracs

The Virtua digital console offers reconfigurability of its inputs and outputs in a manner normally associated with high-end digital desks. Aimed at postpro and music recording, the console has total dynamic automation of all parameters plus snapshots and a channel strip that features eight auxes, 4-band fully parametric EQ and two dynamic effects sections. On-board transport control functions run MIDI Machine Control

RPG

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Innovative Acoustics for the Digital Age

Founder member of Studer's D19 range, MicAD, will be keeping the company of the new MultiDAC

and Sony 9-pin. The system employs an assignable desk-controller surface containing the processing and an I-O rack containing 32 mic-line inputs, eight group outputs, eight aux outputs, 16 direct outputs and an additional 16 basic inputs. An additional 16 channels of ADAT digital I-O can be connected optically. In its basic form, the control surface has eight touch-sensitive motor faders but can be expanded to 32 faders and is supplemented by a computer monitor to help visualisation of desk and parameter status.

Struven Audio Vertrieb

The Stramp TachTimer, which has time code, tacho and transport direction inputs and a time code output, helps to reduce the lock-up times of ADAT and DA-88-style digital multitracks when in chase mode to analogue or video machines as the box generates time code from tacho and direction signals. Tach value and direction information is 'learnt' by the box from incoming time code and an accompanying handbook has details on 50 popular machines complete with pin configurations.

Studer

The On Air 2000 digital console is aimed at small production studios for TV and radio and broadcast continuity control. The desk is based on a proprietary DSP and has a patented user interface in which each input fader is assigned a field on a touchscreen on which to display channel status pictorially. Touching a symbol opens a central control screen which shows parameter values which can be adjusted using rotary encoders. Snapshot automation is built-in and the console is available in 6, 12, 18 and 24-channel configurations.

A remote control interface card for the D19 MicAD preamp can be connected to any RS422 device at three selectable baud rates with suitable controllers including Studer's digital desks, PCs, workstations and the D19 MicAD master controller for 128 mic or line input configurations.

The D19 family has been expanded with the D19 MultiDAC which has eight channels of 23-bit D–As and AES-EBU inputs as standard. Analogue outputs are transformer balanced and have a level pot and each channel can be set individually to operate over a wide range of sampling frequencies. ADAT and TDIF digital inputs are an option. A Super-ADC module is now available for the MicValve which improves the S–N ratio by 12dB.

The D424-2 M-O recorder offers 16-bit, 20-bit or 24-bit linear recording with the option of 20-bit A–D and D–As. Applications that require bit-reduced recording are also catered for with optional Lossless Real-Time Compression, Dolby AC2 or ISO-MPEG 1 Layer 2 plug-ins. Traditional transport keys are complemented by a jog-shuttle wheel, LED peak metering and editing capabilities from front-panel controls.

The machine has a built-in synchroniser with time-code reader and generator and can lock to an external video reference, to word clock, AES-EBU audio inputs or an AES-11 reference signal. Remote control is via Sony 9-pin and RS422.

Switchcraft

The TTP96 patch panel comes with a choice of three jack configurations: full normal, half normal and open circuit. Featuring fanned solder terminals for easy connection soldering, Gold switching contacts are also fitted.

Symetrix

The 620 20-bit A–D convertor, features 20-bit quantisation, selectable output word size, dither and noise shaping. The 620 outputs digital data in AES-EBU or SPDIF at sample rates of 48kHz, 44.1kHz, 32kHz or 22.05kHz. Digital inputs are also provided and the box can bit-rate convert from 16 to 8 bits.

Tascam

The DA-38 is a more affordable version of the DA-88 modular 8-track digital multitrack. Full DA-88 tape compatibility is assured in a machine that is effectively a stripped down version of the original. The DA-38 cannot slave to time code on its own but as it is sync bus compatible with the DA-88 it can be piggy-backed onto a suitably equipped DA-88 as an affordable means of expanding track capacity. A MIDI Machine Control interface is optional. Features include digital track copy and an internal digital patchbay, track advance and delay, shuttle control and offset along with auto punch in-out with rehearse.

The M1600 is an 8-bus in-line available in 16 and 24-channel frames with 3-band mid sweep EQ, one stereo and four mono auxes, four stereo effects returns and an optional meter bridge. The desk has eight mic preamps but more mic inputs can be added with the 8-channel MA8 outboard mic preamp. The M08 compact mixer has four mono and four stereo inputs with 2-band EQ, two auxes, four mono effects returns, channel mute and PFL and 60mm faders. Mono channels have inserts, there are four XLR mic inputs with phantom power plus stereo and aux bus sub inputs.

TimeLine

The MMR8 modular multitrack recorder is designed to replace magnetic dubbers in the film process. Working to M-O or hard disk the stand-alone machine has bi-phase control, ten

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TL Audio's new EQ-2 is reviewed on page 43 of this issue

time-code memories, event location, transport controls with record, rehearse and loop functions, a jog-shuttle wheel, master time-code offset and memory register trim functions, track slip and external remote track ready and transport control.

New software for Lynx-2 and Micro Lynx machine control systems adds Fostex RD8, Sony DVW500 and UVW1800, Studer D827 and Tascam DA-60 to the products' lists. Version 6.20 software for the Studioframe DAW adds a fast waveform display: OMF, SDII, AIFF and wave file transfer; new data sort functions; and an enhanced user-configurable editing interface.

TL Audio

The Indigo valve range is positioned in price below the company's existing range of valve outboard and offers stripped down facilities in a 4-channel mic preamp; two stereo EQs, one of which is fully parametric; a stereo compressor and a stereo valve overdrive unit. The Crimson range consists of low-cost solid-state versions of the Indigo range with mic preamps, equalisers and compressors.

Tube Tech

The EQ1A all-valve parametric equaliser is available in studio and mastering (EQ1AM) versions offering three, fully parametric, mid bands complemented by high and low shelves with high-pass and low-pass filters all operating on switched frequency pots. The mastering version is differentiated from the standard equaliser only by the use of precision switched gain controls rather than continuous pots.

Ultrasone

Ultrasone's 4-channel headphones exploit the ear's natural psychoacoustic capabilities to give a frontal or 3-D reproduction effect. All headphones offer loudspeaker-compatible spatial hearing with two (or more) channel recordings. They can also be used in conjunction with THX, Dolby, HDTV and DAB surround systems.

Yamaha

02R Project Manager support software runs on an Apple Macintosh computer and provides editing, librarian and remote-control functions for the digital desk. It allows the computer keyboard to be used to enter file names and dedicated key commands can be used to provide quick control of some 02R functions, such as the Automix Transport. Additionally, graphic and numeric displays of 02R data (such as time-code readings and EQ curves) can be displayed on the computer monitor.

The ProR3 digital reverb uses the DSP3 processor at the heart of 02R for 32-bit processing and is complemented by 20-bit A–D and D–A convertors.

The MD4 digital 4-track multitrack recorder is based on the MiniDisc Data format, and offers almost instant track locate and start capability with eight programmable locate points per song. Editing employs 'combine and divide' functions and cue-list style programmable playback and bounce down to any track with simultaneous 4-track playback. Other features include precision programmable punch-in and punch-out, simultaneous 4-track recording and a maximum record time of 37mins (4-track)/disc. A 4-channel mixer features gain, 3-band EQ, aux send and pan on all input channels, while the master section has four groups, stereo out, monitor out and direct track outputs, stereo sub in and stereo aux return. Sequencer synchronisation is via MTC.

Yamaha's latest affordable reverb unit, the Pro R3

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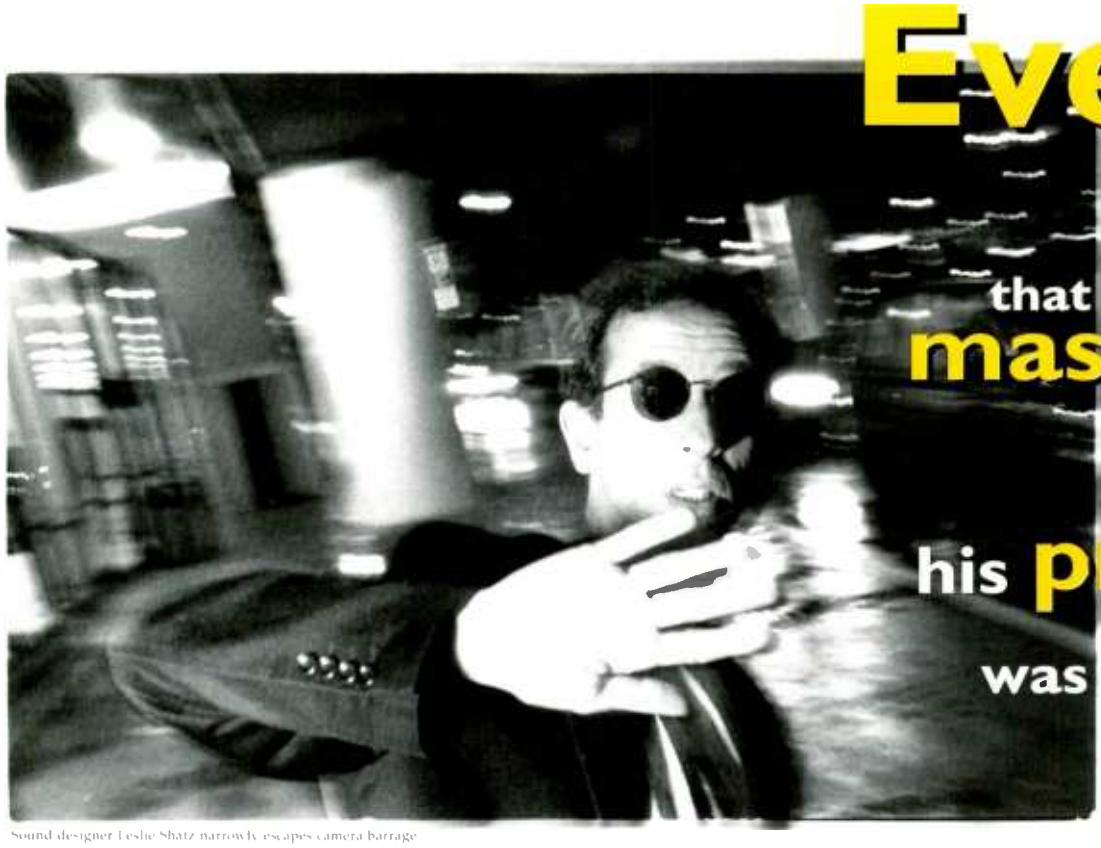

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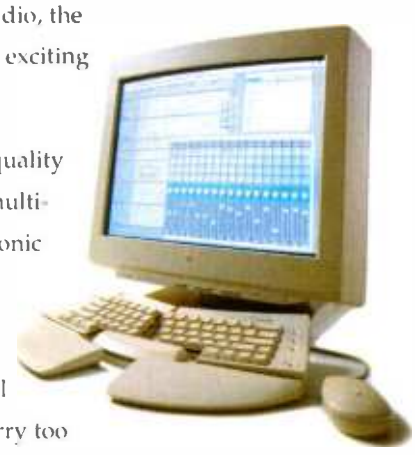

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Apollo 13 sound edited at Hastings Sound and Wonder Dog Music. *Dead Man Walking* sound edited at Hastings Sound.

Audio 96

The UK pro-audio industry and a strong international representation will again descend on the National Hall, Olympia, London from 19th–21st June for country's premier audio event—Audio 96—the Association for Professional Recording Services (APRS) show

The 1995 APRS show

AS IN THE LAST COUPLE of years, the show has again redefined itself because its organisers understand the dynamics of the changing face of the industry.

'The show needs to be different year on year because the market it serves doesn't stand still,' explains APRS Chief Executive Philip Vaughan. 'Our show does evolve–the mix of exhibitors develops and evolves.'

One of the constants is the popular briefings and workshop programme which has grown from a tentative start in 1994 to a major feature and crowd-puller at the annual event.

'We thought this was a worthwhile extra element to the show although we are not in the business of trying to replicate the many learned papers of an IBC, Montreux or an AES,' he adds. 'We feel that because the show is originated by a trade association, our thrust is toward the user of the products as they are now or will be shortly.'

Adrian Kerridge Chairman of the APRS

However, the biggest change for Audio 96 is one that visitors might not even notice but this is the first year that the APRS has enlisted the skill of exhibition organisers Single Market Events (SME). Audio 96 is in good company as the SME's exhibition and events credit list includes such high profile happenings as London Fashion Week for the British Fashion Council, BBC Good Food Show, the Cosmopolitan Show at Earls Court, consumer travel show Destinations, television conferences for PACT the independent producer's association, the London Television Programme Market and the Vision exhibition. According to Vaughan it was time to 'bring in some more ground troops' and SME's experience with the Vision show in particular meant it had a strong background in the broadcasting hardware side.

'The logistics of setting up the show are pretty demanding,' explains Vaughan. 'The APRS office is not just for selling and organising a show, it's involved in all the other activities of the industry's trade association and they don't get any less demanding.'

SME MD Tim Etchells says there is a good nucleus of support for Audio 96 which is being built on and that the relationship with the APRS is enjoyable and really working well. 'We are professional exhibition organisers,' states Etchells. 'We're bringing exhibition expertise and helping the APRS to structure the show in how it's promoted and making sure that the right information gets out to the right people at the right time.'

> 'As long as there is a professional audio industry there will be need for the manufacturers to put their products in front of their customers'

Vaughan acknowledges that some national pro-audio shows have suffered in the face of increasingly large and unifying European trade exhibitions, but points out that Audio 96 is not just a national show because the British industry it represents is an international one.

'Many countries have not had a national

show until recently,' he observes. 'The APRS exhibition began as a national show 28 years ago and it has been regarded as the definitive national show although it has a rather unique position because the UK music and broadcasting industries can't really be looked at as purely of national interest. Both are highly respected in other countries that look to what's going on in the UK as an example of how their own markets might run or how they might like them to run.'

The UK is, of course, also uniquely blessed with an unusually high proportion of market-leading manufacturers who have benefited from the show and from the UK's strong music and broadcasting force and Vaughan draws on the example of AMS Neve, which has recently returned to private ownership, and is putting on a powerful presence at the event.

'As long as there is a professional audio industry there will be need for the manufacturers to put their products before their customers,' he adds 'There are many ways of doing this but exhibitions remain a cost-effective way.' Audio 96 is also a mechanism for bringing new blood into the industry as it represents the industry's future—to take the project recordists and to introduce them to the wider world of pro audio. APRS Chairman Adrian Kerridge welcomes them all and is passionate about the importance of the exhibition. 'We need an audio show in the UK because as I've said many times in the past—to those that don't have long memories or those that are new—we should remember where the roots of this industry are. Between the 1960s and 70s it was the UK that lead the way. The Americans were producing lots of great hits but we actually put the music market on the map.' He adds that support from manufacturers and visitors is essential if the British are to continue to lead the world and with the increasing internationalisation of business it's more important now than it ever has been.

'We lost the shipbuilding industry through complacency, we lost the motorcycle industry to the very same attitudes of mind; we could end up losing what we have in the audio industry in the UK,' he suggests.

The British music business, he continues, is huge on export and needs to be supported by a strong studio and pro-audio infrastructure. Kerridge believes nothing should undermine the status of Britain on the international circuit and the respect for its operators, its facilities and its equipment. The existence of a strong APRS show, where all the UK industry can gather, is essential for its long-term well-being. With a year and a half of his chairmanship to go, Kerridge has many plans and is particularly keen to respond to the criticisms sometimes levelled at the show and the Association. 'They know where they can find me—CTS Studios—and I'll be very happy to talk to them.'

One of the most exciting developments on the cards for the APRS Audio show is what form it will take next year and the potential for a tie up with SME's Vision exhibition as Vaughan explains. 'SME is planning a repeat of its once every two years show Vision 97 and we will be holding Audio 97,' he says. 'The jury is still out on exactly how the two shows shall come together but the demand from the exhibitors, we think, is in favour of the two shows coming together. Anybody coming to London who is interested in the technology of the audio-video business will have a much stronger attraction if there are two events in parallel in the same area and in the same time frame.' SME's Etchells states that the two will be guided by what the industry wants. 'We want to do some research among exhibitors and visitors but the great thing about it is that all our options are open,' he says. 'Shows like Audio and Vision are run for the benefit of the visitors and it's them that we have to talk to to find out what kind of event they would most like to come and see.'

AUDIO 96 BRIEFINGS AND WORKSHOPS

AUDIO 96 is again hosting a series of high profile Briefings and Workshop sessions. Running in parallel with the exhibition, they offer visitors the opportunity to gather up-to-the-minute information on the latest technology, hot topics and techniques. Following two main streams, the Briefings and Workshop Sessions divide into application-based sessions with an emphasis on lively, hands-on interaction, complemented by more advanced technical presentations, designed to explore some of the latest technical and operational trends. The programme closely reflects Audio 96's five main focus areas of broadcasting, postproduction, recording, project studio applications and new media.

Among the speakers are industry authorities John Watkinson and Bill Foster, plus presentations sponsored by *Studio Sound*, *Pro Sound News Europe*, *One to One*, *Audio Media* and *Sound On Sound* magazines and industry bodies such as PLASA and the Institute of Broadcast Sound. Other organisations involved include Gateway School of Music, multimedia specialists Co-Active, and Dolby. Between them they will cover an enormous breadth of subject matter including gain structure, hints and tips for project studios, loudspeaker design, synchronisation, the format jungle, radio on the internet, live sound, sound for multimedia, on-line and off-line editing and developments in mastering, digital recording and the postproduction environment. Entry is free of charge.

The Briefings and Workshops are being organised by Dave Ward, Director of the Gateway School of Music, who is the APRS director responsible for the Association's Education and Training Initiative. In his view, Audio 96 provides an unequalled chance for anyone involved in audio to stay abreast of developments.

'This is London's one event where we can bring together all the threads of the audio business, take a good look at what is going on and disseminate some really useful information,' he says. 'The Briefings and Workshops sessions have been a part of the show for the last two years and grow stronger each time around. Our aim is to enhance the show with an element of potential value to every visitor. Audio 96 is the exhibition which attracts the widest possible audio audience, and as such, we have worked to ensure that the Briefings have a genuinely popular appeal.

'The feedback we received from 1995 was very encouraging,' adds Ward. 'The combination of hands-on sessions, which genuinely offer something to those seeking to develop their skills, with the more high-level discussions about the impact of new technologies, standards and working practices, means that there is something to appeal to everyone.'

Dave Ward, Director of APRS' Education and Training Initiative

Quo vadis, Harrison?

In fulfilling the promise of a hybrid console to become fully digital, Harrison have teamed up with Klotz to build the digital variant of the Series Twelve. Exclusive report by **TIM GOODYER**

ANY OF THE console manufacturers who have so far presented digitally controlled analogue consoles have been likely to speculate quietly about a forthcoming 'fully digital' system. But up until the recent NAB Convention, these discussions have tended to remain speculative rather than prophetic. Harrison have now turned theory into practice with their Series Twelve console and a little help from their friends.

Distinguished by its flexibility—and demonstrably popular in all applications on its home turf—the Series Twelve offers solutions in music tracking, multi-operator film work, A-V postproduction and broadcast (see the review in *Studio Sound*, December 1995). The initially analogue signal processing of the desk maintained the Harrison tradition of quality, and set standards that any move into digital signal processing would be expected to maintain. Entering into a partnership with recognised German digital signal-processing technologists, would therefore, have made an attractive proposition to the prestigious American console manufacturer.

Arranging a meeting with Harrison's Marketing Director Stephen Turley, Technical Designer David Ives and Klotz Digital's Michael Dietl in Las Vegas to discuss the fully digital version of the Series Twelve told a story in itself. What the two manufacturers have successfully undertaken is to incorporate Klotz' VADIS system as the digital signal processing element of the Series Twelve desk.

'We're showing a hybrid desk of which eight channels are digital and the rest are analogue,' says Turley of the desk on display at NAB. 'We've unplugged one of the Twelve's analogue modules and replaced it with the VADIS which is then under the control of the desk surface.

'From a user standpoint, it's fair to say that a digital channel as compared to an analogue channel is completely transparent. You can be using the same set of controls from one channel to the next and not have any idea of what the input and output structure is—the function is identical. That gives you an excellent hybrid, and since the majority of resources are analogue right now, this modular means of transferring yourself from the analogue world to the digital world is ideal. You may even end up with a desk that is some form of hybrid.'

Although the modules (which Turley terms 'chunks') represent eight channels of analogue channel handling, VADIS is able to support up to 256 channels in the same space and feed them onto one bus. The optimum, however, is to substitute 64 digital channels for eight analogue channels. The practical limit to the size of a digital Series Twelve is determined by the capacity of the automation system rather than any aspect of VADIS, which can be indefinitely cascaded. At present the team have accommodated over 280 channels.

The aim has been to substitute digital processing and control in a manner as operationally close to that of the existing analogue console as possible. Including the specifics of channel facilities.

'Essentially we've set it up to imitate what the analogue desk does,' confirms David Ives. 'There's an input trim, 4-band parametric EQ with high-pass and low-pass filter, fader, dynamics section, 16 auxes, 48 multitrack outputs, four programmes and eight groups.

'The approach we took was that we have an architecture that works in these applications and that to mimic this architecture in the digital realm it would be equally applicable.'

THE KEY to the Series Twelve is the interface between Harrison's automation and user-interface and Klotz' digital system. Consequently, the background to the alliance between both companies is Harrison's protracted search for a suitable partner with whom to develop an adaptable digital desk.

'A year ago at NAB we were in a collaboration with AT&T,' Stephen Turley

recalls. 'But that fell through—from their side, and not ours. Without control over AT&T's development programme, they were free to abandon the project and we were left facing the prospect that we still had to make a digital console if we were to meet peoples' expectations. However, after meeting with Klotz, we felt that it had a digital-processing system that worked. Klotz is highly regarded in Europe and we felt it fell in line with our reputation around the world—so the collaboration made a logical choice.'

Technically, the interface between the console control surface and the digital signal-processing electronics were crucial to the perceived viability of the digital Series Twelve desk.

'Since we had been talking to a series of different manufacturers in the course of trying to build a digital console, we had become quite good at interfacing,' comments Ives. 'This last exercise—with Klotz—has been quite smooth.'

Klotz' Michael Dietl picks up the story: 'The problem was one of interfacing between the internal Harrison digital structure and the VADIS netware interface. It was a problem—we had to find some hardware but it was not an insurmountable problem.'

Interfacing the series Twelve with other audio equipment is accommodated by VADIS' A–D convertors and its compatibility with the AES-EBU standard.'

'There are a couple of unique things about the way the user-interface of the console is set up, ' comments Turley. 'There are a lot of shared functions that allow us to operate the entire console from one operator's position; they allow us to configure consoles in a very space-efficient manner—you may need a 96-input console but you only have a 6-foot wide space for it... The fact that you can sit in the sweet spot in the centre of a seven or eight feet of console space and control the entire mix is a significant point. The touchscreen interface has some powerful aspects to it. All of these things have been retained from the analogue console and transferred to the digital console.

'Anywhere you go on the panel is "real" and the same thing applies to the touchscreen. Everywhere you go is a useful section of the panel and the same is true of the panel. Whether the touchscreen offers a "significant technology breakthrough" is not really the point; I think it's just an enabling technology is really the point. I think it's just an enabling technology that makes the technology easier to use.'

These are early days for the Harrison-Klotz enterprise, but the partners are already aligning their philosophies regarding the future of the console and of the industry itself.

'Down the road, digital is going to change in the way it works,' comments Turley. 'I think that both Harrison and Klotz are going to be involved in defining the way digital audio will work in the next decade. I expect that we're going to see more, and more standardised interfaces. We feel that we're the automation experts and that Klotz are the digital audio experts and that by marrying the two together we really have got a good perspective on providing a solution—a solution that works.

'The Series Twelve is such a flexible desk design that we've targeted it at film, video postproduction, broadcast and, of course, music recording. We've got two main issues being driven by each other. One is the market: what does the market demand. The other is what the technology will allow us to do. If you look at it historically and recognise what Harrison has done, look at the innovations that have come out of this company over the years... You can interpret this as pure marketing hype if you want to, but I would suggest that you look to us to continue to be the ones doing the innovating. We've got innovative architecture in our machines, we've got innovative work surfaces. Sure we've made a mistake or two over the years, but ☞

A united front—engineers and marketing men present the new console at NAB. Left to right: Klotz Digital's Michael Dietl, Harrison's Technical Designer David Ives, and Harrison's Marketing Director Stephen Turley

Key to the design philosophy of the digital Series Twelve console is consistency of operation. The routeing section (above) for example, will handle both analogue and digital signal assignments in the same way

we've learned from those and I'd look to us to be on the leading–probably sometimes the bleeding edge–of technology. We're looking at what's available on the market and I think we've got the engineer-ing expertise to take what technology has to offer and make it work in the console.

'Actually, the whole thing is market driven,' says Ives. 'We've been forced into delivering a viable digital desk by the interest of major players in the game.'

'I'd have to agree,' Turley concurs. 'They are demanding: "where is your digital desk?" We've been moving in this direction for a while but in a few years it won't be a question of where is your digital desk, it will be an assumption: "Where's your digital desk?".

'The strength of our automation will continue to be a selling point. There will be places where detailed automation is essential and we're going to be showing up more and more in the future. We've been really strong in the film market because film demands a dynamic automation and we're seeing the desk filter into music market where a lot of mixing is required. The recall functions are so comprehension that we're seeing a lot of interest from broadcast...'

> 'We've got innovative architecture in our machines, we've got innovative work surfaces. Sure we've made a mistake or two over the years, but we've learned from those and I'd look to us to be on the leading–probably sometimes the bleeding edge –of technology'–Stephen Turley

Although the announcement of the digital desk is hot, the conversations taking place at NAB suggest that it is not speculative technology and that orders and deliveries can be expected later this year.

'There's been a lot of interest,' confirms Turley. 'Boy, there are some people in here may have made up their minds this morning...'

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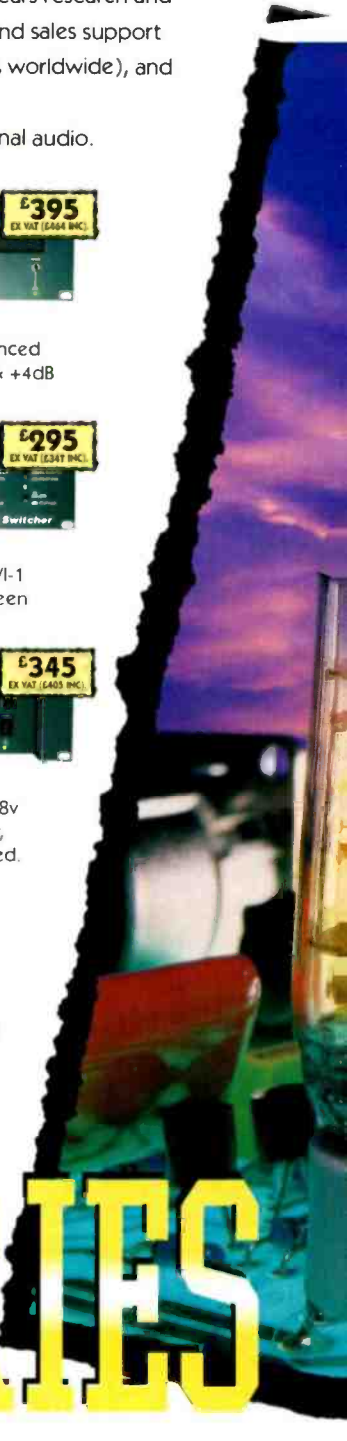

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Mike Exeter - DEP International *(Commercial recording facility owned by UB40)* - *"The EQ on the M-1 is very smooth. The broad bandwidths are great, with particularly nice results on acoustic guitars and bass sounds."*

Dennis Charles & Ronnie Wilson - 1st Avenue *(Producers - Eternal, MN8, Michelle Gayle, Dana Dawson, Louise)* - *"The EQ-1 and C-1 units helped us put the power into the EMI album "Power of a Woman" by Eternal."*

Chris Porter *(Producer - Take That)* - *"I bought one of the first EQ-1s and I've enjoyed using it immensely. It gives a unique quality to the vocals in particular - Take That's 'Back For Good' is a typical example of the EQ-1 adding depth and presence to a vocal track."*

Refurbished Neve 8088 all-discrete console with Flying Faders in Sunset Sound's Studio 2. A companion machine room houses Studer A-800 multitracks and Ampex TR-100 Series mastering decks

Sunset Sound

Sunset Sound is one of the oldest independent recording complexes in the world still under private ownership. **JAMES DOUGLAS** reports on the installation of a refurbished Neve 8088 in the legendary Studio 2

WHEN WORD HIT the street in Hollywood–ground zero for the West Coast's recording industry–that Sunset Sound was planning to install a new console, the rumours ran fast and free. What were studio owner Paul Camarata and his team looking for? With few exceptions, Sunset Sound has a reputation for building custom boards utilising class-A mic preamps and discrete equalisers. What modern console could match these exacting standards?

As it turned out, Sunset Sound stayed with what it knows best: vintage recording hardware. In November last year, the facility unveiled its refurbished Studio 2, which now features a 96-channel, Neve Model 8088 in-line board with Neve's Flying Faders automation. The control room's custom Augspurger monitor system was retained, and extra outboard equipment installed in custom cabinetry. As well as a cosmetic overhaul of the control room and performance area, a third, larger isolation booth has been added.

Founded in 1959, Sunset can legitimately claim to be one of the oldest independent recording complexes in the world still under private ownership. In addition to three recording and production environments within the main Sunset Sound Recorders complex located in the heart of Hollywood's Media District, the adjoining Sunset Sound Factory offers two tracking and remix rooms.

'Sunset Sound has always enjoyed a reputation for offering the best sounding equipment that money can buy,' Camarata reflects. 'In the past, we've built custom consoles using primarily vintage API modules. But our clients were asking us for a Neve-equipped tracking room. We knew that Studio 2's performance area offered great acoustics. What we needed was a great sounding, all-discrete Neve board. These

MODIFICATIONS TO SUNSET SOUND'S 8088

- Expanded master Record, Overdub and Remix switching, with individual channel override.
- A new EFX Return switch per channel to provide multiple, interlinked functions
- Expanded Solo-AFL system interlinked to highly modified and enhanced Record-Overdub-Remix switching
- Programmable Channel and Monitor Cuts plus Inserts via Flying Fader Automation
- Expanded stereo and mono Auxiliary Send buses to provide eight sends per channel, with pre-post switching
- Connection of main stereo bus into the cue system for musician foldback
- Stereo monitoring of Auxiliary Buses
- Extensively rewired talk back system
- Addition of comprehensive patchbay with TT jacks
- Master control for AFL output
- Original quad/4-channel output bus converted a true balanced Left-Right mix bus, for a lower noise floor and reduced RF pick-up, plus the provision of stereo inserts

vintage, all-discrete Neve console are like silk; they have a warmth that you cannot achieve with modern, IC-based designs.'

THE NEVE 8088 in question was originally installed at Rumbo Recorders, a facility owned and operated by Daryl Dragon of Captain and Tenille fame. 'When I started looking for a replacement board, I tracked down this gorgeous 96-channel, discrete class-A 8088 in Canada. It is one of only three larger-format 8068 consoles built by Neve with 48 input channels and 48 monitors. Of those three, ours was the last one to leave the factory, and the last design to be supervised by company founder Rupert Neve.'

But, after almost 16 years, any console needs to be checked out, and any aged components—particularly capacitors and relays—replaced. In addition, recording techniques and practices from almost two decades ago don't match the demands of today's high-tech engineers and producers. 'We'd already planned to add Flying Faders automation,' explains Craig Hubler, Sunset's Studio Manager, 'but we wanted to have our new 8088 thoroughly checked out and some new switching and logic functions added. One person whose name kept on being mentioned to us was Fred Hill, who runs FC Hill & Associates, down in Nashville. For many years, Fred worked for Neve as a factory-trained technician.'

After carefully checking out Fred Hill's credentials, Sunset agreed for him and his crew to receive the various channel modules from the 8088 while the studio's engineering staff checked out the frame wiring and power supplies. Also specified were many modifications that Hill offers as part of his refurbishment and revitalisation package for vintage Neve consoles.

'I didn't want the 8088 destroyed in the process,' Camarata stresses. 'These modifications were done internally, without changing any silk screening or adding switches or knobs.'

'Fred Hill and his team went through every electronic and electrical function, and made sure that every component was up to par,' says Chief Engineer, Mick Higgins. 'He fully checked out every channel module—all capacitors were replaced—and examined and replaced each component if necessary.'

According to Hill, it was a painstaking task, and one that cannot be rushed. 'Our proprietary process is thorough,' he offers, 'and we prefer not to cut corners. Each module is visually inspected, and any sub-boards removed. We then complete a mechanical check, including the repair and replacement of loose or missing hardware, broken knobs, switches and connectors.

'We replace all of the aluminium and tantalum electrolytic capacitors with industrial-grade, ultra-low-leakage components. We also replace all audio switching relays, using high-quality, gold-contact, gas-filled units. We then perform a proprietary cleaning process, which removes any corrosion and other oxidation products from switches and other circuit components. We clean each module with an industrial detergent, after which we rinse them using filtered water and dry them in a climate-controlled oven. Finally, we hand lubricate all of the switches, detents and bushings with a proprietary, silicone-based materials.

'Once the mechanical and cleaning steps have been completed, the daughter boards are tested on custom jigs, using an Audio Precision System One. Any faulty transistors, capacitors or other components are replaced, and the entire module is tested from all inputs to all outputs. A series of automated test sequences have been programmed into our System One to perform frequency response, bandwidth, THD+n, SMPTE IM, absolute noise and MOL at 0.1% distortion. Results from all of these tests are printed out, and will be furnished to Sunset Sound with the completed modules.'

Studio 2's remodelled performance area, with new large-capacity isolation booth

PHOTOGRAPHS BY ELIZABETH ANNAS

Fred Hill (left) with Sunset Sound's Chief Engineer, Mick Higgins, checking out rear-panel wiring and logic interconnections for the enhanced solo-AFL systems fitted to the revitalised Neve 8088 console

HILL ALSO MODIFIED the console to provide additional outputs from the obsolete quad buses. 'We have rewired the four quad switches to offer VR-like functions. Engineers will now be able to access the remix bus directly from the module output, and use spare modules for effects returns.'

Hill also expanded the auxiliary-send functions—increasing the total to eight per channel, switchable either pre-fader or post-fader—plus adding monitor solo and programmable mutes. 'As delivered, the 8088's first four auxiliaries were fed from the channel,' Higgins explains, 'and the last four from the monitor section, as if it were two separate consoles. Now we have full level-assign from channel and monitors into all the aux send busses, with independent pre-post selection. Aux Sends 1+2 and 5+6 are mono, while 3+4 and 7+8 are stereo, so these tend to be designated foldback cues for most of the time. During remix they serve as stereo effects sends.'

'As many *Studio Sound* readers may know,' Higgins continues, 'these vintage Neves were set up with master Record, Overdub and Remix switching; in Record mode, inputs to the monitors was from the bus outputs, and in Remix mode it was from the multitrack returns. All this means that you are wasting a lot of functionality —particularly the multitrack buses as subgroups during mixdown.

'Part of Fred Hill's Revitalisation Package for these boards is to provide revised logic switching for the master status controls, so that we can override the Record-Overdub-Remix status on individual channels, and also access other signal paths. For example, we can go directly from a channel output to the Stereo Mix, or route the post-channel fader signal to the monitor path fader, allowing it to be used as an aux send via multitrack buses 3–16. In this way, when the monitor fader in not being used as a tape-machine return, we gain 14 additional—and very useful—aux buses during mixdown.

'In Remix mode, the post-monitor fader signal is connected to the multitrack assign buses, allowing these faders to be used as mix returns—by routeing to Buses 1+2 and hence to the LR master mix via a new dedicated patch point—or to Buses 3–16 as subgroups.

'The OVERDUB switch on each module now provides multiple functions. When the board is in Record mode, it forces the monitor path to Tape-Out, overriding the master 'Tape-In' status. In Remix mode, it forces the ☞

monitor path to Tape-In, instead of Tape-Out. And finally, when used in conjunction with Hill's new EFX RETURN switch, it forces the channel status into Record mode, and the monitor path to source Tape-Out. In essence, this single button enables a Record Channel, with the monitor fader assuming the Track Monitor function for overdubbing while the 8088 is set up under master status for Remix mode. It make for an extremely intuitive system.'

The EFX RETURN switch per channel also provides multiple, interlinked functions, as Higgins explains. 'It inhibits channel solo-in-place muting for selected channels, and enables channel AFL for quality checking when the module is connected as an effects returns or overdub source. It also forces the selected channel path into Remix mode, so that the channel can be used as an equalised effects return when the Master Status is set to Record mode for tracking.'

Finally, if this switch is used in conjunction with the OVERDUB switch, it provides the function described above.

The Solo-AFL system has also been modified to take into account the overrides available from the master Record-Overdub-Remix switching. 'To allow the monitor paths to be used as mix returns,' Sunset's Chief Engineer explains, 'a Solo-Link function is provided on the monitor panel. When selected in Remix mode, the channel and monitor path solos systems are tied together for a solo originating from either system –provided that they have been set up for Solo-in-Place mode. Now, the engineer is provided with simultaneous muting for monitor path return signals and channel-path mixdown signals when a SOLO switch is pressed on either system. It's a very useful modification.'

The various switch modifications designed by Hill and the Sunset crew were also tied into the 48-channel Martech Flying Fader automation system via a custom interface that allows insert, channel and monitor-cut switch data to be stored with dynamic fader information.

'What we needed was a great sounding, all-discrete Neve board. These vintage, all-discrete Neve console are like silk; they have a warmth that you cannot achieve with modern, IC-based designs'–Paul Camarata

IN ADDITION to the new console that graces the control room, significant changes have also been made to Studio 2's performance area, including the construction of a third, 10-foot x 15-foot isolation booth. 'The original iso booths were simply too small,' offers Hubler. 'Now we have a larger one, designed by George Augspurger, that runs along the side wall, and which we can also interconnect with our other in-studio booth to provide additional space for drums, or small combos. We have improved visual communications between the control room and the iso booth so that everybody can see everybody else.

'George Augspurger conducted a preconstruction and postconstruction TEF analysis of the remodelled studio area to ensure an acoustical similarity. If anything, we have tightened up the low-end; we figure that by rebuilding the acoustic treatment in the ceiling, we have gone back to the original sound of the room.'

First sessions in the new room took place in November, with Sunset regular Don Murray mixing tracks with Producer Akira Toguchi for an upcoming JVC album release. 'Geoff Gillette has also been in recording some film soundtracks,' Hubler recalls, 'and Rick Neigher and Neal Avron have been mixing a new BMG album for Leah Andreone. Ross Hogarth has been recording tracks for a Muppets TV soundtrack with composer Richard Gibbs. In January, we have a tentative booking with Producer T-Bone Burnett and Engineer Pat McCarthy, for a tracking and remix project with Jimmie Dale Gilmour. Many of our clients are vying to make Studio 2 their "new home".

'Fred Hill and his team did a painstaking job for us,' Paul Camarata concludes. 'This board is a dream come true. Fred's modifications will offer outstanding flexibility to our clients, and the sound is going to be remarkable.'

ORIGINS OF A LEGENDARY CONSOLE

SUNSET SOUND'S revitalised Neve 8088 was built originally for Rumbo Recorders, LA. 'When I started looking for a board,' Paul Camarata recalls, 'everything kept coming back to my getting get a discrete Neve. I had a lot of people assist me in making this decision, including Ed Wong at Groove Masters—Jackson Brown's studio, which has a very nice 8078. Ed has a 'Neve Registry' book that lists the heritage of all these vintage boards—like a family tree of where they started life, and where they ended up.

'The standard 8078 format was 40 inputs plus 32 monitor, but with class A-B electronics, not class-A discrete. There just weren't any 8078s available; according to Ed Neve only built like 20 or so of them. And the other thing is, Hollywood is really loaded with 8078s, so if we'd put one of those in at Sunset, we'd have competition. Eventually, I came across this very unusual 48+48 8088, of which Neve only built three; this is the last one they produced. And, as far as I can tell from the Registry, this is the largest frame class-A discrete board ever built by factory, with 48 inputs and 48 monitor. And it's also the last one they ever built in this format.

'So I talked to Rumbo, located the original tech that was there, and talked to him. It turns out that, in 1979, Daryl Dragon [founder and original owner of Rumbo] wanted to purchase a NECAM automation system for his big Trident. But Neve said that they couldn't sell him a NECAM system, but they could sell him a NECAM system on a new board—that was their policy back then.

'In the end he agreed to buy a large-frame 8068—he was using double 24-tracks at that time, and needed more that 32 inputs, which was standard on the 8068s of that era. Neve agreed to build Rumbo a custom board with 48 input channels. At the time, Neve had built a couple of 8088s—which were 40-input versions of the 8068—and Daryl got them to built an 8088 with an 8-channel addition. According to the Registry, this is the only 48-channel version they ever built, although there are boards out there that have been fabricated from two 8068s, for example, to provide bigger mainframe sizes. But that modification can put extra strain on power supplies, bus noise and other system elements. This board we have in Studio 2 was designed and built as a 48-channel frame, with 48 monitor inputs for 96-input mixdown. It's a unique piece of recording technology.

'According to the paperwork, this board was contracted around March of 1979, and delivered in October 1979 for an unbelievable amount of money! The 8088 was at Rumbo until 1986, when it was sold to the Columbia Broadcasting Academy, in Canada. We eventually heard about it through a contact at Bearsville Studios in New York, which also has a very early 40-input 8088. By then I had decided on what I wanted to buy, and had been calling everybody I knew. Eventually, through Bearsville, I got in touch with David Moyles at Coast to Coast Audio, who brokers old Neves and other vintage gear. Everything just kept coming back to this 8088 that David Moyles had; everybody had told me that David only sells gear that's pristine, that he was someone you can trust in terms of its condition.

'For me, those vintage boards have a way of finding their owner—you rarely have to advertise a board like our Neve 8088, if you put the word out to the right people. So I flew up to Toronto, went through it with David, and it looked great. I already had the history on it from Rumbo, who had told me that it had been in really good shape when they sold the console to Columbia Academy. I then talked to the head tech there, and it sounded like it was his only child; he was really sorry to see it go! I could tell from the way he talked to me about the 8088 that he hadn't abused it with any modifications—a lot of these boards get really hacked up. That's when I decided we should have this board for Studio 2. It's going to be a great investment for us.'

CONTACTS

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Classical

Typically modest and self-effacing, Andrew Cornall is none the less one of the world's most influential and respected classical producers. **SUE SILLITOE** is in conversation with the man from Decca

VETERAN CLASSICAL PRODUCER Andrew Cornall is not the easiest man to track down. When he isn't out on the road recording, he's either immersed in paperwork or busy arranging his next session for Decca or for the recently revamped classical label Argo that he manages.

Given the grandeur of his titles–Senior Executive Producer for Decca and General Manager of Argo–his office at Decca's Chiswick HQ is a rather anonymous place. There is little evidence to indicate what sort of a man Cornall is. No family photos litter the desk, no gold discs adorn the walls, there is no evidence of the Best Classical Producer Grammy he picked up in March last year, or the nomination he received at this year's Grammy Awards. This is probably because 43-year-old Cornall is a modest man who seems bemused that anyone might want to interview him, even though amongst his peers he is considered to be one of the unsung heroes of the classical music industry.

Since joining Decca in 1977, Cornall has worked with some of the world's most famous classical orchestras and artists including Charles Dutoit, Herbert Blomstedt, Sir Georg Solti, Sir Charles Mackerras, Bernard Haitink, virtuoso violinist Joshua Bell and Vladimir Ashkenazy with whom he has a particularly close relationship.

He has only ever worked for Decca although some of his recordings are conducted on behalf of other labels.

'Like most people in the record business I fell into it by accident,' he recalls. 'I don't think anyone ever decides to do classical music production as a career move. I know I didn't.'

If his life had gone the way he'd anticipated, Cornall would now be a Decca engineer rather than producer, because that was the job he originally applied for after getting a taste for recording during his student days. In Manchester, where he studied for his degree, he ran the university's electronic music studios and became involved in electronic composition which he thoroughly enjoyed.

'In a sense the recording angle started then,' he says. 'But I didn't know it. I just enjoyed electronic music and that was all.'

From Manchester he moved to East Anglia to study for his Master's. However his tutor was killed in a car crash the week before the course started, so finding himself minus the composition element of his course he concentrated instead on recording and an historical thesis.

'At that time, the university had a very large electronic music studio which was run by Tryggvi Tryggvason, an ex-Decca man who is now a freelance recording engineer. As there was no-one to teach me composition I did a lot more recording and got a very good grounding from Tryggvi.'

At the end of the year, Cornall followed Tryggvason's advice and applied to Decca's head of studios for a job as an engineer. When no response came back, he took a job with the Open University as a music consultant. Then, months later, a letter arrived–not from the head of the studio but from the head of A&R who had been passed his original application because the studio felt that with two degrees he was overqualified to be an engineer. The A&R department immediately offered him a job as a producer and the rest is history.

As a classical producer, Cornall is quite clear about the differences between his role and that of his contemporaries in the pop field.

'All producers are there to get the best out of the artist so that you create the best recording possible,' he explains. 'But the fundamental difference between classical and pop is that classical producers are primarily there to get someone else's musical interpretation down on tape and are not part of the creative team to the extent where they are responsible for the orchestra's sound. Classical orchestras and musicians already have their own sound and all the producer should do is capture it. You certainly don't want to influence it.'

Cornall adds that at Decca producers are the bridge between artists, engineers and the technical crew.

> 'The fundamental difference between classical and pop is that classical producers are primarily there to get someone else's musical interpretation down on tape and are not part of the creative team to the extent where they are responsible for the orchestra's sound'

'The producer runs the session and represents the artist's view to the engineer and visa versa,' Cornall says. 'Our role is to help everyone do the best job they can. I'm not hands on with the equipment–I leave all that to the engineer–so I'm not rushing about moving faders up and down. Of course I discuss what I want with my engineer and will ask for specific things such as more woodwind or advise caution if there is a big *tutti* coming up. The engineer reacts to me as though I'm the map reader–the navigator who steers everyone through the session.'

Another fundamental difference between classical and pop producers is where they record. Classical producers are rarely, if ever, studio bound. Usually they are out on location, going to wherever the orchestra is based.

'If I'm working with Riccardo Chailly, for example, I'll go to Amsterdam where he is based and record the orchestra *in situ*. Most of the time we record straight onto stereo 2-track and mix as we go along, but we always bring the tapes back to the Decca studios for editing.

'You do get producers who work in certain places more frequently just because they are responsible for a particular artist. Basically we have set artists that we work with as it makes sense to develop a good relationship by working with them regularly. But everyone has worked with each other's artists and none of us are indispensable.'

LIKE DEUTSCHE GRAMMOPHON, Decca works its location-recording equipment as a number of complete recording systems. In Decca's case it has four sets of equipment that are constantly touring the world. One set is constantly in the US, while the others could be anywhere–it depends on who needs them.

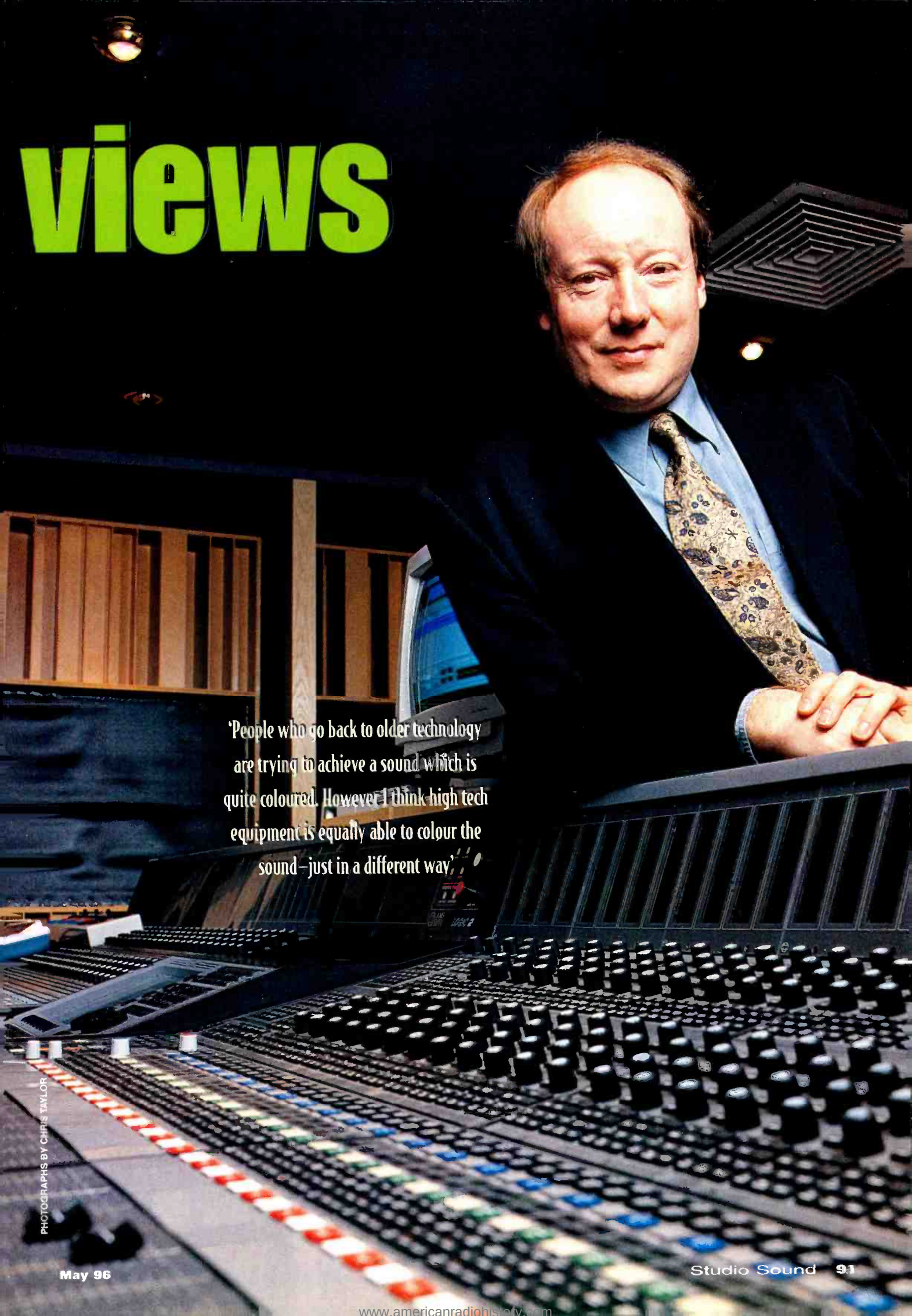
'In Europe we have two driver riggers who ship the equipment around and help set it up at the venue,' explains Cornall. 'All the setups are pretty much the same so we are familiar with everything, no matter which rig we get.

'We rarely hire in equipment and only hire a mobile for specific live events or for concerts that will be TV broadcast. I'm about to go on tour with British composer Mark-Anthony Turnage and we will use a mobile because we are recording a series of live concerts in London, Cologne and Frankfurt.'

The majority of classical venues don't have dedicated control rooms which means equipment is often set up in wild and wacky places such as the vestry of a church, an adjacent office or even–and it has been known–the bathroom. Even so, Cornall reckons that finding recording venues is becoming increasingly difficult because of the problems of external noise.

'Ideally the venue will have good acoustics and be noise free,' he says. 'We are always on the look out for older buildings, concert halls and so on where we can record but ☞

views



'People who go back to older technology are trying to achieve a sound which is quite coloured. However I think high tech equipment is equally able to colour the sound – just in a different way'

PHOTOGRAPHS BY CHRIS TAYLOR

often you find somewhere with great acoustics but no space for a control room or there's dreadful traffic noise.

'Pop people wouldn't be seen dead working in some of the places we work in because they are used to specific studio locations with proper acoustic treatment and plenty of isolation.'

Cornall's favourite venue is Concertgebouw in Amsterdam—because he knows it so well. 'It's one of the great halls of the world and it has a wonderful acoustic, which doesn't mean to say that it's easy because you still have to work hard and know it well to get a good sound. You can't just stick your microphones up in any old place and hope something will come of it. The reason it works for us is because we know what to do in there. Other people have used it and got it terribly wrong.'

Familiarity and experience are important but even then it depends on the repertoire being performed, claims Cornall.

'You have to modify your recording technique to suit the repertoire. Ideally one would find a venue that suited the music, but if that isn't possible you have to pay close attention to microphone technique to get the right sound.

'It's the small variations you make to the microphones that are crucial in the context of the particular piece being performed. If you are recording a romantic orchestral work you will be looking for a different feel to that of, say, a smaller contemporary piece so you have to modify how you use your microphones because the acoustics of the hall won't change.'

Cornall—like most Decca producers—is an advocate of the famous Decca Tree, developed in the 1950s and much copied even to this day. Cornall explains: 'There are variations of the tree—for example some people miss out the middle microphone. But it is basically a system of fewer rather than more mics that we use for all sorts of projects from full orchestras to chamber music, string quartets and even soloists.'

Cornall believes that the strength of this system is that it provides a uniform sound. However, each produce0r and engineer will have a slightly different approach in terms of the height of the microphones, where they are pointing and the filler mics used to score instruments around the whole stereo imaging.

'If we are recording an orchestra, we might have some woodwind mics or a mic on the horn, brass section or percussion' elaborates Cornall, 'but we don't go littering the place with them. We try and use the skill of the engineer to capture the sound of the orchestra and let the musicians balance themselves as they would in a concert so that they are not forced to play unnaturally.'

Cornall says he rarely gets involved in decisions about mic preamps, preferring to leave that to his engineer. Likewise with the choice of microphones unless he is working with a freelance engineer in which case he has more of a say.

'The Decca engineers know what I like —so much so that we don't even discuss microphones unless they are trying out something new. When that happens I express an opinion and ask them to change mics if I don't like the sound we are getting.

'To me the important thing is the end result. Frankly I couldn't care less if a mic is held together with string and wire provided it sounds right.'

Cornall explains the retro mic revival in terms of more modern equipment giving a more clinical sound. But he adds: 'I don't always agree with the argument because sound can be coloured in more than one way. People who go back to older technology are trying to achieve a sound which is quite coloured. However I think high tech equipment is equally able to colour the sound—just in a different way.

'What I want is to hear the timbre of the instrument replicated back in the box. If it doesn't sound the same then I have a problem because essentially it's these timbral sounds that I am trying to capture. The skill of the crew—especially the engineer on the desk—is by far the most important element,' he concludes.

The only time Cornall finds himself in a recording studio is when he is mixing and editing. Once there, he certainly isn't conservative about the choice of equipment and welcomes any technological advances. He is currently enamoured with the new Logic-2-equipped mix room at Decca.

'I've used a similar desk at Air and I know it's going to be good. It will certainly help in a lot of the mixing situations, particularly with my Argo projects which are more multitrack-based than the standard classical fare. Those are usually two-track and we only record on multitrack for backup. The only time we use multitrack on Decca projects as the principle source is during live concerts where the sound has to be remixed.'

As CD is now the preferred format for classical music, you might expect a producer like Cornall to have strong views on future technologies. However, he would only say that he looks forward to multichannel CD which he believes will gradually creep into people's homes.

'There is no doubt that the surround sound experiments we have done to create a concert hall ambience sound fantastic,' he adds. 'I can imagine it even in my small living room because it expands the sound out to the extent that you feel part of the acoustics. As a future development, I think it's great.'

Anything that helps spread enjoyment of classical music is welcomed by Cornall who believes the market as a whole is shrinking again after its rapid expansion in the 1980s when CD was first introduced.

Cornall believes concentrating on quality and promoting listenable new talent is the only way to keep classical music alive. 'I suspect we will end up with a situation where we only ever record the best. Quality is good for the whole industry, especially if it encourages a star system because—as with the pop market—it is the stars that will keep sales buoyant in the end.'

ARGO

AS GENERAL MANAGER of the Argo label, Andrew Cornall is responsible for the day-to-day running of the label which he took over in 1989. His aim is to promote the best in contemporary classical music and to encourage new talent, particularly from the UK and US.

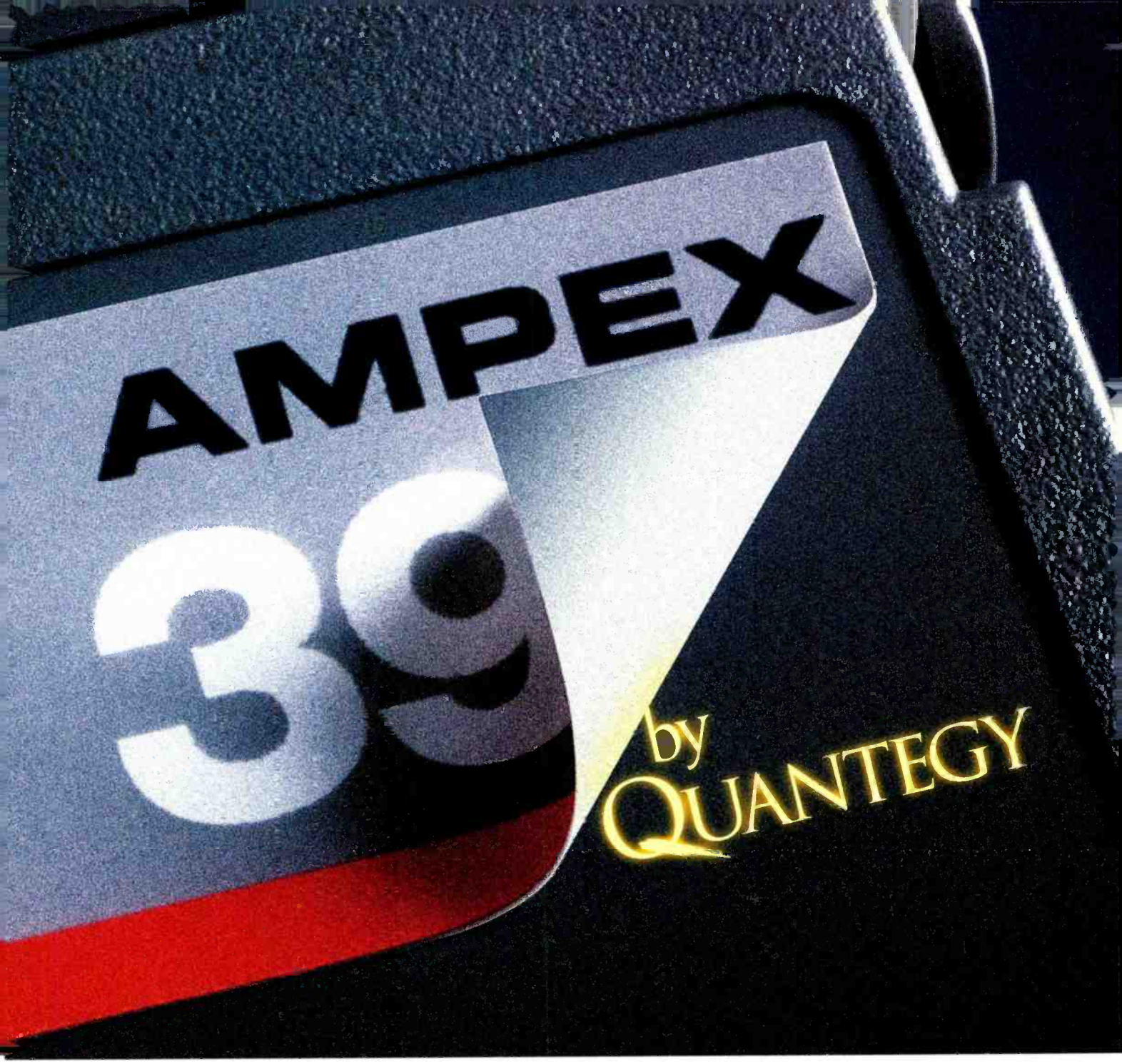
Artists associated with Argo include Michael Nyman, John Harle, Aaron Kernis, Mark-Anthony Turnage, Michael Torke, Graham Fitkin and the Balanescu Quartet.

'The pop industry thrives on new music but in the classical industry contemporary writing has in many cases become removed from the general public.

'I don't see why contemporary music should be ghettoised and I'm keen to encourage composers who want people to listen to their music and can capture an audience without intellectual compromise,' Cornall asserts.

He believes that this move toward new, yet listenable music is the only way to stop the classical world from becoming a museum art form. Composers like Mark Anthony Turnage, whose work has a bluesy quality to it, and Michael Nyman who can pack out a live venue with audiences who would never normally go to a classical concert are the essence of what Cornall claims Argo is trying to do.

Much of the music Argo handles is multitrack based and is created on electronic instruments and synthesisers. Even orchestral material is being approached in a more modern way. Cornall points out that while many of the Argo composers inhabit the classical concert world, their influences are just as likely to be pop based, so its hardly surprising that their compositions and recording techniques are well and truly up to date.



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Another green world

When a company wants to reposition itself in the marketplace it has to be sure it's making the right move: **ZENON SCHOEPE** looks at the design philosophy behind Focusrite's new Green range

AT THE END of 1994, Focusrite began work on a range of processors intended to be a more affordable option than the company's present lines. Despite manufacturing multi-thousand-pound bits of gear that are popular and desirable, the word on the street from distributors was confirming what the British company had suspected for some time—there was room for their equipment further down the food chain. 'What we were getting back from the market was that there was a serious change going on in the way people go about recording,' explains Project Manager Rob Jenkins. 'It's on the tail-end of the digital recoding boom—they buy an ADAT and Pro Tools system with a Mackie 8-bus and they want a high-quality front-end to go into it so they buy Red 6s and 7s as their recording paths. The percentage of that market we were hitting was pretty small because few people can afford a £2,000 box. Our distributors were telling us there was this community in the recording industry that aspires to Focusrite products but can't attain them.'

Answering this call for cheaper modules presented a problem for Focusrite because it has made its bread and butter on the hand-wired and labour intensive nature of it's equipment and reducing cost was not a simple matter of down grading an existing product. New designs and most importantly new methods of manufacture—at least for Focusrite—were needed and how the company coped is a good example of design and production engineering.

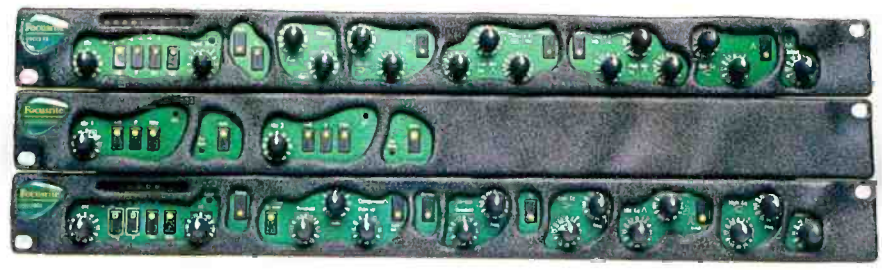
The resulting Green range was designed with the same Focusrite philosophy of high-quality performance with savings made in components, labour and the box itself.

The Reds are very visible in a rack and had created a strong brand image for the company that it had never had with the Blues but the curved and contoured Red front panels are expensive to produce so the company opted instead for the unusual solution of aluminium moulding for the entire casing which contributes to the distinctive look of the modules. This method also has advantages for manufacture in that all units share the same back panel and main box and only the front-panel moulds are specific to a particular product. Spray painted, the box employs little features like a rear overhang to protect connectors and also saves on assembly time as the electronic boards can be literally slid into the box and fixed in. The moulded finish is stippled and hides any defects as Jenkins explains. 'The shape has resilience in the manufacturing stage and if there are slight differences between one panel and the next—who cares? —that's part of the feature. It means the reject rate is very low,' he says.

Focurite wanted the 1u-high modules to stand out from the rack and differentiate themselves from other pieces of gear and it has to be said that this has been achieved with these peculiar looking pieces of outboard. The front panels actually stand proud of the rack surface due to their thickness and require only two, recessed, rackmounting screw holes due to the strength of the material. Some would say that it would have been enough for Focusrite to have built cheaper ordinary looking modules but Focusrite Managing Director Phil Dudderidge disagrees. 'The problem is that if its a me-too box then people might think its me-too electronics,' he says. 'As with the Red range, if it's sitting in the rack it has to say something about what people can anticipate from the product. The Green range had too stand out and say "Look, we're not just an average piece of kit".'

> 'For every person that buys a Red unit there are probably 99 that would like to but can't afford it,' adds Dudderidge. 'We want to reach some of that aspirational market although we'll never reach all of it'

Electronically, additional savings were made in the use of surface-mount components and automated assembly techniques of the 4-layer PCB allowing a Green board to be ☞



The first three of the Green range: distinctive or simply wacky?

made in minutes in comparison to the hours it takes to hand-wire a Red unit, for example. Jenkins claims that the biggest savings were made in the Green's lack of transformers.

'We looked at everything else like the quality of the op-amps, resistors and capacitors but there was nothing in it. A transformer, on the other hand, costs £25-ish to us and when you multiply that up to the customer it's a big burden. Every-thing we've done up to now has been interfaced with transformers and the nice thing about them is that they're really easy—they're the perfect interface but for the Green range we had to get rid of them. That wasn't an easy decision for us because how good the inputs and outputs are is the heart of a product. The internal electronics are simple to take care of and control but if you don't interface correctly you've blown it,' he states.

'We used the qualities of transformers as the major elements in starting the design process—we had to have a high quality first stage and the output stage would have to have the resilience of a transformer.

'We've used video drivers to drive the output because they're used to operating at about 100MHz and driving a piece of dodgey co-ax whereas a 5534 wants to see a minimum of 600Ω and no capacitance at all. Ours doesn't care what it's connected to,' he claims.

The output stages are thermally protected so if it is driven straight to ground the unit will simply overheat, switch itself off and self-heal when unplugged.

Performance-wise the Greens are no slouches according to Focusrite and compare favourably with Reds in most areas—transformer properties aside. 'Its very, very quiet with low distortion but its going to sound completely different than the Red—its going to be more transparent for a start,' says Jenkins.

Perhaps the biggest question is what the appearance of affordable Focusrite modules will do to the credibility of the brand name but neither Jenkins nor Dudderidge expect any problems with this. They also point out that the arrival of the Reds actually boosted sales of the more expensive Blue units among those who wanted the exclusivity. 'If we say we're producing a module for less than half the cost of the Red there will be certain impressions in the market for what that will mean and we're attempting to meet them,' explains Jenkins. 'It will look unique, it will have it's own character and style, and the sound quality will be far superior to anything in its price range.'

'For every person that buys a Red unit there are probably 99 that would like to but can't afford it,' adds Dudderidge. 'We want to reach some of that aspirational market although we'll never reach all of it.

'People used to use very expensive Studers and now they use very cheap ADATs. I understand that there are some-thing like 100,000 ADATs not to mention DA-88s out there—how many Studer 24-tracks are there?' He asks. 'There is a huge market out there that we've only been scratching the surface of. We don't want to spoil the brand positioning of Focusrite—we worked hard to achieve that—but we do want to reach a larger part of it than just the thin air at the top of the pyramid.'

THE RANGE

INITIALLY there are three products—a dual mic preamp (Green 1), the Focus EQ (Green 2) direct recording module, and the Green 3 'Voicebox' microphone signal path module. These are expected to be followed up with three other new units towards the end of the year with as many as another six rumoured to be in the pipeline.

What is significant about these products is that they will all be very much sub £1,000 in price.

The Green 1 dual mic preamp has a variable gain from +10dB to +60dB with switchable phantom power and phase reverse, a 75Hz high-pass filter, externally controllable mute and a peak LED. The direct recording module Green 2 has mic, line and instrument level inputs passing through six stages of EQ—high-pass and low-pass filters, low and high mid parametrics and low and high shelving bands with bell switches. There's also an overload LED, output fader and VU response bar graph metering. The Green 3 voicebox is designed to offer a single channel of high quality mic input processing for critical applications. It has a single mic preamp identical to that in Green 1 but follows this with a notchable mid parametric and low and high shelving EQ, a compressor, de-esser and noise reducing expander.

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Digital sound quality: preference and preservation

Movements in mastering have exposed unexpected shortcomings in our understanding of the process. **FRANCIS RUMSEY** gets to grips with the complaints and their possible causes

THE RECORDING INDUSTRY now has at least 15 years of experience of digital audio, yet many of those who use it still have nagging doubts about aspects of sound quality, and many conversations go on about certain devices or processes affecting the sound in a way which on the surface seems technically improbable. There are certain unalterable facts about digital audio which cannot be argued, and if sound quality is being undesirably modified at some point in the signal chain it can only be due to one of a limited number of factors. The problem in understanding what is going on arises when none of the factors which affect sound quality seem to be responsible. Here, strictly theoretical stances are not usually helpful because if the man with the ears says that the sound has been changed while the man who wrote the book on digital audio says that he must be talking rubbish, nobody is going to get anywhere near the bottom of the issue.

Particularly interesting is the area of mastering and replication. Mastering is the final stage at which anyone responsible for a recording can have control over the sound. Among the mastering community lie some of the best ears in the business, and uncomfortable issues keep arising about the way in which certain CDs come back from the pressing plant sounding nothing like what left the mastering house. A large armoury of tools is available to the mastering engineer which can be used to 'fine tune' the sound quality of an album before it is pressed, often these days involving redithering and requantising of high resolution recordings for issue on 16-bit CD. As we shall see, mastering is often not the 'purist' trade one might expect. In fact it is a case of almost anything goes if it makes the final result sound better, but we have known for years that mastering is a black art—that's nothing new. The question is not whether mastering changes the sound (it often does), but whether the sound the mastering engineer produces is the sound which ends up being played to the consumer when listening to the CD. One is dealing with the difference between intentional changes made to the sound during mastering, and unintentional ones which arise because of some as yet unexplained phenomenon.

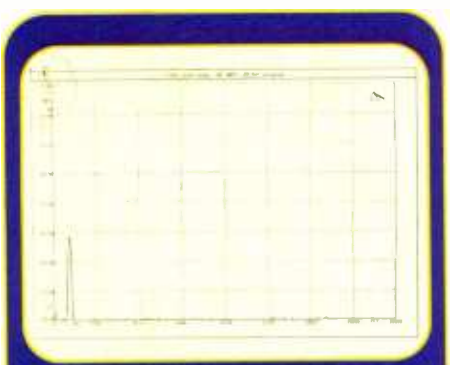

Fig.1: 1kHz 20-bit tone truncated to 16 bits by removal of 4 LSBs

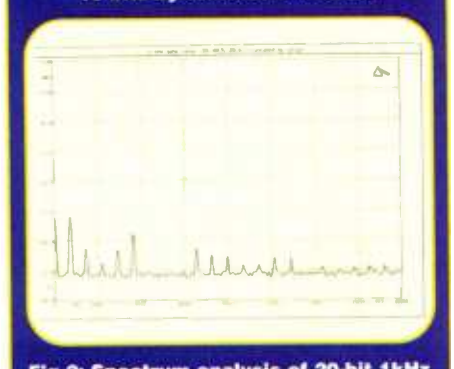

Fig.2: Spectrum analysis of 20-bit 1kHz tone at -90dBFS with TPDF dither

We used to send CDs to the pressing plant on U-matic-1630-format tapes, and nothing else. Now there are all sorts of options including CD-R and DDP Exabyte tapes. More than one person has claimed that CDs pressed from one type of master sound different to those pressed from another, or that CDs mastered at double speed sound different to those mastered at normal speed. Are they talking rubbish? Clearly if a record company thinks it is worth paying a mastering engineer a lot of money to get the sound just right, they don't want all the hard work to be ruined by the replication process. But we thought that digital replication was a transparent process, didn't we? Has the bottom fallen out of our digital world?

Now, I'm as sceptical as the best of them when it comes to subjective claims that appear to have no foundation in theory, but I have been sufficiently persuaded that there is an issue to look at further. The effects we are talking about must be described as subtle, although those whose business is listening to subtle effects may describe them as glaring or obvious.

INCREASINGLY IT IS necessary to deal with recordings made at resolutions greater than 16 bits, especially in the classical field. Also in the future one may have to deal with recordings made at all sorts of higher rates and resolutions, especially if 'one bit', high sampling-rate recording systems become popular.

There is no escaping the fact that CD is a 16-bit medium, and so high-resolution recordings have to be reduced to 16 bits during mastering. There are various ways of doing this, some theoretically 'correct', some less so (you might be surprised what goes on). Theoretically one should not simply truncate high-resolution samples, removing least significant bits to shorten the word-length, because this produces distortion. **Fig.1** shows a spectrum analysis, performed in the digital domain, of a 1kHz sine wave at a level of -90dBFS. This was generated at 20-bit resolution with appropriate TPDF dither and you can see just the 1kHz component with a flat noise-floor lying just at the bottom of the graph. **Fig.2** shows the same signal truncated to 16-bit resolution (by simply removing the 4 LSBs). The result is the addition of a number of low level distortion components, both odd and even harmonic. This is what would happen if a 20-bit recording was simply copied to a 16-bit format without any intermediate processing.

Normally, what is required is some redithering of the signal to remove this distortion. This should be applied at the correct level for the target resolution, but before the signal is requantised, as shown in **Fig.3**. This has the effect of increasing the noise floor slightly. Interestingly, though, I have come across cases where people actually prefer the sound of the truncated version to that of the redithered version, which, perhaps, seems surprising. Is it though? Some of us have the possibly naive belief that there should be a 'pure' signal chain from source to end product, but in many cases this is not what people want to hear. As we are only too aware, people actually like certain types of distortion. Why else would particular mics, preamps, A–D convertors be used for the sound they produce, if not because they introduce some desirable distortion? We are deluding ourselves if we think that people always choose equipment for its sonic purity. They may think they do, but in fact they may simply have a preference for one kind of distortion over another.

This extends to the mastering process as well. Recordings may be passed through whatever devices, analogue or digital, are needed to give the 'right' sound, even if this means doing what technically seems like the wrong thing (in other words, affecting the 'purity' of the signal chain).

Most people will be aware of relatively recent systems such as Sony's Super Bit Mapping (SBM) which act to improve the ☞

perceived dynamic range of CDs. These processes are used during mastering to shape the noise floor so that it sounds quieter, and are usually performed on high-resolution recordings so that the dynamic range of, say, a 20-bit recording can be as closely approached as possible on the 16-bit CD. What happens is that the signal is processed during requantisation and redithering so that the spectrum of the noise is not flat but shaped roughly according to the sensitivity of human hearing. So the noise is reduced in the middle frequency region where the ear is most sensitive, at the expense of increased noise at frequencies where it is less sensitive, giving the impression of a lower noise floor than with a flat noise spectrum. Fig.4 shows some examples of noise-shaping curves available from the Meridian 518 mastering processor. The more complex curves involve higher-order digital filters and result in a greater degree of perceived noise reduction. There are also curves optimised for different listening conditions, known as Minimum Audible Pressure (MAP) and Minimum Audible Field (MAF), MAP being more suitable for headphone listening and MAF for loudspeakers.

Whereas purists might assume that there is only one 'correct' noise shaping curve for a certain situation, in fact mastering engineers use them as a creative tool, choosing whichever one sounds nicest on the album they are mastering. When I visited DG in Hanover, they were experimenting with a very wide range of different 'flavours' of noise to be used in their Authentic Bit Imaging (ABI) processor, finding that the engineers liked to have a wide choice of possibilities rather than just one or two 'correct' curves. Clearly flavours of noise are just as important as different flavours of distortion when it comes to personal preference.

ALL THESE ISSUES can be found to apply just as readily in the remastering of analogue recordings for CD. There is still a very strong view in the industry that analogue recordings contain information which current digital systems do not adequately convey. This may either be in terms of bandwidth (if you believe that spectral content above 20kHz is audible) or other factors such as dynamic range and inner detail (which can be used to describe all sorts of low-level effects). Now clearly it is hard to argue that the noise floor of most analogue recordings is lower than that of a 16-bit digital system, but it is often proposed that analogue recordings preserve inner detail and spatial effects that are not always preserved when remastering the recording digitally. I have found this quite hard to accept, and my mind has conjured up pictures of Emperors and New Clothes, but there are 16-bit convertors and 16-bit convertors. Convertor linearity (or lack of it) and other effects such as timing stability (or lack of it) clearly have a significant effect on sound quality, and the correct choice of convertor is of paramount importance, as is ensuring that a stable enough clock reference is used when externally locking systems. A correctly dithered 16-bit convertor is capable of conveying information down to very low levels, below

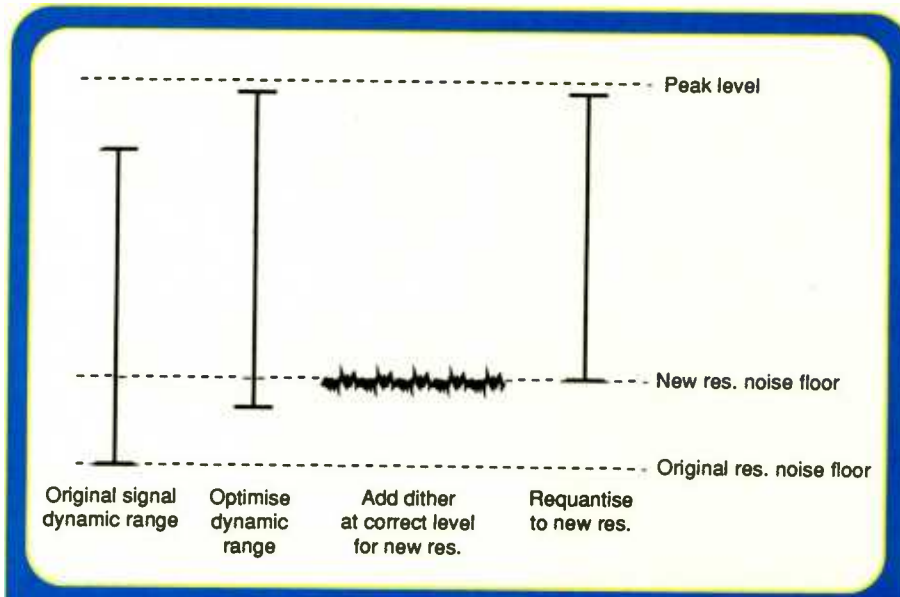

Fig.3: Redithering and requantising high-bit audio

the -96dB that many often quote as the dynamic range limit. Fig.5 illustrates the waveform and spectrum, analysed in the digital domain, of a 1kHz sine wave at -104dBFS, generated digitally at 16-bit resolution and with ±1 LSB TPDF dither. Clearly, the information about low-level signals is able to be preserved by a 16-bit digital signal-chain provided that dither is properly implemented.

Even though it may be necessary to accept that technically analogue tape recorders have poorer distortion and noise specifications than modern A–D convertors, the digital remastering process is concerned with representing the sound of the analogue master as accurately as possible. In other words the digital system should not add undesirable artefacts of its own to the sound, even if they are much less significant than the distortion already present on the analogue master. The fact that an analogue tape recording has more distortion of just about all kinds than the digital system used to remaster it is no excuse for cutting corners on the digital system. Many are now using the best 20-bit convertors and noise shapers to remaster analogue tapes for 16-bit CD, and believe strongly that the result is the better for it. There is simply no point in arguing about specifications and orders of magnitude of difference in effect if someone likes the sound achieved by a certain approach.

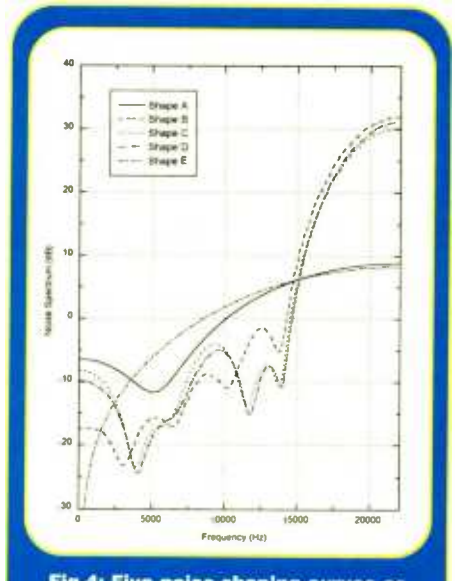

Fig.4: Five noise-shaping curves as applied by the Meridian 518 processor

The bandwidth issue is a difficult one. Certainly a good analogue tape recorder has a frequency response that extends above 20kHz, and often up to well over 30kHz, whereas standard sampling rate digital equipment removes components above 20kHz to avoid aliasing effects. Performed well, using oversampling convertors and digital filtering, this process can be as benign as possible in terms of side effects. Even so, there is still a feeling among a number of engineers and listeners that very high frequency components make a subtle difference to sound quality, and various people have tried to demonstrate this, including a Japanese professor who has tried to show that they have a relaxing and pleasing effect on the brain. We are dealing here with effects that are orders of magnitude smaller in their significance than the types of effects that we have been used to dealing with, but they may nonetheless exist.

We should perhaps be more concerned with the probability that people will notice certain effects than with trying to prove conclusively that they are or are not audible. If listening tests show that there is only a 5% probability that trained listeners can tell the difference, then how important is it? The reason for the lack of properly controlled blind tests in this area is obvious, because they are difficult and expensive to conduct, and in any case there would probably be little purpose except to settle arguments and allow us to score points off each other. If a process is thought to make enough difference to make it worth using, then, provided the marketing people can sell it to the punters, it has to be a good thing.

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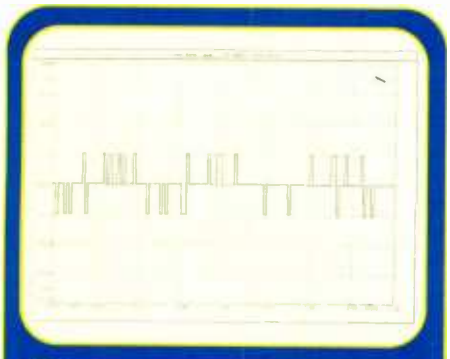

Fig.5a: Spectrum analysis of 1kHz 16-bit sine wave with ±1LSB TPDF dither

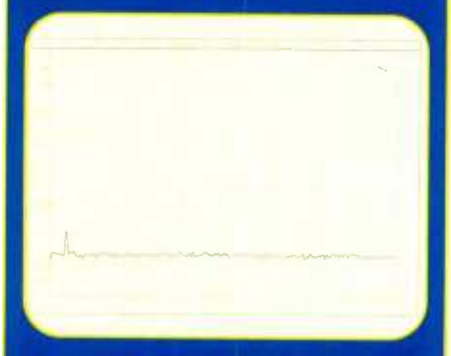

Fig.5b: Waveform of 1kHz 16-bit sine wave with ±1LSB TPDF dither at -104dBFS

Where is this leading? It is to the thorny question of what happens to a master when it goes to be turned into CDs. Up until this point one can still feel as if one has some control over what is happening to the signal, but many seem to be concerned about what happens later. If we are dealing with the preservation and control of very subtle effects in the mastering process, then presumably it is expected that these will be passed on to the consumer, otherwise what is the point of taking any trouble? Yet there are a lot of rumblings about CDs sounding different to the master tapes which formed them. What then is the meaning of this, and what are the likely causes? I think we must accept that it happens, and that audible differences do arise, but that it depends very much on the CD player and D–A convertor used to replay the disc. It is the D–A convertor at the end of the chain that governs the replayed sound quality.

Firstly, some facts about digital signal chains (because despite all this subjectivism there are definitely some hard and fast rules). The A–D convertor used at the beginning of a signal chain governs the baseline sound quality achievable in the rest of the system. After A–D conversion the sound quality can only be made worse. Artefacts introduced by poor A–D conversion become inherent in the digitised signal and cannot be removed. This then becomes part of the information contained in the digital signal. After conversion there are only really two things that could be said to affect the 'quality' of a digital signal. One is changes to individual sample values and the other is changes in the sample timing. These manifest themselves in an audible fashion only when the signal is converted back to analogue by a D–A convertor. In looking at the issue of replication and replay we are therefore concerned with whether or not the process affects either of these two parameters.

The replication process is entirely 'digital' in the sense that the binary data from the master recording is used directly to derive the binary data that is recorded on the CD. There are no signal processing or analogue stages in the chain that act to modify the sample values. People get worried about the process of 8–14 modulation used in coding the data for recording on CD, but this is a transparent form of channel coding and simply maps the original data values to specific longer values for recording. On replay the coded values are turned back into exactly the same binary values that existed on the master tape. Unless this was so, CD-ROMs would not be possible, since they also use the same form of channel code. If the CD coding process changed the data values in any way CD-ROMs would be useless for storing computer data such as databases and text!

THE ONLY OTHER WAY that sample values can be changed is by errors, but replication errors are normally extremely rare and replicated CDs compare byte-for-byte with the original master, using computer verification. Again, unless this was so, CD-ROMs would not be possible. CD-R masters vary in their error rates, but most errors are fully correctable in the glass mastering process, and the master should normally be rejected if uncorrectable errors are encountered. So the data on replicated discs is almost certainly okay. There is, though, the strong possibility that replay errors may arise in the consumer's CD player, resulting from problems with tracking, poor transport and servo design, dirt and so on. Initial evidence suggests that there may indeed be wide variations between CD players in this respect, and that interpolation errors may be more common than we had at first thought. CD players resort to interpolation when they cannot fully correct an error, and this results in a temporary loss of high frequency signal (a reduction in bandwidth). CD-ROMs, on the other hand, have an extra layer of error correction and the possibility to reread blocks of data if a replay error arises, which means that uncorrectable errors are less likely to be encountered (which is indeed our experience).

What, then, about the possibility of timing errors during D–A conversion, somehow arising because of timing irregularities in the data stored on the CD? The great thing about digital audio is that timing irregularities (jitter) in the data stream can be reduced before the signal is converted back to analogue. Jitter in itself is not necessarily a problem, provided the signal remains in the digital domain and can be communicated successfully; it only becomes a problem if it is not removed before conversion back to analogue. If it is not then it can cause distortion and intermodulation effects in the converted audio signal. Indeed anything which affects the stability of the D–A convertor clock can cause problems with sound quality.

CD players vary widely in the quality of their design and construction. They use D–A convertors ranging from the poor to the exceptionally good (probably better than many professional convertors). All sorts of factors can affect the stability of the convertor clock in a CD player, and there are cases where interaction between parts of the player results in clock modulation due to power supply and transport irregularities. Technical purists would say that this is simply bad design, and I would agree, but, of course, there are an infinite number of degrees of 'bad design' which extend all the way to good design. It is unlikely that even the best designed CD players can totally isolate one clock signal from another, although one can do one's best to minimise interaction. Indeed this is an analogue design problem, not a digital one! If this is the case then it cannot be beyond the realms of possibility that timing irregularities in the data stored on the disc might also affect the timing of the D–A convertor clock, thereby resulting in sound quality impairment. I am also prepared to believe that different methods of cutting the glass master can result in different degrees of data jitter in the stored information, and that it might not be possible to reject this completely in decoding and D–A conversion. Clearly, a good CD player would do a better job than a bad one at rejecting such artefacts, but there is no such thing as a perfect CD player, nor a perfect D–A convertor. We must remember, though, that the magnitude of the effects we are talking about is extremely small in most cases, and should not be blown up out of proportion.

It has also been suggested to me that a by-product of 8–14 modulation may have something to do with the differences in sound between CDs and master tapes. Three packing bits are used between the 14-bit words of the channel code, which serve the purpose of minimising its DC content. Apparently, 8–14-bit modulators in glass mastering equipment do not all behave in the same way with respect to these packing bits, resulting in different variations in the long term DC value of the code. Although this does not change the reconstructed data values (as previously explained) it has been suggested that the changes in DC code content might result in some clock modulation of the replaying CD player. Again, in purely digital terms this is something which can be ignored, because a channel code is just a way of matching data to the storage medium, but I am not prepared to discount the possibility that some CD players might not sufficiently be able to separate 'analogue' and 'digital' parts so as to make the effects inaudible. Experiments are currently going on in various quarters to determine whether or not any of these effects are real, and what their magnitude might be. The results should make interesting reading, although if they show differences they will really only point to limitations in the CD players used to replay the discs.

The upshot is that although the mastering engineer can do his best to control the way a CD sounds, some of it may be out of his hands. It ultimately depends on the ability of the consumer CD player to reproduce accurately the information represented in the stored data, and this implies that it should have a perfectly stable convertor clock, unaffected by any external influences, a perfectly linear convertor and an ideal reconstruction filter. If you find one, let me know.

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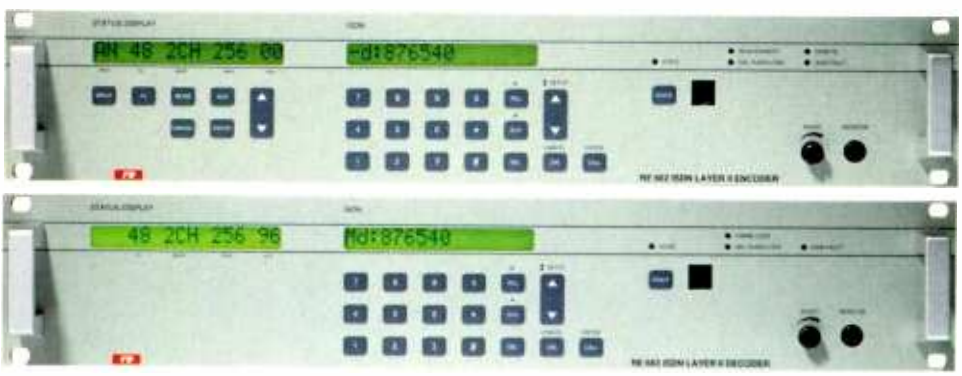

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Rockin' with the XT

Aerosmith reckon to have blurred the distinction between the recording studio and the hotel room. **DAN DALEY** tracks the recording of the band's forthcoming studio album on Alesis' ADAT XT

THE SOUTH FLORIDA sun is shining intensely on the portico of the Marlin Hotel in Miami's trendy South Beach neighbourhood. Francis Buckley is sitting beneath an umbrella in round, mirrored, sunglasses, talking equally intensely about audio gear and the parade of tanned skin and blonde hair that is South Beach. But we know that the LA-based Detroit engineer gets to see the sun for about an hour a day, if he's lucky, before heading into either South Beach Studios in the hotel or into Criteria Recording where he's maintaining his studio pallor as he works on the next Aerosmith album with Producer Glen Ballard (who swept this year's Grammys with Alanis Morissette), and fellow engineer Chris Fogel.

This high-profile project is based around a once-humble technology—the newest edition of the modular digital multitrack ADAT, the XT. In the process, the machine is redefining not only how music is recorded but how it is created. Describing himself as a 'garden-variety engineer' who has done well by putting music before technology, Buckley has worked off and on with Ballard since 1981, when he was running MCA's publishing studio in LA and Ballard was a staff songwriter. Buckley was also an early devotee of the ADAT format, and has been a long-time beta-tester for Alesis, using the ADAT format for much of his work on Quincy Jones' acclaimed *Q's Juke Joint* in 1995. (The ADAT was also extensively used in the making of Morissette's multiplatinum *Jagged Little Pill*.)

'The engineer makes the choice [of formats] on a record,' says Buckley, stroking his fuzzy blonde Van Dyke. 'But for me, it's not the medium, it's what you record that makes a difference. I've chosen the ADATs because they sound great and they give me the ease of use and flexibility to get to the music and not waste time on dealing with the technology.'

Buckley's methodology has helped move the format into higher artistic and commercial realms, however. Aerosmith producer Glen Ballard had four ADAT XTs, as well as a Mackie 32-bus console, Genelec 1031A monitors and various keyboards and outboard gear, in his suite at the Marlin Hotel. This rig was used by he and Aerosmith to write songs for the new record, a process begun last January. Eight more XTs were at Criteria Recording, five of which were for the basic recordings and three for comping.

'They would take the basic ideas for the songs and start to lay them down in the hotel room,' explains Buckley. 'Because of the quality, they would actually go for final parts in many cases, like guitar solos, right in the hotel room. We'd lay the parts down as though this were the studio session—kick, snare, hat on separate tracks. Then we'd mix from three of the [hotel room] ADATs to the fourth, putting the drums to mono, doing comp tracks of the guitars, keyboards, vocals and so on, with a click track on track eight. Then we'd take that tape to the studio. The band cuts all together, so what they'd be doing is actually tracking to that tape made in the hotel room, using it as a guide track which was fed to each person in the band a customised mix from the ADAT. That's why the click track is so important—it is the greatest synchronisation reference you'll ever have. The front panel of a synchroniser may tell you that you have lock, but there's always a chance for there to be some drift. Even on large-format digital machines, their SMPTE may show lock but if their 48kHz clocks are out of sync they'll start to walk away from each other. The sample rate is 16 times more accurate than the subframe [rate]. One way to test this is that if you take the click outputs of two machines and bring them up on separate faders and put the console in mono, they will phase-cancel each other out if they're in perfect sync. If they're not, ☞

Above: Engineer Francis Buckley (background) operates the Alesis XT, 8-track, digital-audio recorder at South Beach Studios in Miami, Florida during a session for Aerosmith's latest album. This is the first major project for the new Alesis-XT. Also pictured is Engineer Christopher Fogel operating a Solid State Logic 72-input SL 4000 Series console. Yamaha NS10 monitors are also pictured.

PHOTOGRAPH BY STEPHEN M LIPSON

you'll know it, even if the SMPTE read-out says they're locked. That's how I do a click reference, and after I do that I listen to the click reference from top to bottom. If you're dealing with multiple tapes, whether they're from a hotel room a few miles away or from other countries thousands of miles away, there's always the possibility of SMPTE information damage and drift. The click track will save you every time.'

Buckley does use SMPTE, however, as a secondary reference, and believes that the ADAT BRC remote control's ability to generate and read SMPTE onto subcode tracks continuously was useful in aligning click tracks from various source tapes. 'Once I find the same spot on each tape via the click track, I calculate the time difference between each machine, set the offsets and then rerecord the SMPTE tracks. The SMPTE can be moved around after the fact.'

Aerosmith would track to the hotel room prerecords (Buckley's neologism for what he regards as the dated term 'demos') to several XTs. Back in the studio, six XTs are running. Deck one is the prerecord mix; deck two and three hold the drums, deck four the bass and guitars, deck five with Steven Tyler's vocals. The same information is also recorded to a Studer A827 analogue multitrack deck.

'Even though I love the way the ADATs sound, the A827 gives the band and Glen the option of using the analogue tracks for individual instruments later on,' he says. The band's usual preference is for drums off the ADATs and guitars from the analogue source. They would take several passes at each song (17 were recorded on the entire project, including B-sides for release separate from the still-untitled album). On each pass, the prerecord ADAT is rewound to the start of the hotel room version; and the tracking ADATs offset six minutes from the start of the previous take. This is in order to have a common and simple location reference for each take on each tracking machine. Buckley says the BRC's ability to assign different offsets to different machines under its control was especially useful for this purpose. The analogue deck is started and stopped in place. The practice is also useful for anticipated crossfade splices of entire sections of tracks with each other.

'Let's say Glen likes the drums on the first verse of take one and the chorus drums from take four and the rest of the drums from take three,' he says. 'Since all the takes are playing to the same musical reference—the click, as opposed to SMPTE—we can slice and dice from the ADATs and do all that in like four minutes by comping over from the source machines to another ADAT.'

THE XT HAS numerous enhancements and new features over the original ADAT. When the Aerosmith sessions began on older ADATs—now everything is being done on the XT version. The issue of compatibility between old and new is moot in this instance. The changes that Buckley says contribute most significantly to the Aerosmith project are the XT's faster (4 times that of the original ADAT) shuttle speed, new locating capability, and improved A–D conversion (and the new spec of 128 times oversampling). The XT's faster shuttle speed and its newly acquired ability to set up to ten location points from the front panel of each machine help in setting offsets off-line from the BRC.

'You have a more functional system if you're not using the BRC,' Buckley says. 'Now you can also change the sampling rate and the clocking functions from the front panel. It improves overall locking of multiple machines.'

And with the enhanced location-storage capability of the XT, he recommends that location points be used to move from place to place on the tape rather than using the REWIND and FF buttons.

'The thing is very fast now, and using the rewind and fast-forward controls doesn't give the accuracy,' he explains. 'And that also contributes to burning out the transports. In fact, using the location memory on any ADAT will help extend the life and reliability of the transports.'

Buckley is also using the BRC in the Beats & Bars sync mode, the approach he usually favours. The BRC will perform all the offset calculations and you can keep the operation of the deck closer to a musical basis rather than a frame-based one.

Aerosmith's mixes will be done to half-inch analogue. As mentioned earlier, an analogue 24-track parallel recording will give Ballard and the band options when tracks in either domain right up until the final mix. The pattern throughout most of the project has been to take drums, bass, vocals and keyboards from the ADATs and guitars from the analogue tracks. (On the Quincy Jones project, Buckley would have stayed with the ADAT as the primary recording format except that engineer Bruce Swedien was called in to mix the project and Buckley want to save him from being 'lost in a sea of tapes,' so he transferred the source tracks to a Sony 3348 for mixing.)

For modular digital multitrack technology to have risen so far in the ranks of the recording hierarchy—for a high-profile band such as Aerosmith to rely on ADAT for most of what will be a highly anticipated album—speaks reams for the format's progress both technologically and perceptually in a relatively short time. More than simply a technological platform, the ADAT has become a methodology for Buckley.

'It's the format that the songs are conceived and written on, that the arrangements are experimented with, the tempos are played with, the solos and overdubs are tried and in some cases kept on right through the studio stage, and the primary format for the studio recording, right down to the final mixes,' he claims. 'It's become almost transparent to the entire process, something that 2-inch 24-track recording never did or could because of those machines' size and price. It addresses the fact that I don't want to get bogged down in how I'm going to do something; it just lets me do it.

'I don't see any reason to switch formats just because I'm moving from the writing environment to the recording and mixing one. Once I start writing the record, it already is the record. Thinking in this way really changes the nature of the way that we think about demos versus records. It makes what used to be the line between them now a very grey area.'

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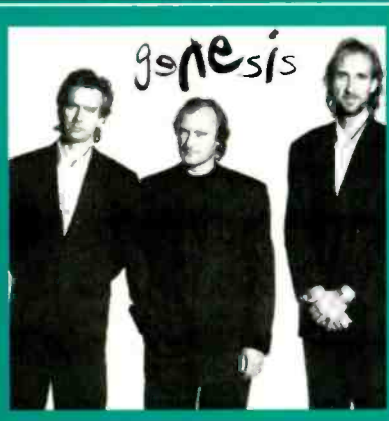

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Wonderful, wonderful Copenhagen

Despite the fact that the 100th AES Convention is primarily a professional sound display, radio broadcast is growing in presence **KEVIN HILTON**

It can hardly have escaped your notice that this month marks the 100th AES Convention. 'Try to work in the significance of that' I was asked, which left me pondering for a couple of minutes. Unfortunately, the only importance that immediately occurred to me was that 11th to 14th May 1996 marks the one hundredth time that audio professionals have wandered about badly laid-out convention centres wearing identity badges over their left breasts, carrying lots of plastic bags that eventually give way under the strain of collected brochures.

This is, of course, of shallow significance, designed merely as a vehicle for a couple of cheap gags at the expense of exhibitions.

What is really significant about this 1996 Convention is the host city—and not merely because Copenhagen is home to the near-legendary White Mouse and Elephant bar.

This is the first time the AES has come so far north, and to Scandinavia in particular, coinciding with Norway's historic hosting of the Eurovision Song Contest a couple of days later. The Norwegians have long been Eurovision fall-guys, having notched up more nil points than anyone, but last year they won, causing everyone to rethink all their old jokes and prompting unmitigated joy in Ireland, which was praying that it would lose.

Eurovision may be billed as a song contest but, as the continuing poor standard of entries shows, it isn't anything of the sort—it is really a vehicle for the EBU's laudable aim of uniting the continent of Europe. This is made possible by the simultaneous live broadcast of the event in all participating countries, making it a hugely involved annual TV and radio link-up.

Elements of the technology that will be used at Eurovision will be on display during the AES, including Sony's WL-300 wireless microphone system, along with other broadcast equipment that is being used extensively throughout Scandinavia. Which brings us to perhaps the most significant aspect of AES, one that has been developing in recent years.

In Studio Sound's report on the 1995 Paris Convention I remarked: 'At some turns it seemed more like a radio event than a general professional sound display.' That trend continues this year and takes in some important implementations of digital technology, particularly in the fields of servers and acquisition-management systems.

Stockholm-based Broadcasting Control & Communication is to launch the Broadcast Media Server (BMS) software for Windows, which is currently used in both Sweden and Norway, with NRK's Mostly Classical station being a wholly automated, digital service with a staff of only five. Back in Sweden, P6 Radio, which has stations in most of the country's major cities and towns, is using the US SMARTS Broadcast Systems 'hub and spoke' approach to create a self-contained, fully automatic radio network.

Other radio automation displays include DAVID's DigAs CartWall SQL database; On Air Digital with a variety of integrated automation radio systems; Dalet Digital Media Systems's four different control panels; and Audio Follow's Changeover Concept.

Oh, and there'll be some recording stuff there as well and probably even more DAWs, lots of plastic carriers spilling their contents all over the place and people spearing themselves with the pins on their identity badges. All of which is, of course, significant.

ON THE FACE OF IT CableTel's take-over of NTL is a deeply significant turn of events. Here is a US-owned company that is also the UK's third biggest cable operator buying Britain's leading TV and radio transmission infrastructure provider, an operation that has proved to be hugely successful since entering the private sector and has forced the issue on DTT.

An emotive element in all this is NTL's long-standing bid for the BBC's transmission division. The island mentality runs that if the UK Government does hive off the Beeb's transmitters to another company, which seems increasingly likely, then the means of distributing the Voice of the Nation will be in foreign hands. And American ones to boot.

While this conclusion is true in principle it is also a knee-jerk reaction. The wider view is that although CableTel is indeed a US concern, it has deep roots in the UK; NTL itself will continue with its present structure, which already includes an American as CEO, something that went unremarked at the time.

It could be said that the transmission of ITV and Channel 4 (S4C in Wales) is already under Stateside control but that would be emotive and inaccurate. Let's face it, the majority of cable companies have some US or Canadian or French influence and the fabric of British society hasn't fallen apart because of it. In commenting on this latest development, Bruce Randall, Publicity Manager of NTL, also covers the wider situation by saying that it's just 'business as usual'.

Eurovision may be billed as a song contest but it isn't anything of the sort—it is really a vehicle for the EBU's laudable aim of uniting the continent of Europe

In the specific case of NTL, a long-term owner was always the intention as the company's takeover by a venture capital operation after privatisation was always the intention. If we're going to resurrect the observation by the Earl of Stockton (former PM Harold Macmillan) that it's akin to selling off the family silver, then it should be directed again at its original target, Margaret Thatcher, who used the 1990 Broadcasting Act to dismantle one of her *bête noirs*, ITV. It is a situation we've been living with for six years and if an American company can come in and make it work, we shouldn't circle the wagons. Unless it's a deal that involves Rupert Murdoch, of course.

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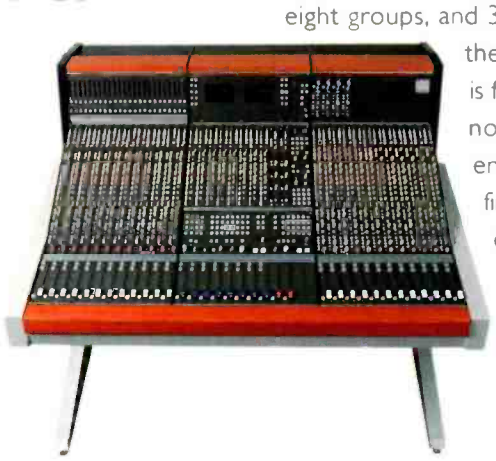

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The monochrome set

Progressive production ideas flood the set of one of Britain's best music television shows for years. **TIM GOODYER** went behind the scenes at *The White Room* during rehearsals to find out more

INSIDE THE STUDIO the set is white throughout, with no concessions made to backline, monitoring or even the show's own logo. The dress code is black and white –and as strictly enforced as it is simple. On first appraisal it would be easy to think *The White Room* just another token music show whose priorities lie with visuals rather than music. Yet all the acts play live and the show boasts an team of seasoned sound men and uncompromised sound systems for both monitoring and recording.

With the only other worthwhile British music programme–Jools Holland's *Later* –now into its seventh series on BBC TV, Channel 4's *The White Room* is certainly the junior partner. The show I saw recorded was the penultimate of its second series, although the third is already confirmed for the Autumn and a is hot topic of conversation on the set–closely following that of next week's 'wrap party'. Having suffered working titles such as *Shed* and *Trash, The White Room* is an independent production from Initial, and the work of some of the same team who have previously had a hand in the innovative British music series *The Tube* and the enviably successful *Rock Steady*.

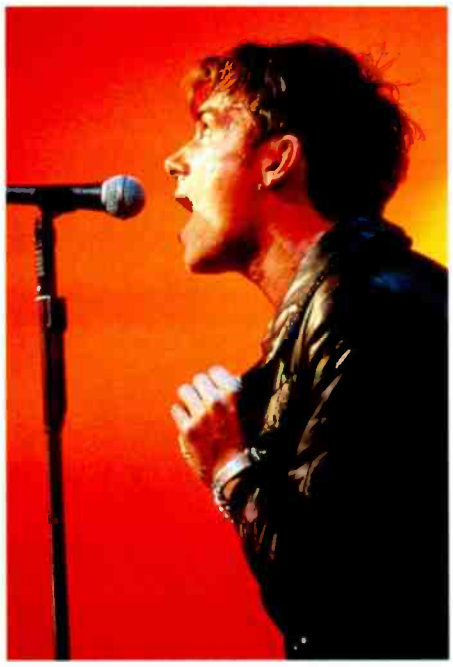
In common with *Later, The White Room* plays heavily on its use of five or so artists per show and the personality of its presenter

–in this case the wry observational humour of cult radio DJ Mark Radcliffe. At the glossy end of the musical scale, the show has seen performances from the likes of [The Artist Formerly Known as] Prince, Sting, Lou Reed, and David Bowie while acts such as Tricky and Dubstar have lent the show a pleasingly contemporary note. But perhaps most interestingly, the producers have arranged collaborations between artists with no recorded precedent such as that between The Kinks' Ray Davies and Blur's Damon Albarn. Today the pairing is between Terry Hall and Tricky.

SITTING IN ON Nick Cave's midday rehearsal at London's Westway Television Studios–where the series has been recorded –I was able to take a closer look at the set itself. Four stages occupy the corners of the room with the PA rig flown from three points between. The backline amplification for each stage is hidden behind white grilles similar to those that hide the floor monitors along the front. The brief here is not to short cut the sound in the interests of the set design but to work with it. There are a number of platforms (white, naturally) for presenting the programme links and one of the stages also accommodates the show's four dancers. Close up, the studio bears the scars of the passing musicians and their equipment but kindly camera work ensures that this is bleached out of the broadcast picture.

Tucked around the outside of the set on one side are the lighting consoles and on the other, the four Midas XL3s which handle the live sound–the latter area having being dubbed Monitor City by the production crew. In charge of the sound here is Clive Trimby, veteran of such shows as *The Tube, Wired* and *Big World Café*. And his experience shows: there is little of the battlefield feel of many backstage bunkers here. Instead the desks are neatly lined and clearly marked. Adjacent racks contain

16 tc electronic 1128 third-octave graphics, the PA's TOA SAORI crossover, Crest and Crown 2401 amps, AKG C5900 wireless mic systems, and five in-ear monitor bases (one mic-monitor pair is used by floor manager Simon Hodge). Above each of the desks sit video monitors showing the director's pictures and a spy picture, along with Yamaha MS202 speakers for audio monitoring. Significantly, the sound here is neither loud



nor unintelligible. On top of the outboard rack sits the 'simple but sophisticated' SSE patching system and BSS splitters that link the stage system to the OB vehicle. The key to the setup is what Trimby calls 'the grid' through which each act has access to an assigned 40-channel monitor desk and control of the graphic rack through the tc electronic remote with which up to 99 individual EQ settings can be stored and recalled.

On the set, the sound system uses 6kW of Crest amplification driving three identical flown rigs handling the mids and highs, and floor bass cabinets. Glossing over the details of the rig itself, Trimby is at pains to point out that the rig has been time corrected to the centre of the studio. 'The difference in quite noticeable,' he asserts. 'Obviously it can't be corrected for everywhere at once so we chose the centre of the room as the best compromise. So far I think only one person has been at all unhappy with the sound on the set.'

Trimby is also anxious to explain that he feels the spirit of cooperation between all aspects of the audio staff has been instrumental in making the the show a success not just with its viewers but with the artists. As evidence he reels of the list of headline acts which have happily overrun their four-song spot, '...Sting did seven songs, Bowie played for over an hour...' His sentiment is later echoed by numerous other members of the team.

The artists I saw were pretty relaxed—Nick Cave was playing the gentleman to PJ Harvey's lady and even Iggy Pop was obligingly putting in his time at the afternoon rehearsal.

Clockwise from top left: *les enfant terribles*, the brothers Gallagher, for once doing what they're best at; Blur's Damon Albarn is equally unconvincing in angelic mode; Little Richard's Madame Tussaud's replica is rolled in to tinkle the ivories, while Supergrass' Gaz is totally at home with his vintage Burns guitar

But it hasn't always been that easy, as OB Sound Supervisor Andy Rose relates. 'Prince just came in and played without any sound check. His band had played during the afternoon but didn't sound anything like they did with Prince. Much the same thing happened with Little Richard.'

Regardless of the complications, Rose, like Trimby, felt the support was largely responsible for the performances being 'absolutely stonking'.

Rose's centre of operations is the BBC's MSC3 OB sound vehicle parked outside the studio. Two 32-way multicores split the audio off from Monitor World to the audio MSC3 and TV (Chrysalis OB4) vehicles. These are changed over between acts at the monitor end and ensure that there is independent working between the monitor and broadcast sides. For the audio operation, the tie line numbers form part of the setup of the MSC3's Calrec Assignable console (see sidebar).

Inside the audio truck—hired from the BBC in its latterday commercial guise—Andy Rose works with 'resident' engineer Simon Scrivener on the broadcast sound. The 32 inputs from the studio are grouped into 22 record outputs and normalled to the inputs of a Sony DASH 3324S along with a uncompressed stereo 'audience reaction' mix. The 3324's outputs are then returned to the console as the stereo mix feeds. The outputs of the track returns are grouped and fed through a pair of Summit Audio TLA100s and an SPL Vitalizer, and routed to ☞

MSC3'S CALREC ASSIGNABLE

NOW NINE YEARS OLD, the Calrec Assignable console installed in MSC3 was the second such desk ordered. At the time, it was undoubtedly the most powerful broadcast desk installed in a mobile for the sole purpose of music balancing.

It has 112 inputs available at mic, line or tape return status. These can be assigned to 12 stereo mix buses (groups) and-or to one or all of four main outputs in order to cope with the broadcast requirements of specified mixes—for example, clean band, clean presenters or non-dynamically-controlled music feeds. There are 72 nominated channel faders, 12 nominated group faders and four nominated echo returns. The term 'nominated' indicates how the desk differs from most other mobile consoles.

The console surface does not handle audio. There is an apparatus room in MSC3 which accepts and processes audio according to instructions from the console—thus the surface is digital in operation but analogue in output like the Euphonix CS2000. On inspection, the surface bears little resemblance to the traditional concept because the channel strip lacks the usual format of gain, EQ, aux sends, pan and fader. The fader still exists as does a pot which may be used for panning if so assigned, but the similarity stops here.

At the hub of the console are two Assign panels which live either side of the comprehensive broadcast monitoring panel. These are identical and contain four sections:

The input section covers gain, pans (mix and record send), phase-leg, insert enables (pre-EQ, pre/post-fader), channel-fader assignment.

EQ is 4-band with variable high-pass and low-pass shelf/boost with individual defeat switching. The low and high mids are dealt with a 4-shape bell variable from notch to very wide, and a variable Q control with boost-cut of ±15dB. Defeat switching is also available for the mid range. There are also very steep high-pass and low-pass filters for source cleanliness which is always a problem in the broadcast environment.

Dynamics are available on each channel and consist of comp-limiters and gate-expanders with link buses available at various insert points. Gain reduction-compression meters are sited at the top of each fader to show levels of activity or status.

Eight auxillaries sends are provided per channel with four dedicated returns and LED bar graphs on master sends and returns.

The Assign panels are accessed by two large selectors positioned at the top of each fader. When both of these are touched—for example on fader 37—the displays indicating fader and channel in the Input section of the Assign panel show that fader 37 has been selected for adjustment and the audio associated with the channel number displayed in the channel window (which need not be the same as the fader number) will be affected by any changes made. Selecting different faders with each of the selectors allows simultaneous access to two channels; each fader can be mono, stereo or a VCA master.

At power up, it is possible to start with a completely unassigned surface which can be tailored to an individual Sound Supervisor's requirements. This means that any fader can be assigned to any input channel bringing a particularly ergonomic approach to operating an 88-fader console.

A complete desk setup—including all routing (record bus or stereo mixes), dynamics, EQ, auxes, inputs and fader assignments—can be stored on a PC which monitors the desk status and recalled at a button push. Every facility is returned to its stored operational state except for the faders. The audio level of the faders is returned but level control cannot be effected without moving a fader through a indicated null point.

Quick turnarounds can be made in a live situation. Further, there is the facility to make the fader-channel setting absolute such that their status is not altered during a memory flip—shows can therefore be worked around a central desk configuration. These facilities are, in part, what enabled The White Room to run transparently within its production schedule.

Left. Criminal record: former Police bassist Sting single-handedly saves the world on *The White Room*. Below. Jeans means funds: Babylon Zoo's *Boy with the X-ray eyes*, considers his bank balance

three of the main outputs to give duplicate master outs. Additional outboard processing is available through MSC3's Lexicon 480L, Eventide H3000B, AMS S-DMX, Yamaha Rev7 and two SPX90 IIs and two Neve 33609 comp-limiters. Additional outboard processing is introduced here at the request, and under the control of, an act's own engineer.

In addition to the 3324S machine, two Tascam DA-88s and two Sony PCM7030 DAT machines are run in the mobile—the DA-88s are configured as a main and backup pair and fed the main stereo output from the console, an uncompressed stereo mix, uncompressed presenter's mics and uncompressed audience mix. The DATs are fed the main output again as a main and backup pair. Reference video from the picture vehicle is fed to each machine via video DAs and time code is also fed to all machines for subsequent production sync. Everything is recorded to 48kHz. In OB4, the main audio output goes to Analogue Betacam machines while main, mono clean audience and clean presenter feeds go to a Digital Betacam machine for off-line editing—the final sound being sourced primarily from the main DAT machine. Using this setup, any problems with the DAT mix can be rectified by recourse to backup tapes or ultimately the 3324S multitrack. Here the resettability of the Calrec console again offers considerable savings in time, money and consistence of the finished mix. According to Rose, the expectation from previous shows is that about 10% of the recording will need attention of some kind—and the timescale is tight as the show is broadcast around 24 hours after recording.

AS SOUND SUPERVISOR, Andy Rose enjoys a free hand with the audio production values. General practices such as adding reverb to vocals and harmoniser to guitar come naturally to him but the scope of *The White Room*'s artists guarantees that he will be meeting new challenges at each show.

'It's a wide taste range,' he confirms. 'It goes from something I wouldn't really want to listen to to things that are a dream to work with. But if you've got good musicianship, you're on your way regardless of musical style.'

Rose's first encounter with some of the music is at the sound check. In some cases this is a deliberate policy. 'To go out and listen to the album would give me preconceptions about how people should sound,' he explains. I listened to Moloko after the show and they were doing things on a completely different vibe on the CD to what they were doing live. I gave the live performance a much bigger sound than they had on record.'

This fresh attitude lends itself well to the impromptu pairing of artists. 'Some of them have never even met before the show,' Rose reveals, 'so there can be no preconceptions about how it should sound.'

It's easy to see that Rose has found a slot that suits his talent and temperament well. But it's taken a couple of series of *The White Room* to crack it, having taken on the post duties for the first series. 'I was up at 7 o'clock on Friday and didn't get to bed for 36 hours,' he recalls. Today at midnight, all the tapes will be couriered to London post facility t'c Audio Post Production for an all-night session.

Earlier recording arrangements were more demanding too; using the Voyager's Neve VR, for example, involved 15-minute changeovers between bands. The alternative option for previous series' and for this series was to use two mobiles and for Rose to rush between them. MSC3 with its Calrec Assignable, it seems, comfortably takes care of all the audio requirements and offers sufficient financial saving to allow Rose to have specified changes to the monitoring system.

'We has LS5/8s on the preparation day but by the second day I'd got them changed to ATC100As. I've also got a pair of LS3/5s in now so that I can monitor in a circle: big-small-mono, all the time.'

The saving was also instrumental in getting the Summit units onto the mix buses. Who says you can't have it all ways?

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Reelin' in the years

The fact that the Audio Engineering Society is now 50 years old makes its own comment on the current state of professional audio. But where, asks **CHRIS EDWARDS** will a further 50 years worth of progress take us?

Normally, when you look back at 50 years of advances in technology, the changes seem enormous. You see such inequalities as computers that took up the best part of a warehouse yet were less powerful than today's pocket calculators–but changes in audio recording technology, at the surface at least, look a lot smaller. Magnetic tape continues to rule supreme and many of the original principles of microphone, amplifier and mixer design remain intact.

The frequency-response curves and signal-to-noise ratios of modern systems are much better than the best available 50 years ago but the circuit designs are often quite similar. Some older techniques –particularly the use of valves–have even made a come back to help overcome the clinical sound of digital recording.

The valve turned out to be a surprisingly good fit for audio recording. The fact that it does not have a flat frequency response is its biggest subjective advantage and, luckily, it is relatively easy to make valves that have a frequency response that lead to what most people call a 'warmer' sound.

The big strides in design have been in the manufacturing techniques for both the components and the systems using them. The steady move to silicon and from discrete components to integrated circuits has made the designs much more stable and responsive. The problem facing analogue designers is that they have been unable to derive the cost and space benefits that digital designers have, simply because there are things needed for high-level analogue performance that simply cannot be achieved with just one chip of silicon. So, even now, there is quite an amount of discrete circuitry in the analogue section of any pro-audio equipment.

THE BIGGEST CHANGE in the last 50 years has been the move to digital recording–but this is far from wholesale. Some producers refuse to employ digital recorders because they do not like the sound of what is available from the digital world today. Digital's prime benefit is one of low cost over subjective quality: it does involve a compromise. The decision to select 48kHz sampling at 16-bit resolution was a balancing act between subjective performance and the cost of storage.

It is a process that loses audio information and, although no-one is quite sure how the human brain perceives frequencies above 20kHz, it is becoming clear that high frequencies do have an effect, perhaps indirectly by modulating lower frequencies in the main audio band.

As we move into the next century and the next 50 years of the AES, it is the conversion between analogue and digital to recapture the dynamics of analogue recording that will become one of the main areas of focus. Some of the battle lines are being drawn in pro-audio on the basis that whatever takes over the studio as the high-resolution recording standard will affect any high-resolution distribution format for the consumer. To get something that better resembles analogue, it is highly likely that we will end up using a system that samples in the range of 90kHz–120kHz at a resolution of between 20 bits and 24 bits.

But the process does not stop here. Physical modelling techniques using high-speed digital signal processing will begin to move into the grey area between analogue and digital recording where the valve has made its return. Although it becomes progressively more difficult to model analogue processes as you increase the accuracy, the fact that it will become cheaper to warm up signals using digital simulation will tend to drive digital processing back into this area. However, the imperfections in valves mean that you will probably be able to tell the difference, just about.

Where digital will tend to have an edge is in much more flexible dynamic EQ techniques with which you can selectively cut and enhance the characteristics of each channel to get a balance instead of working with the volume faders. This is where the valve simulations are likely to end up going, adding a little extra edge to the dominant signals once the EQ has been set.

And those signals will end up going to tape, albeit by a more indirect route. Tape still has major advantages in terms of capacity, portability and storage life, although some of the new synthetic binders can have their problems. Hard-disk recording cannot substitute for tape completely: it simply provides a more flexible medium for cutting and moving tracks. It is a poor medium for archival, simply because each disk costs at least ten times as much as tape.

Optical discs have potential for archiving and offer some of the benefits of hard-disk recording. But their capacities continue to lag those of digital tape and that is going to be a big problem as studios move to higher-resolution recording formats. The digital tape systems devised for computers and video are now being recast as high-end digital audio recording media. Where optical-disc storage is stuck in the 2.5Gb region, tape has jumped from 1Gb to 5Gb to 20Gb in the space of ten years and further jumps are promised.

Physical modelling techniques using digital signal processing will begin to move into the grey area between analogue and digital recording where the valve has made its return

Although the studio will become even more digitally focused than it is now, many of the principles that it will use will be those employed over the last 50 years. Instead of trying to be something new, that digital technology will always be striving to sound as natural as analogue.

Yamaha 02R

In the second part of his bench test, **SAM WISE** turns his attention to the performance of the console's equalisers, dynamics and effects facilities, as well as reaching his general conclusions

LAST MONTH we studied the 02R as a system, looking at the performance of the main signal chain, convertors and digital faders. This month, we will concentrate on the processing elements—equalisers, dynamics and effects.

Four-band parametric equalisers are provided on all mixdown inputs and the stereo output. All processing is in the digital domain, though the equaliser curves are generally designed to replicate typical analogue-mixer characteristics. Settings of Q (width of filter), F (frequency) and G (gain) are displayed on LED read-outs adjacent to their respective rotary encoders. Each of the four bands is selected by a push-button switch and set by this one set of encoders. Otherwise, operation follows tradition and works well, with only one minor quirk—the Q control alters filter bandwidth just like those on most analogue mixers; turn it clockwise and bandwidth increases, turn the other way and the filter narrows. However, the numeric display shows the value of Q which increases as the control is turned anticlockwise—turn 'up' the Q control, and the Q goes down. The control would really be better marked BW for bandwidth, since filter width does increase as the control is rotated clockwise.

This was probably the subject of debate in Yamaha during the design phase.

In addition to the LED read-outs, adjacent to the controls, the EQ can be called up to the central LCD graphics display which shows: all four bands at once, a graphic of the combined response, a channel clip meter which helps when equaliser boost may push the signal into clipping, the state of the EQ ON button and some other information.

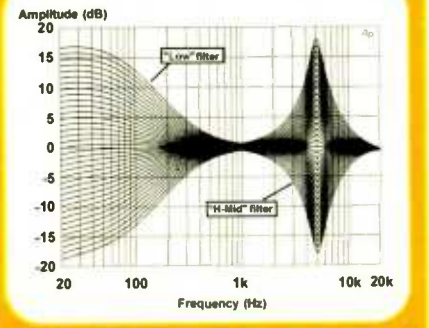

Fig.1: Equaliser action—the effect of G (gain control), with Q (width) and F (frequency) fixed. 02R gain steps are 0.5dB, 1.0dB shown here for reasons of clarity. Two bands are plotted together but were active separately. See fig.2 for interaction

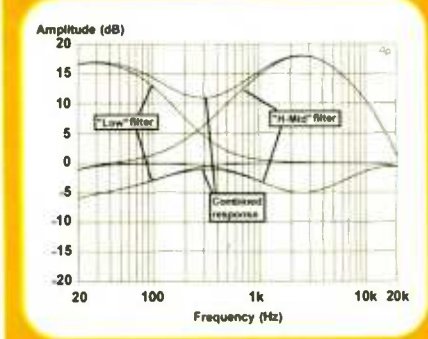

Fig.2: Equaliser action—combining of equaliser filters. Two bands are shown operating separately, then the result of the two operating simultaneously. Note that the total amplitude is equal to the sum of the two independent curves

Fig.1 shows the result of varying the gain control on independently operating high-mid and low filter sections. Though these are on the same graph, they were not operated simultaneously and therefore show none of their interaction. The LF filter is set to shelving mode at 99Hz, and the high-mid to a Q of 4 at 5.04kHz. Curves are shown at 1.0dB steps, though the equaliser minimum steps are 0.5dB. Personally, on a conventional mixer, I prefer an equaliser gain control law to be expanded in the middle—allowing easy setting of subtle EQ. The 02R's is perfectly logarithmic, giving equal 0.5dB steps over the full ±18dB range. But 0.5dB steps are small enough, and I'm happy operating this equaliser.

Having been a mixer designer in the past, these efforts impressed me. The accuracy of the filter steps is superb, and the system gives absolute repeatability and matching between channels—all of which is difficult to achieve with analogue circuitry. Considering voice recording applications for example, with a mixer like this you should no longer have to endure the variations often evident in edited speech. The recordist need only store the equaliser settings used, the mic serial number, and note the recording position. If further dubs are necessary, a practically exact replica of the vocal quality can be obtained. Nice.

In **Fig.2** we are looking at how the filters combine when used together. Here the LF filter is the same as in **Fig.1**, but the high-mid filter settings have been changed to 2.52kHz and a Q of 0.5. Both filters are set to a gain of +18dB (top curves) and then a cut of -5dB (bottom curves). Note the levels of the two curves in the zones where they overlap. The combined curve provides almost exactly the same boost as the dB sum of each curve taken separately.

This means that the equalisers are working as though they are a set of independent filters wired in series. The same holds true for any selection of filter bands within a channel. This type of architecture is common in high-quality analogue mixing consoles but in lower-priced models unwanted interaction often occurs. The graphic display of the combined equaliser response accurately depicts what the equaliser is doing.

Fig.3 shows the effect of varying Q while F and G are fixed. The Q range is 0.10–10.0 and step size is small enough, though also quick to operate. At full clockwise, the low and high filters assume a conventional shelving-filter shape which can be boost or cut, and fully anticlockwise 12dB/oct high or low-pass filters respectively ☞

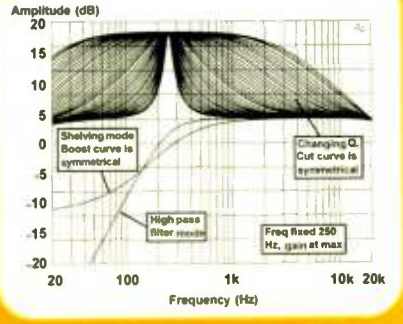

Fig.3: Equaliser action—effect of Q, with G and F fixed. Filter shown is the Low-HPF filter band. Q fully clockwise produces the shelving mode response, intermediate positions produce the bell-shaped curve, and fully anticlockwise produces the 12dB/oct high-pass-filter shape

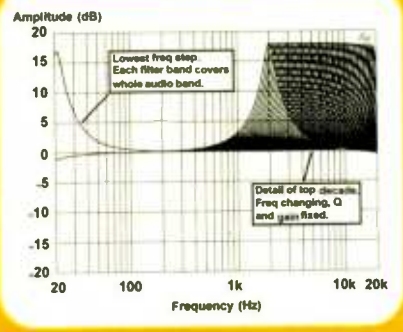

Fig.4: Equaliser action— effect of F, with G and Q fixed. All filter sections behave the same. The top decade is depicted in detail showing 40 frequency steps per decade. Each filter will cover the whole audio band. The lowest band is shown on the left

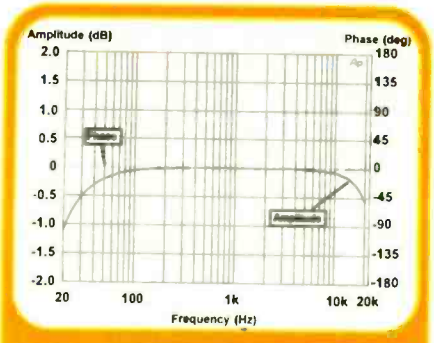

Fig.5: Equaliser action—combining of equaliser filters. This shows the amplitude and phase response of all four filters set to cancel one another. The result is as if no filters were in circuit at all

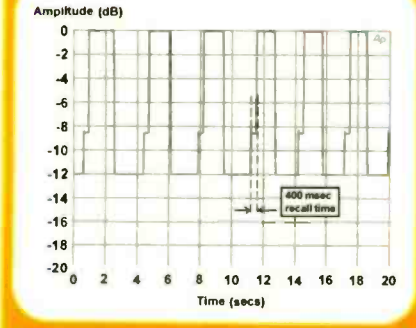

Fig.6: Equaliser Recall Test—this is a plot versus time of the amplitude of Channel 1 created by repeatedly recalling alternative presets. One has equaliser active, the other not. All other channels also had equaliser recalls, but faders were shut to prevent direct noise interference

where only the centre frequency can vary. This is a useful and effective range of facilities, though the maximum Q (minimum bandwidth) is a little wider than ideal for feedback control in live applications, or for removing a discrete tone which is unwanted without otherwise affecting the audio quality.

Fig.4 shows the frequency control range and precision of the High-LPF band of the equaliser. The curves on the right show the frequency step size indicating 40 steps over one decade. The curve on the left is the minimum frequency for this band. As you can see, each of the four bands covers the whole audio range in about 120 steps or 1/12-octave intervals. The resolution and range of these equalisers makes them excellent for almost any use.

As an experiment, all four filter bands in one channel were set to the same frequency and Q, with two at maximum gain and two at minimum gain. The result was a perfectly straight line amplitude and phase response as shown in Fig.5. As only digital circuitry can provide, this is a mathematically perfect cancellation of four filters. This also makes evident careful filter algorithm optimisation in terms of DSP power utilisation. A further experiment involved activating all four EQ bands on all possible inputs at the same time in order to detect any limitations on DSP power. Problems have been seen on other manufacturer's digital products when DSP systems were stressed. In this case, there are no shortcomings—everything works properly.

Finally in the equaliser section, we look at the dynamic action of the equalisers under automation control. Fig.6 shows the signal level of the left and right monitor outputs during an equaliser snapshot scene recall. In this setup, equalisers are active on the measured signal via both the input channels, and the stereo outputs. EQ settings are also recalled on all other channels and tape returns, though on these the faders remain shut so that they will not affect the audio signal directly. Note that equaliser reset took about 400ms to complete across the mixer. There are also some glitches in the response. Testing might have revealed a ramping function that kept the impact of a recall as unobtrusive as possible, but as expected, we are not able to recommend that major real-time recalls be made with outputs open. Although, to be fair, the signal remained within the envelop of the changes—in other words it went only from 0dB to +18dB and did not disappear into a peculiar status at any point as the recall occurred. Therefore leaving the mixer active during a recall is safe, tape will not be overloaded nor loudspeakers blown up, it is just that it may not be dependably audibly subtle.

For comparison, Fig.7 shows the replay of a real-time equaliser move, involving one channel equaliser with only the high-mid band active. The input signal is a 1kHz sinewave. The bottom curve starts with the equaliser at 0dB gain then the EQ is gradually reduced to -18dB, then up to the +18dB position and so on. The top curve depicts the effect of varying centre frequency with fixed narrow bandwidth and maximum gain. Both are smooth and glitch-free. The same is true of panning, or any other dynamic automation action. In this case the results are smooth and subtle, so you need have no fear of using equalisers during a mixdown or live performance.

Favourite equaliser settings can be stored for future use using the 02R library function, saving time when setting up for a recording session.

There being no mention of automation on delay, an attempt was made to record a sliding delay operation, which might be typical of a flanging effect. This was not recorded, though snapshots of delay changes are possible.

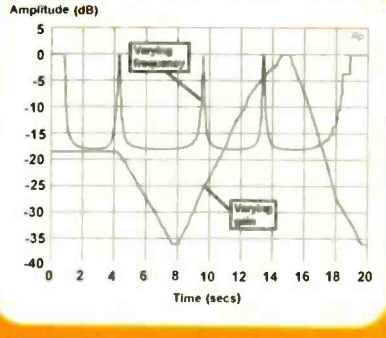

Fig.7: Equaliser real-time automation test—this is a plot versus time of the amplitude of Channel 1 created by real-time activation of two different equaliser controls. In the top curve the centre frequency of a Q of 10 filter is varied, in the bottom the gain is controlled between +18dB and -18dB. There are no obvious glitches

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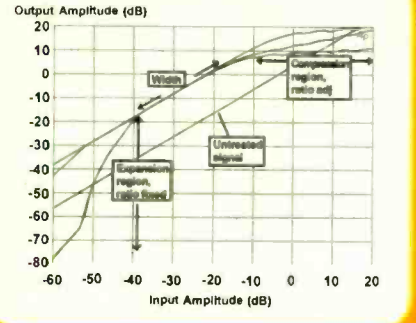

Fig.8: Dynamics test—compander action. Several possible setups are shown overlaid. Threshold is -20dB and -30dB, Ratio is 5:1 and 10:1,Width is 30dB, 40dB and 50dB

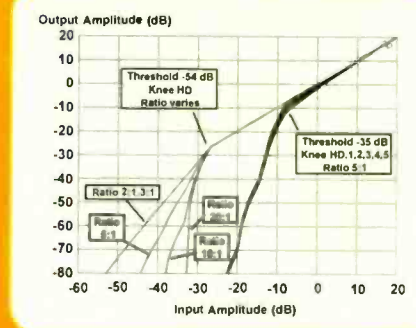

Fig.9: Dynamics test—expander action. Several possible setups are shown overlaid

DYNAMICS ARE ALSO extensive, with full recall of a snapshot of all parameters, but no real-time automation of the setting adjustments. As with the equaliser, each of the channels, tape returns, and stereo outputs are provided with full dynamics sections. Key (side chain or trigger) input sources on all dynamics modes are limited to self-keying pre-EQ or post-EQ or, if an external key is required, keying from the left adjacent channel pre-EQ or post-EQ, or the Aux 1 or Aux 2 path. Although this is obviously not as versatile as completely free patching of key inputs, it will allow most people to realise much of what they require.

The dynamics section offers several modes of operation: compression, gating, ducking, expanding and companding. These are selected via the Dynamics Library page, complete with preset adjustments which the user can alter and save to additional library memories. Each offers slightly different control functions with up to six adjustments available per dynamics device. As a test, a dynamics unit was selected for every possible channel, tape return and stereo output. All assignments were accepted and though it was not possible to check the dynamic response with all inputs simultaneously driven, it can be assumed that all of them will work at the same time. This is as it should be.

Fig.8 shows some typical compander curves using a steady state 1kHz tone swept in level. Hard-knee types are shown and compared with an untreated signal. There is an otherwise identical compander available having a soft-knee characteristic. The variety of action available is very useful. Although setup is not entirely intuitive, it was particularly easy to adjust parameters while maintaining a constant maximum output level. The THRESHOLD control sets the point where compressor action begins. Below this threshold, the channel returns to a linear gain structure. WIDTH then works backwards at unity gain down to the signal level at which a fixed expansion of about 5:1 ratio starts. RATIO alters the compression ratio. This mode of operation looks useful for signals where some gating action is required, but the principal need is for compression.

In Fig.9, you can see the expander action; rather than providing hard-knee and soft-knee versions, a Knee control is available. The expanders differ from gates in that the expansion occurs from the threshold downwards to signal infinity, and the ratio of signal input to signal output is variable on the expander, and fixed to infinity on the gate. In both, above threshold, the signal maintains a linear gain structure. The expander also worked well.

The middle cluster of curves depicts the knee control action, which is fairly subtle in its effect.

Gate function is plotted in Fig.10. Again, operation is easy, with usefully constant maximum output level. The plot shows the function of the two main level adjustments —above the THRESHOLD and below the RANGE setting, the signal path is linear in gain.The compressor action is shown in Fig.11. This is quite conventional and operates as expected.

Lastly is the ducker. The measurement here is slightly different. The previous dynamic elements were tested self-keying—that is using the signal level of the controlled signal to alter

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itself. A ducker is rarely used this way, so this one is triggered by an amplitude varying tone on the adjacent channel while maintaining a constant amplitude sine wave on the channel under test. The ducker has two level related controls; THRESHOLD which sets the key level at which the signal begins to duck, and RANGE, which sets the level by which the signal is reduced when ducking occurs. Except for the ducking action itself, the signal input to output ratio remains 1:1. Fig.12 shows the action and effect of adjusting the two controls.

THE AUTOMIX SYSTEM on the 02R does not record real-time moves of any of the dynamics controls, though their initial channel settings can be recalled as part of a scene memory.

The last set of graphs examine the time characteristics of Dynamics section operations. A 1kHz sine wave burst signal is used having 80ms at full level, followed by 120ms at a level reduced by 30dB. For ease of graphing, only the signal level envelope is shown, individual sinewave cycles are not.

Fig.13 compares the unaltered signal (dotted plot) with compressor action (solid plot). Here, Attack is set to 120ms (slowest) and Release to 11ms. Note that although there is no hold adjustment available, the signal appears to hold for about 40ms before release begins. In Fig.14, Attack is altered to 0 (instantaneous, fastest), while Release remains as 11ms. Compression does appear almost instantaneous, and indeed could be with a digital architecture as feedforward limiting is possible. In this plot there is no hold period evident.

In Fig.15 the compressor release time is increased to 59ms. Now the signal does not return to an uncompressed state before the burst level increases again. Here, also, there is no evidence of a hold condition. Further experiments revealed that the hold time is dependent on the Attack time setting, increasing with increasing attack time.

Moving to the expander mode, there is quite a degree of apparent interaction between controls

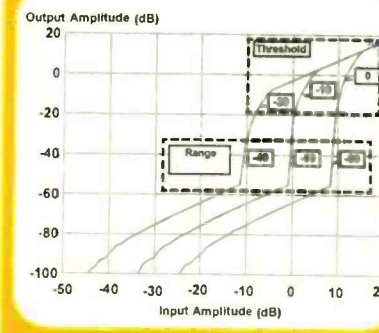

Fig.10: Dynamics test—gate action. Several possible setups are shown overlaid

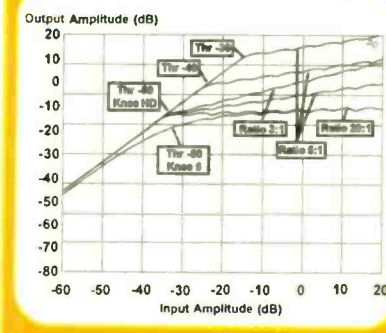

Fig.11: Dynamics test—compressor action. Several possible setups are shown overlaid

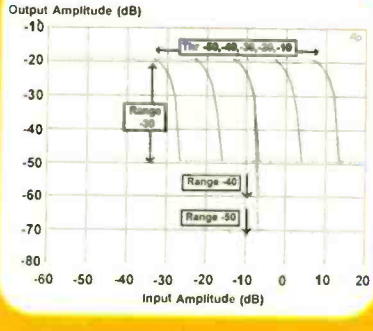

Fig.12: Dynamics test—ducker action. Several possible setups are shown overlaid

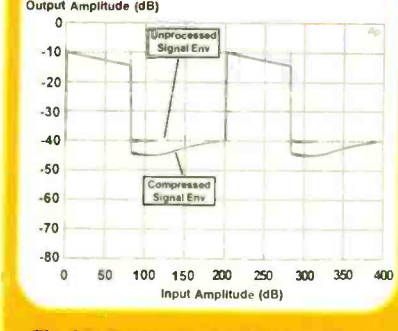

Fig.13: Dynamic signal envelope—compressor mode input signal is 1kHz tone burst, on at +10dBu for 80ms, on at +20dBu for 120ms, repeating. Dotted line is envelope of signal with no dynamics. Solid line is compressed with attack set to 120ms (slowest), release is 11ms

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when using the burst waveform described above. This is because expander release times are usually longer than the cycle that this test allows. Fig.16 shows the effect of varying the Attack control while leaving Release constant.

The dynamics worked well and were flexible in their range of settings. Since these are particularly awkward to reset, the 02R's library function will prove useful in speeding up their use.

Effects in the 02R consist of two independent mono input—stereo output digital-effects devices. Routeing into these is via Aux 7 and Aux 8, with returns on dedicated Eff 1 and Eff 2 return channels. Though not indicated clearly in the manual, these are stereo returns and the associated equalisers also appear to be stereo. The returns can be routed to the stereo and any of the eight mix buses. All routeing and processing is in the digital domain, so there are no quality losses due to signal conversion.

There are quite a variety of effects available in the unit, and though it is not as versatile as some of the independent external effects units available, it makes up for this in ease of use and improved signal quality. You should not, however, attempt to use the controls on the effects devices in real time as there is a high likelihood of incurring audible glitches.

There is a lot more to look at, and a lot more potentially to say. Those comments should now be reserved for those who have purchased a unit and who will be using it to earn their living over a period of months and years. Each aspect of my examination has revealed another feature or two, mostly useful, and the occasional limitation.

There is little to criticise in the 02R. In system with a set of today's generation digital multitracks, the performance is as good as a good analogue desk, and can be much better due to a reduction of conversions during the recording process. And where can you get a set of operational features like this for anything like this cost. S

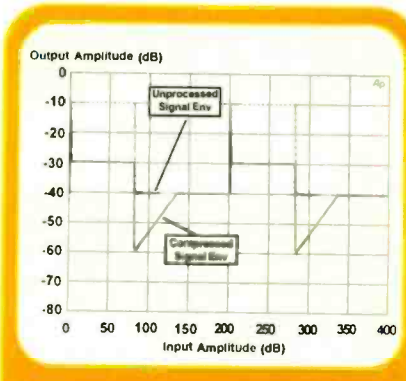

Fig.14: Dynamic signal envelope —compressor mode input and dotted line as fig.13. Solid line is compressor with attack set to 0ms (instantaneous), release remains at 11ms.

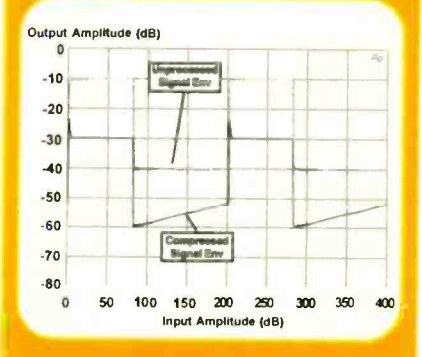

Fig.15: Dynamic signal envelope —compressor mode input and dotted line as fig.13. Solid line is compressor with attack set to 1ms (very fast), release increased to 59ms

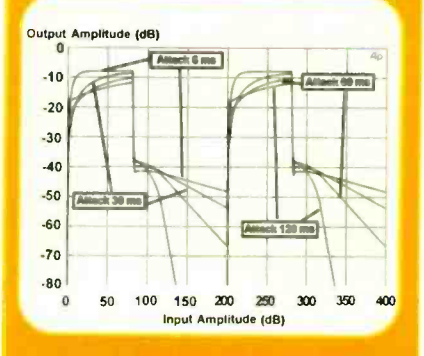

Fig.16: Dynamic signal envelope —expander mode input and dotted line as fig.13. Solid line is expander with attack set to 6ms, 30ms, 60ms, 120ms. Release is constant at 48ms

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National Vintage Communications Fair, NEC, Birmingham, UK. Tel: +44 1398 331532.

◆ **May 11th–14th**
100th AES Convention, Bella Centre, Copenhagen, Denmark. Tel: +45 9785 1122.

◆ **May 12th–14th**
NSCA, St Louis, US.

◆ **May 14th–16th**
Midem Asia, Hong Kong. Tel: +33 1 44 34 454 44.

◆ **May 15th**
AIR DAB Seminar
Cumberland Hotel, London, UK. Tel: +44 171 727 2646.

◆ **May 17th–20th**
BTV China, China Foreign Trade Centre, Guangzhou, China. Tel: +852 2862 3460.

◆ **May 23rd–26th**
CES Speciality Audio & Home Theatre Show
The Hilton, Walt Disney World Village, Orlando, Florida, US. Tel: +1 703 907 7600.

◆ **May 25th–28th**
Pro Audio, Light & Music China 96, Beijing Exhibition Centre, Beijing, China. Tel: +852 2861 3331.

◆ **May 28th–30th**
7th Conference and Exhibition on Television and Audio Technologies
Thermal Hotel Helia, Budapest, Hungary. Tel: +36 1 153 0127.

◆ **May 29th–June 1st**
Expo Show 96
St Petersburg, Russia. Tel: +7 812 271 4147.

◆ **May 29th–June 3rd**
International Electronic Cinema Festival (IECF)
Makuhari Messe Conference Hall, Chiba City, Japan. Tel: +81 3 3408 4111.
Viacom, US. Tel: +1 212 258 6324.
IECF, Europe. Tel: +41 21 963 3220.

June 1996

◆ **June 1st–4th**
Nightwave 96
Exhibition Centre, Rimini, Italy. Tel: +39 541 711249.

◆ **June 4th–6th**
REPLITech
San Jose, US. Tel: +1 914 328 9157.

◆ **June 4th–7th**
Broadcast Asia, World Trade Centre, Singapore. Tel: +65 338 4747.

◆ **June 5th–8th**
Third Annual Latin-American Pro Audio & Music Expo, World Trade Centre, Mexico. US. Tel: +1 914 993 0489.

◆ **June 6th–9th**
Radio Montreux, Montreux, Switzerland. Tel: +41 21 963 52 08.

◆ **June 6th–9th**
DigiMedia, Montreux, Switzerland. Tel: +41 21 963 52 08.

◆ **June 6th–9th**
Montreux International Radio Symposium and Technical Exhibition including 1st Interactive Media Symposium and Exhibition, Montreux, Switzerland. Tel: +41 21 963 52 08.

◆ **June 10th–15th**
Americas TELECOM 96
Rio de Janiero, Brazil. Tel: +41 22 730 6161.

◆ **June13th–15th**
Infocomm International
Philadelphia, US. Tel: +1 703 273 7200.

◆ **June 19th–21st**
Audio 96: Technology & New Media (APRS)
Olympia, London, UK. Tel: +44 1734 756218.

◆ **June 20th–22nd**
World Lighting Fair 96
Pacifico Yokohama Exhibition Hall, Yokohama, Japan. Tel: +81 3 3706 5687.

◆ **June 25th–30th**
ITS 9th Annual Forum and Exposition, Chicago, US. Tel: +1 212 629 3266.

July 1996

◆ **July 10th–12th**
Pro Audio & Light Asia 96
World Trade Centre, Singapore. Tel: +65 227 0688.

◆ **July 12th–14th**
Summer NAMM, Nashville, US. Tel: +1 619 438 8001.

August 1996

◆ **August 7th–10th**
MacWorld Expo
Boston, US.

◆ **August 15th–18th**
Popkomm, KölnMesse, Cologne, Germany. Tel: +49 221 8210.

◆ **August 26th–29th**
Windows Solutions Expo & Conference
San Francisco, US.

September 1996

◆ **September 3rd–6th**
Broadcasting China 96 Exhibition & Symposium
Beijing, PR China. Tel: +86 10 609 3775.

◆ **September 4th–8th**
CEDIA, Dallas, Texas, US.

◆ **September 6th**
International Monitor Awards, Beverley Hilton Hotel, Beverley Hills, California, US. Tel: +1 212 629 3266.

◆ **September 7th–10th**
British Music Fair, Earls Court, London, UK. Tel: +44 1442 215435.

◆ **September 8th–11th**
Plasa, Earls Court, London, UK. Tel: +44 171 370 8179.

◆ **September 10th–12th**
6th Australian Regional AES Convention, World Congress Centre, Melbourne, Australia. Tel: +61 3 9682 0244.

◆ **September 12th–16th**
IBC 96, RAI, Amsterdam, The Netherlands. Tel: +44 171 240 3839.

◆ **September 16th–20th**
Seventh International Symposium on Electronic Art (ISEA 96)
Rotterdam, The Netherlands. Tel: +31 10 477 8605.

◆ **September 17th–19th**
Interactive Multimedia Association Expo
New York, US.

◆ **September 18th–23rd**
photokina, KölnMesse, Cologne, Germany. Tel: +49 221 8210.

◆ **September 21st–23rd**
cinec, MOC Events Centre, Munich, Germany. Tel: +49 89 51070.

◆ **September 24th–29th**
Live 96, Earls Court, London, UK. Tel:+44 181 233 9306.

October 1996

◆ **October 24th–26th**
Broadcast India 96 Exhibition & Symposium
World Trade Centre, Bombay, India. Tel:+91 22 215 1396.

November 1996

◆ **November 5th–7th**
Broadcast Engineering Society Expo 96, Hotel Taj Palace, New Delhi, India. Tel: 91 11 371 9978.

◆ **November 5th–9th**
PT/Expo Comm China
China International Exhibition Centre, Beijing, Peoples Republic of China. Tel: +52 525 592 3257; US Tel: +1 301 986 7800.

◆ **November 6th–9th**
AV & Broadcast China 96
Beijing Exhibition Centre, Beijing, China. Tel: +852 2862 3460.

◆ **November 7th**
21st Sound Broadcasting Equipment Show (SBES)
The Metropole Hotel, NEC Birmingham, UK. Tel: +44 1491 838575.

◆ **November 8th–11th**
101st AES Convention
LA Convention Centre, Los Angeles, California, US. Tel: +1 213 258 6741.

◆ **November 15th–18th**
Tonmeistertagung
Stadhalle, Karlsruhe, Germany. Tel: +49 2204 23595.

◆ **November 21st–24th**
LDI 1996, Orlando, US. Tel: +1 212 229 2965.

◆ **November 21st–24th**
News World 96, Inter-Continental Hotel, Berlin, Germany. Tel: +171 491 0880.

December 1996

◆ **December 3rd–5th**
Online Information 96, Olympia 2, London, UK. Tel: +44 1865 730275.

◆ **December 8th–11th**
Broadcast Cable & Satellite 96 including Pro-Audio India, Lighting India and TV India, Pragati Maidan, New Delhi, India. Tel: +91 11 462 2710.

◆ **December 8th–11th**
Communications India 96 including Networks India and Wireless India, Pragati Maidan, New Delhi, India. Tel: +91 11 462 2710.

February 1997

◆ **February 22nd–25th**
Middle East Broadcast 97
Bahrain International Exhibition Centre. Tel: +973 550033; Tel: +44 171 486 1951.

April 1997

◆ **April**
Entech 97, Melbourne Exhibition Centre, Australia. Tel: +61 2 876 3530.

August 1997

◆ **August 25th–28th**
SMPTE, Sydney, Australia.

October 1997

◆ **October 16th–20th**
International Audio, Video, Broadcasting and Telecommunications Show (IBTS 97), Milan, Italy.

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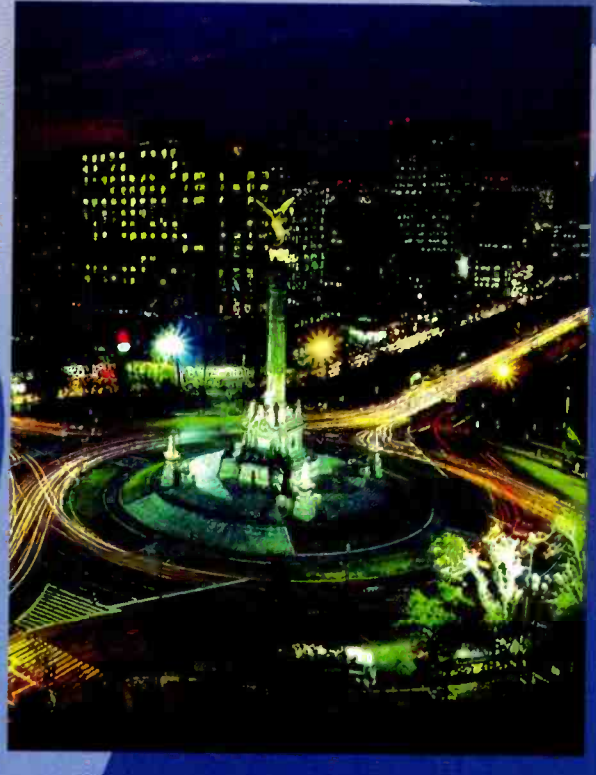

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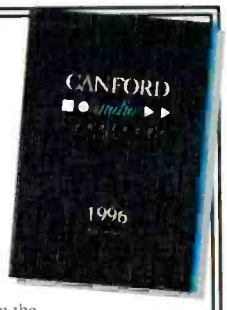

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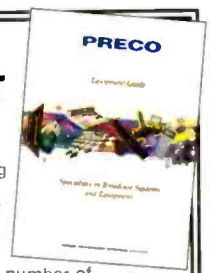

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Meek and contentious

Joe Meek remains a contentious figure almost 30 years after his death. In the wake of *Studio Sound*'s recent reappraisal of the great producer **ADRIAN KERRIDGE** files his personal memories

IMAGINE THIS: It is nine years after the end of WWII. Britain is still celebrating the release from the shackle of war and is experiencing new-found business optimism. The Popular charts are dominated by US artists, by today's standards labelled 'middle of the road' but here they were making (some of them) star money. In the UK artists were making the hit parade some whose careers had started in the Big Bands, some remembered today, some not.

So it was a young Joe Meek who came to London. He passed a stringent interview with Allen Stagg, General Manager of IBC studios, and joined the company in mid-1953. Joe's job was primarily to record radio shows for Radio Luxembourg which entailed travelling the country far and wide and it was during those long steam-train journeys that Joe wrote his songs. I was privileged to be assigned to work with Joe on the vast majority of those shows and he frequently asked my opinion as we travelled.

Of the many positive things at IBC one was in the choice of people, and Allen Stagg intuitively picked people with talent. The attitude at the time was that some people have a natural ability, creatively and empathy to work with and record artists, and the company pointed you in the right direction with encouragement. Unfortunately, many highly talented individuals under one roof doesn't make for good teamwork and Joe was no exception.

Equipment was primitive—consoles were valve-type with simple treble and bass equalisation, made in house; microphones were dynamic made by STC and Coles, also large BBC AXBT ribbons and a couple of condenser mics of dubious origin but which sound fantastic on strings and as ambient mics. Processing gear was limited to a couple of limiter-compressors and an extremely good multifrequency valve equaliser (both charged as extras). Tape machines were ¼-inch full-track mono EMI BTR 1's and BTR 2's. All recordings were made at 30ips —firstly on EMI tape until there were problems and the company switched to Agfa FR4—using the CCIR curve. Although the company had a ¼-inch test tape the recording levels were 'suck it and see'. Flux density? What was that? Nobody worried too much about levels. This then was the climate in which we worked.

When Joe and I were not on the road we did sessions in house and I became his confidante, friend and kindred spirit. I was working all the time as his assistant, and at a later stage at Lansdowne, taking some of his workload. At IBC he was continually experimenting with mic techniques especially close microphone techniques. When he asked me to go to the workshop and have them build him a mike splitter feed to parallel two mikes into the same input the tech guys said: 'Okay but you'll overload the mic input'. Joe's attitude was: 'We'll try it'. Next morning the other engineers were curious to hear the finished tape. No-one had heard a trumpet sound like it—fairly clean but so up-front, Joe had 'cooked' (using the semi-multiband EQ) the tape and used the limiter over the final master. Until Joe came along all the British pop 'band' sounds were 'roomy' and dull. Consequently, Joe became very much in demand, to the point of being overworked on many occasions. The management always required him to be in on time next morning, whatever time he finished the night before.

THE PRESSURE at IBC eventually became too much for Joe. In 1958, Dennis Preston invited him to find suitable premises and build a studio that Joe could manage, and have all the equipment built that he wanted. Consequently, Lansdowne House, Holland Park was bought by Preston and the name changed to Lansdowne Recording Studios. Work commenced immediately with acoustic design by Sandy Brown, an employee of the BBC at that time. EMI Hayes designed and built the console to Joe's specifications at a cost of £4,500—the finish was purple with gold-edged trim. It was all-valve with 12 inputs and two outputs. There were only four valves in each channel making for a short signal path with a wide-band frequency response and extremely good transient response. Tape machines were TR51s, also supplied by EMI Hayes. I remember Dennis saying he could have bought a top-of-the-range Mercedes for the money! This time we bought Agfa calibration test tapes, so there could be no discussion about arbitrary levels.

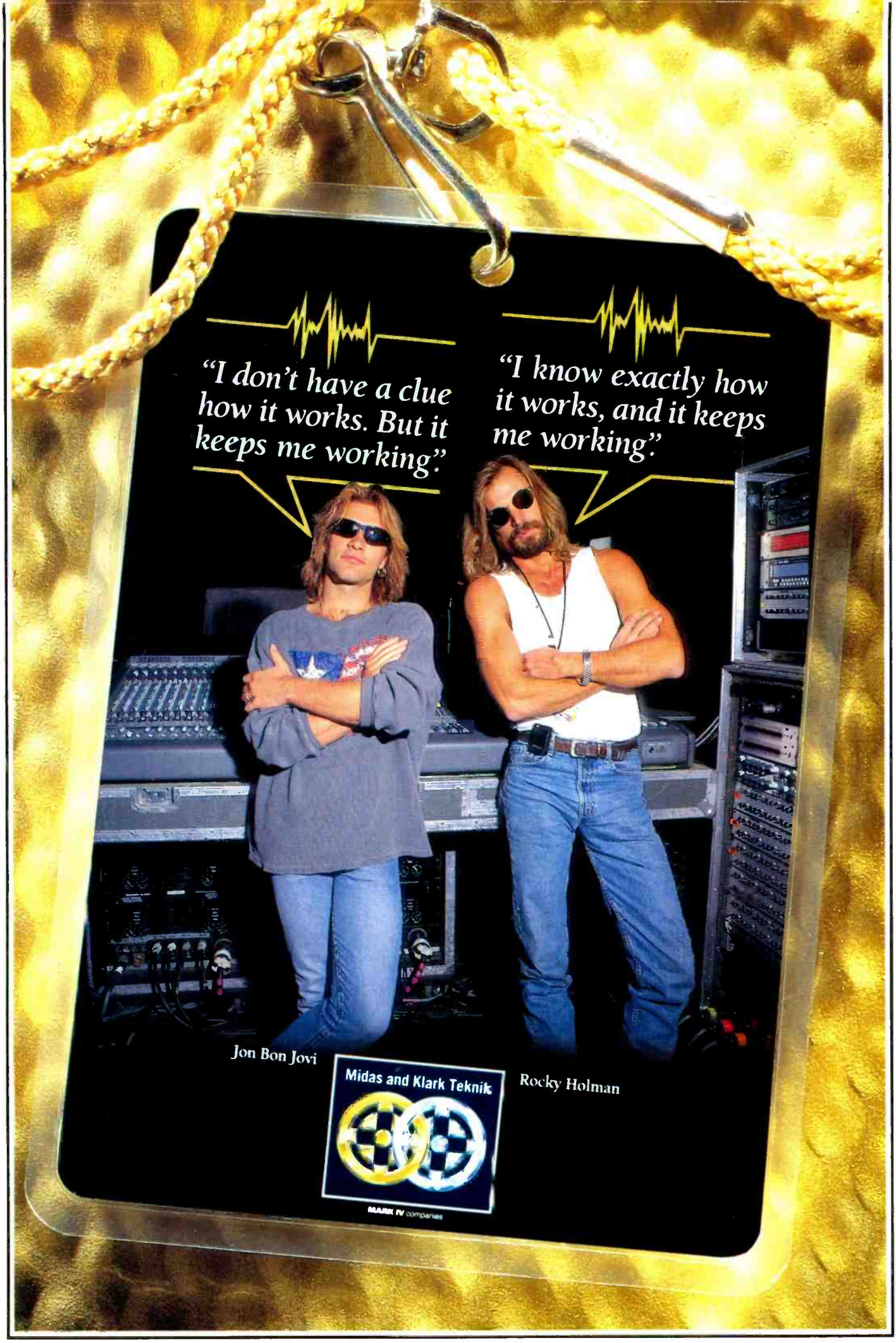
The studios opened in February 1959 and were an immediate success. The pulling power of Joe was immense. We worked to a standard of flux density of 32 milli Maxwells/metre as a guide, for Joe pushed the tape to its limits. The general guide being how much level could we get on without too much obnoxious distortion. We were still being criticised by the major record companies' white-coated technical men, especially the cutting engineers of those companies who by know we dubbed as 'frustrated balance engineers'—some of them were failed engineers. And he worked many times a week throughout the night and that took its toll on his tolerance level. He also began to take slimming tablets called *Preludin*, which kept him high and enabled him to work those very long hours. This habit produced pronounced mood swings to the points of paranoia about people stealing his ideas.

One day in early 1961, on a Dennis Preston jazz session, there was a tremendous argument in the control room, during the recording over some songs that Joe wanted Dennis to record. Dennis quite rightly told Joe, 'the time and place to discuss this is after the session'. Joe would have none of it and walked out. I took over the recording and finished the tracks. The next day Joe was fired and sadly I had very little contact with him save for the odd telephone call.

So we continued at Lansdowne, and Joe at home in the Holloway Road. I leave the Holloway Road era to those who better knew him there.

To put the record straight further, I coproduced the Dave Clark Five hits. The stomping on 'Bits and Pieces' was an entirely original idea created by us in the studio on the day. The term bouncing tracks was not in use, we made composite tracks by dubbing from one machine to another, adding live instruments as we dubbed. The original limiter-compressor Joe use had a pentode valve as the control element. It was made by Langevin, modified by Joe, and later by me. I still have that old limiter-compressor and it suited me well on all the Dave Clarke Five hits.

In my opinion, Joe Meek was the founding father of the basic approach to pop recording we use today—close miking techniques and loads of processing. A true visionary.



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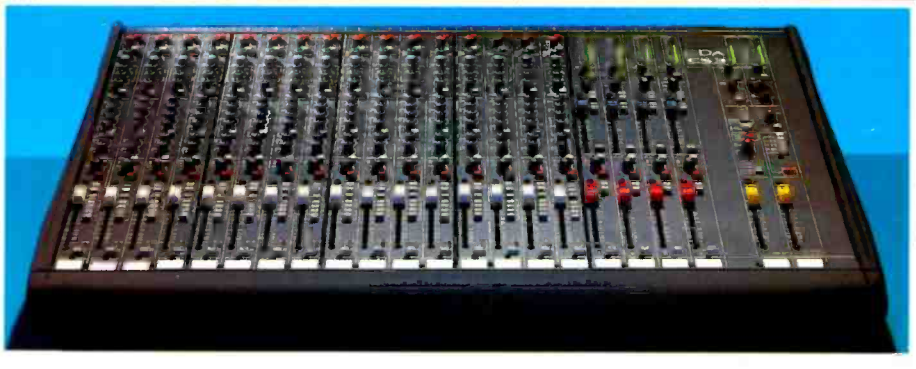

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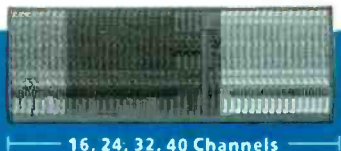

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