August 1998 $7.20 £3.75

Studio sound

THE INTERNATIONAL PROFESSIONAL AUDIO MAGAZINE
FOR RECORDING, POSTPRODUCTION AND BROADCAST

EXCLUSIVES

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Eventide DSP 4500
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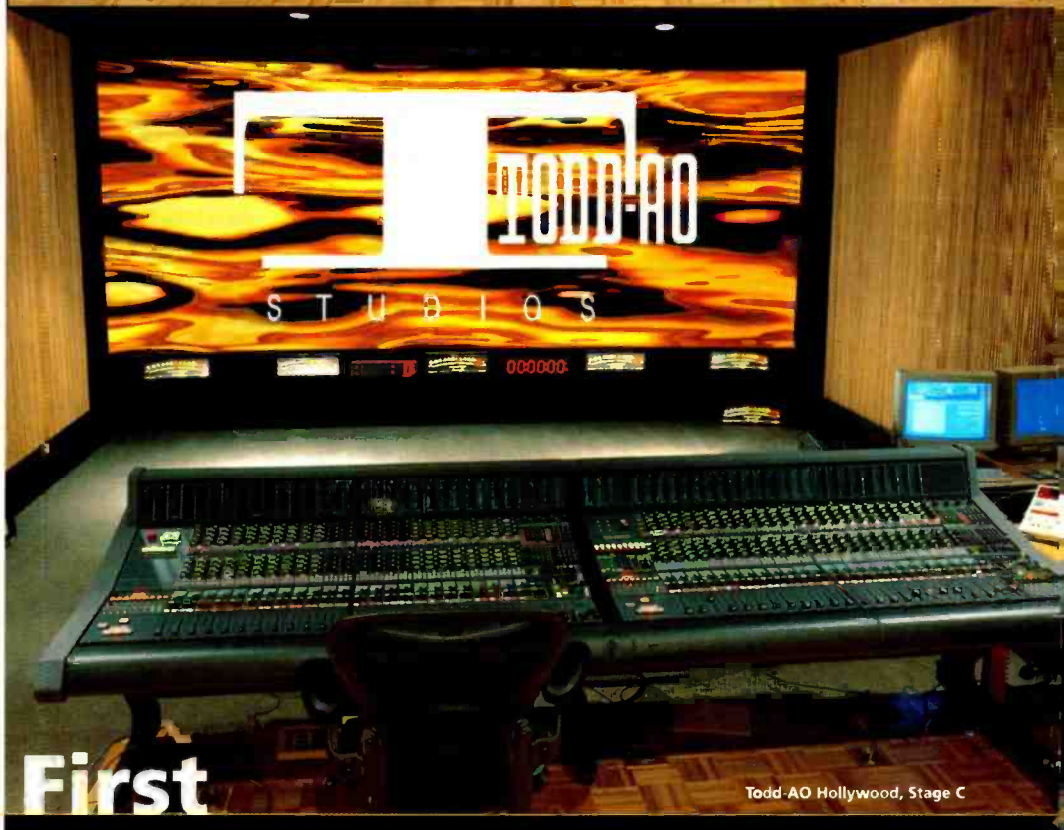

Todd-AO Hollywood, Stage C

First

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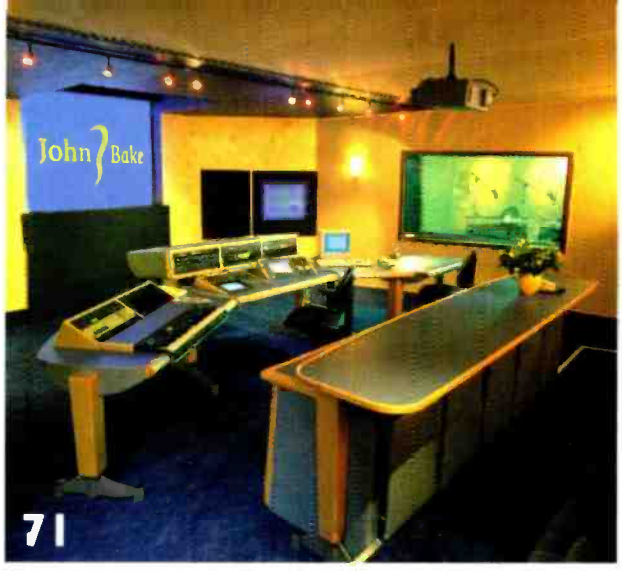

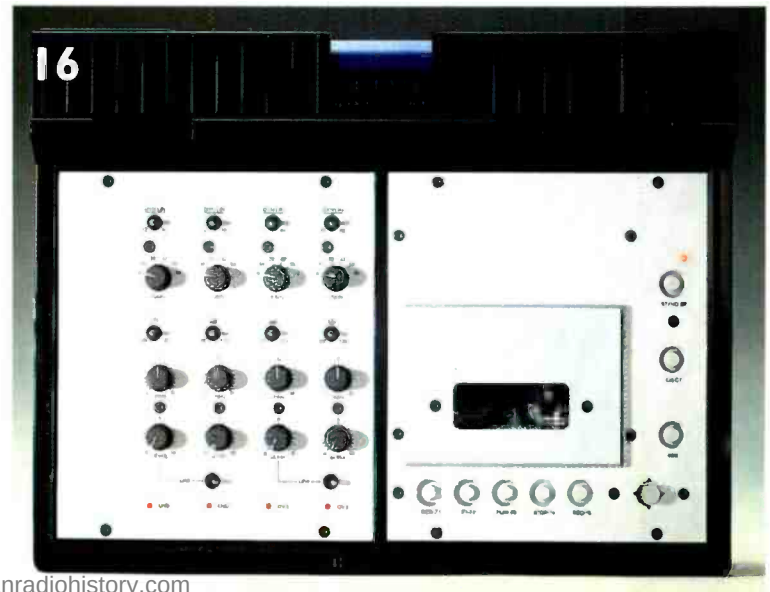

Grey scale

THERE WAS SOMETHING IN THE WATER in Las Vegas and Amsterdam when a fester of DAW-oriented manufacturers used the trade shows in question to announce network-oriented news. This varied from yet more examples of manufacturers saying they intend to network, stipulations of now being even closer to the intended goal than they were a year ago when they first said they intended to network, and assurances that they were very nearly there now. There are also networks that exist and work, but what has been proven time and again is that networking DAW's is breaking hearts. And it's hard.

The situation is complicated by a number of factors—the desire in manufacturers to go the right route, the desire of manufacturers to do the right thing, and the expectation of the end-users. The manufacturers' viewpoint is fairly easy to comprehend. Tough to do, and even tougher to do properly with an eye for what is best for the user, the next generation of sales, the continued loyalty to existing product and the well-being of the manufacturer could all be said to depend on the precise nature of the networking they develop.

For the user the concept is less precise. The word 'networking' is like the colour grey and illicits largely similar responses if you ask someone to describe it. All interpretations are roughly relevant, but the varied nature of the responses is what endures.

Most audio people's experience of networks has probably been confined to office environments, environments in which the system is set up and left for you, you are told what you can and cannot do, and you work within the rules.

The majority of users of stand-alone DAW users in post will not be acquainted with system wide 'rules', and I would suggest that the majority of facilities, even if they've put their names down for the net option, will not have given any detailed thought to how they would organise themselves, the work-flow and staff over a network, and what operational impact and limitation it would impose upon their trade.

If they had then they might have given thought to the type of network they might want, and this might not be the type that the manufacturer who made their DAW's is planning for them. Ask some more questions. **Zenon Schoepe, executive editor**

Paper heroes

SOMEBODY ONCE QUOTED the apparent inability of some women to program video recorders as the contemporary arbiter of the difference between the sexes. Yet somehow the prospect of a man sitting down with a beer to watch a recorded sports broadcast from the other hemisphere only to find he's caught a 'nite-time' talk show on the unsettling subject of Impotence, and an investigation into the causes of crop circles, must also happen.

Certainly, some men use the obfuscation of video recorders to intimidate women, but conceal that fact that they regularly fail the same test of technical prowess themselves. So why haven't we mastered the video timer? What can be so difficult about setting a TV channel and clock that it eludes us so consistently? The fact is that society is a technologically complex entity and as such it is difficult to keep the public at large consistently educated in the workings of its amenities.

The point was brought home to me recently when I was on my way to take a look behind the scenes of the recording and broadcast setup backstage at the UK's showcase opera operation, Glyndebourne—a setup that had also piqued the curiosity of one of the British national newspapers, the *Mail on Sunday*. The *MoS* interest was in the collision of high art and technology, or tradition and progress, if you prefer. The technical journalist in whose company I found myself claimed a background in the technical press, but had no understanding of matrixed surround sound, let alone digital technology. How many bits are on a CD? Pro Logic, is that a manufacturer? And so on.

Let's just make clear that this lack of information was openly declared—there was no (typically masculine) temptation to bull through the situation. Rather, it seemed odd that a national newspaper should tender someone so unfamiliar not only with the professional audio and video technology, but also with consumer technology already found in many homes.

So I am left to ponder: how does the popular press intend to inform its better informed readers? And how can we hope to see the likes of DVD, digital radio and digital TV accepted by a general public served by such a press?

Oh, and yes, the *MoS* journalist was a woman. **Tim Goodyer, editor**

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Incorporating Broadcast Engineering

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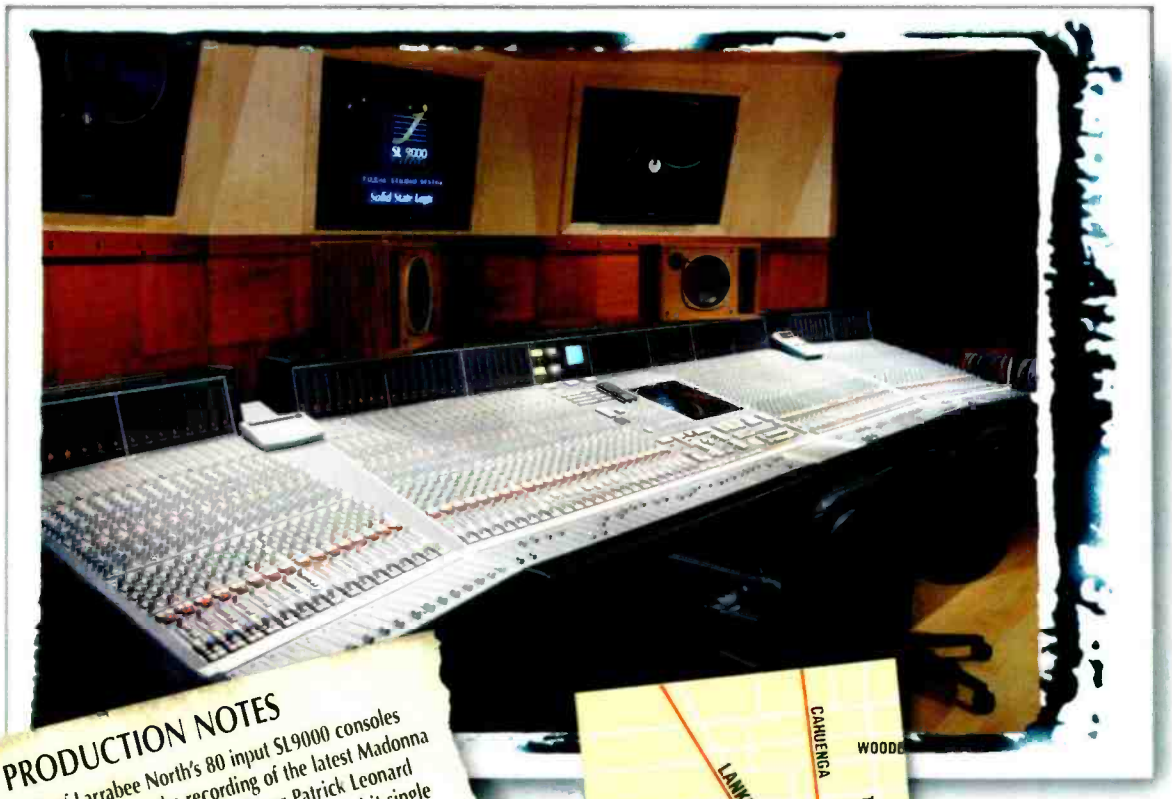
Great Studios Of The World



PRODUCTION NOTES

Both of Larrabee North's 80 input SL9000 consoles were used during the recording of the latest Madonna album 'Ray of Light'. Co-producer Patrick Leonard also co-wrote several tracks, including the hit single 'Frozen'. "This was a very complex album because it included a large amount of virtual information" says Leonard. "In working with the SL9000, it was wonderful to have all of the virtual setups recalled perfectly every time. It all worked so well together, and the sound is so big! I really enjoy working on this console."

Larrabee SOUND STUDIO

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▲ UK: While Sam Neill's image graced the preview screens in Abbey Road's Studio 1 recently, the LSO worked on Trevor Jones Score to Mysterious Passions' production of *Merlin*. Directed by Steve Barron and also starring Helena Bonham Carter, Miranda Richardson and Rutger Hauer, the recording used the facility's 64-channel Neve VRP and 48-track digital-Tascam DA-88 recorders in the care of engineer Simon Rhodes. Abbey Road Studios, UK. Tel: +44 171 266 7000.

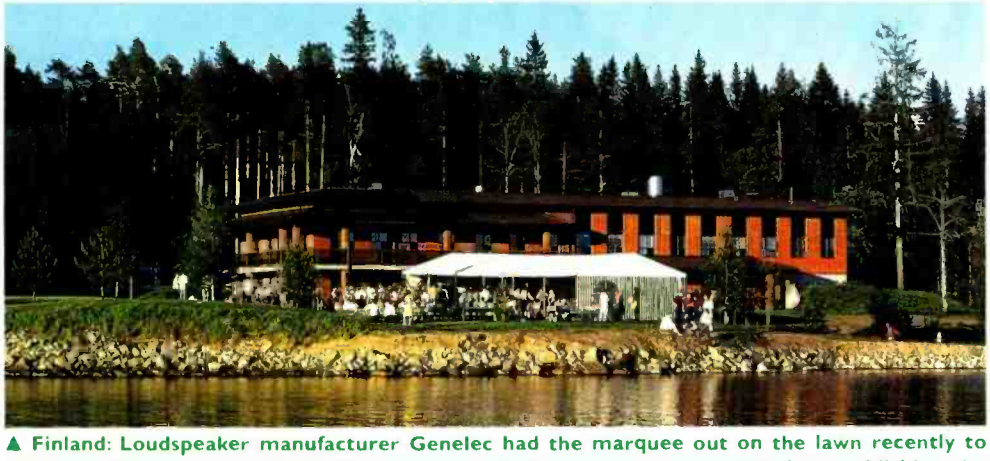
◄ US: The recent refurbishment of Superdupe's No.1 Suite has seen it upgraded to a 5.1-capable digital room. Designed by the Walters Storyk Group, the new look for the New York-based facility is centred on a 160-channel, 48-fader Soundtracs DPC-II and an active Spendor surround system. Completion of the refurbishment has signalled orders for two further DPC-IIs for Superdupe's sister Lower East Side. Soundtracs & Spendor Audio Systems, UK. Tel: +44 181 388 5000.

Japan: Tokyo's Nippon Sogo Seisaku has dubbed its 1-inch, 2-track analogue recorder 'the Monster Machine'. Its arrival coinciding neatly with the resurgence of interest in Japan's *Godzilla*. NSS' custom Studer was built by American JRF Magnetics and will be available for hire alongside NSS' other speciality—high-performance digital equipment. Already blooded in the recording of a forthcoming release from Japanese drummer Syuichi Murakami on which the multitrack machine was a 3348, the Monster Machine prompted NSS president Masamichi Ohashi to comment, 'It illustrates analogue's continuing ability to compete in an increasingly digitised industry. We started doing mixes with it at JVC Studios in May and the engineers were astonished. They thought such quality simply couldn't exist even though they have used the newest 192kHz, 24-bit systems'. He also commented, 'I've felt that the ½-inch standard didn't fully demonstrate the potential of analogue but what we have heard from the Monster machine is truly unbelievable'. On the test bench, the Machine delivered 25Hz-25kHz frequency response, 'ruler flat' between 250Hz and 12.5kHz with a 78dB S–N ratio. **www.jrfmagnetics.com**

▲ US: San Francisco's Music Annex music recording and post facility has been celebrating its 25th anniversary, as can be seen from the gathering around the Neve 8108 console in Studio C. Founded by David Porter in a San Jose garage, the facility is still under Porter's presidentship—any drink in the Flying Faders system is, therefore, his responsibility. The Music Annex. Tel: +1 650 328 8338.

UK: Hosted by Emtec Magnetics and London mastering-duplication facility Chop Em Out, a recent Archiving and Remastering forum saw a panel of speakers address a selection of studio and record company personnel on topics including technical considerations and project case studies. Opened by Simon Heyworth who was involved in both the original recording and remastering of Mike Oldfield's *Tubular Bells*, the session continued with a personal account of tape ageing problems from BMG's Andy Griffin who is currently cataloguing and rearchiving the company's 70,000 tapes. Charged with building and archiving the Elvis Presley catalogue, Castle's Roger Semon offered a mixture of education and anecdote on the recordings themselves, and what has been done to them in the name of restoration. While Emtec's Matthias Bechly offered an insight into the tape manufacturers' challenge. Mixing recording spanning several decades with footage of Elton John to match, Henry Scott-Irving offered a unique insight into the problems and opportunities offered by a useful archive before Sony's David Walstra made a bid for the future with an account of the Sony-Philips Direct Stream Digital system. The technical depth of the session was limited by the inclusion of record companies, but offered insights into the considerations of archiving, and the attitude of some of those involved.

Tim Goodyer



▲ Finland: Loudspeaker manufacturer Genelec had the marquee out on the lawn recently to celebrate 20 years of business. The company, which has been responsible for establishing the concept of active monitoring in the professional environment, has also completed a substantial expansion of its manufacturing facility in the north of the country. Genelec. Tel: +358 17 813311

UK: Reopening after a major refit, Scotland's Castlesound studio now boasts a newly refurbished Neve 51-series console complete with Optifile automation, Studer 24-track analogue and 32 tracks of Tascam digital recording, and an 8-track Spectral hard-disk system. Founded by producer Calum Malcolm in 1978 and based in Scotland, Castlesound is characterised by an impressive and varied client list that includes acts such as Simple Minds, Aztec Camera, Wet Wet Wet and the Royal Scots Dragoon Guards. Castlesound, UK. Tel: +44 131 447 0991.

▲ Singapore: The recent Pro Audio and Light Asia exhibition marked the 10th anniversary of an event which has grown consistently in size and in the awareness of the Western faction of the pro-audio industry. Setting out as an exclusively Eastern event, PALA increasingly tempted European and American exhibitors who came either to seek distributors for South-East Asia or to support their established distributors, or a little of both. This year, however, the dramatic downturn in the regions economy was evident in terms of exhibitors (down by some 25% by number to 17% by floor space) and visitors (down 25% to 6,108 with some 34% coming from 22 overseas countries). Notable absences extended from the support previously lent by the likes of Focusrite to its distributor Team 108 and previous years' independent showings from such as Spirit. ARX and the Australian Monitor Company, meanwhile, continued to wave the flag for Oz, while Fairlight continued not to. The show continued regardless and still provided opportunities for business and product launches. The finalising of HHB's arrangement with Sennheiser Electronic Asia, for example, saw the British-based manufacturer-distributor secure its presence in no less than 14 Asia-Pacific territories; the Victor Company of Japan (JVC) thought PALA a suitable opportunity to champion its XRCD (extended resolution CD) system against Sony's SBM without any opposition; East-West Communications unveiled its go- events.com website; while BSS and Wembley Loudspeaker premiered their FCS-966 constant Q graphic and close-field monitor respectively. It may have been down-sized, but there was enough going on.

The 1999 PALA exhibition will take place between 8th and 10th of July, and will see its venue move from the World Trade Centre to the Singapore International Convention and Exhibition Centre. Significantly, it will mark the last annual PALA for the time being, as the exhibition is to become biennial in the face of the economic situation. Whether this is a wise strategy remains to be seen. Tim Goodyer

Worldwide: The recent PALA show presented the platform for Turbosound's re-establishment as an independent company away from prior parent Harman Audio. In the US, the announcement has been made of the merger of E-mu Systems with Ensoniq Corp to form Emu-Ensoniq who will 'continue to offer audio and music products with different features and prices under separate brand names'. At the same time, in the northern reaches of the UK hands were being shaken over the merger of the digital division of Scottish music retailer Sound Control with KGM Studio Specialists as KGM Sound Control. The merger brings together the most prominent equipment distributors outside of London. KGM Sound Control, UK. Tel: +44 1924 371766.

■ London recording studio activity has seen London's newest studio, Stanley House, open its doors with an SSL SL9000 j-series console, restaurant, bar and roof garden. The studio has been set up by Andy Morris and occupies a Victorian warehouse near Shepherd's Bush. Olympic Studios, meanwhile, is preparing to install an SL4080 G+ SE console in its newly refurbished Studio 3. The announcement accompanies a move towards providing a new studio, Studio 4, to house the displaced SL4064. Abbey Road has adopted two Genex GX8000 MO recorders as a floating digital recording resource. The studio sees the machines as a practical alternative to DASH and Hi-8 machines, capable of combining portability and high-resolution recording. CTS Studio joins the list of TL Audio's C-1 stereo valve compressor.

Stanley House, UK.
Tel: +44 181 735 0280.
Olympic, UK.
Tel: +44 181 748 7961.
Abbey Road, UK.
Tel: +44 171 266 7000.
SSL, UK. Tel: 1865 842300.
HHB, UK. Tel: +44 181 962 5000.
TL Audio, UK.
Tel: +44 1462 490600.

■ Istanbul's Sinefect has installed a 16-fader AMS Neve Logic 3 and 24-output AudioFile Spectra in its Studio 1. Already up and running, the newly-equipped studio has seen use in posting commercials for BMW and Pirelli.

AMS Neve, UK.
Tel: +441282 457011.

■ New York's Bearsville studios has installed an SSL 4064 G+ console with Total Recall to coincide with the facility's 10th birthday. The new installation has already been field tested by Bob Clearmountain and Stephen Hague on a Pretenders session. Out on the streets of New York, Randy Ezratty's Capricorn-equipped L7 mobile has seen the installation of Spendor SA500 3-way monitors to assist it on its tour of major live recording and broadcast projects. NYC's Sony Music Studios has invested in a pair of Martech MSS-10 mic preamps.

Bearsville Studios, US.
Tel: +1 914 679 8900.
Effanel, US. Tel: +1 212 807 1100.
Sascom Marketing, US.
Tel: +1 905 469 8080.
MartinSound, US.
Tel: +1 203 466 1104.

■ Kuala Lumpur's KLCC building project has involved Abbey Road as design consultants in the building of the control room of a prestigious new concert hall, Dewan Filharmonik Petronas. The hall is part of the Petronas Twin Towers where it forms a low-level bridge between the two halves of the tallest building in the world. The project, which is optimised for classical use, was overseen by Abbey Road's Neil Aldridge, is built around an SA&V SADiE 24-96 and has considerable ability to link with OB vehicles. The hall will be home to the newly formed Malaysian Philharmonic Orchestra.

■ Los Angeles' Novastar Digital Sound Services has installed a 128-input Otari Advanta digital console as the centre of its 2-year renovation of Mix Stage A. The 2-man desk sits in a THX-certified, surround capable room designed by George Augspurger where it will support Novastar's rosta of TV and film post-production services.

Novastar, US. Tel: 213 467 5021.
Otari, US. Tel: +1 650 341 5900.

■ Italian state broadcaster, RAI, has taken delivery of 15 Orban Audicy workstations to make a standard in its facilities around the country including those in Milan, Rome and Turin. The choice of the Audicy follows a period of consideration and consultation involving all those concerned in their operation.

RAI, Italy. Tel: + 39 11 810 4442.
Orban, US. Tel: +1 510 351 3500.

■ Chinese Ninjang Television has installed a Euphonix CS2000 console. The move brings the facility in line with a number of key facilities in Beijing and has already been used on a television production entitled Linlin Mumu Sensen as well as a musical variety show, Meidi Zhi Chun.

Euphonix, S. Tel: +1 818 766 1666.

■ Hire: London-based Orbital has purchased some 61 d&b E3 loudspeakers, 16 902s and 12 602s together with matching d&b P1200 amplifiers. The company has also taken 25 Lab Gruppen 1200C 4-channel amps. Elsewhere in the UK, CVA has purchased 48 Nexo PS-15 compact loudspeakers.

Orbital, UK. Tel: +44 171 978 7828.
CVA, UK. Tel: +44 1932 341400.

■ LA-based Ruthless Records has installed a pair of Westlake Audio BB-10SWP loudspeakers in its demo control room. The scale of the monitoring system reflects Ruthless' specialisation in rap and R&B artists. Memphis' Woodshed studio has commissioned its 40-channel Otari Status console while Bobby Croft's The Grey Area studio in Orlando has purchased a Martech MSS-10 mic preamp for use with its Augan 24-track digitalworkstation.

Ruthless, US. Tel: +1 213 782 1888.
Westlake, US. Tel; +1 805 499 3836.

■ BBC Belfast has ordered a 6-channel Calrec X-series on-air desk for installation in its Radio Studio 2. The digital console will be used for broadcasting hourly news bulletins on Radio Ulster. Elsewhere in Belfast, Production House has purchased 6 XTA GQ600 dual-31-band graphic EQs, and an XTA DP266 speaker management system to complement its new L'Acoustic ARCS speaker system.

BBC Belfast, UK. Tel: +44 1232 338000.
Calrec, UK. Tel: +44 1422 842159.
Production House, UK.
Tel: +44 1232 673786.
XTA Electronics, UK.
Tel: +44 1299 879977.

■ New York post house Jonathan Helfand Music & Post is a new 3-room facility set up by established NY engineer Jonanthan Helfand. Having been early on the Fairlight MFX platform, Helfand's commitment to the MFX3 in the two completed rooms with an MFX2 scheduled for the third comes as no surprise. Already in the can are a short film, and a number of promotional spots for cable and radio programmes. One of the MFX3 rooms also doubles as a music recording room.

Fairlight, US. Tel: +1 212 481 3456.

■ UK-based Fon recording studios has seen the completion of a £0.25m refit to become The Steelworks. Under the control of the Eliot Kennedy, Tim Lever, Mike Percy songwriting-production team, the 3-room facility now houses two identical studios containing 32-channel Soundcraft Ghosts consoles, Alesis ADATs and Logic Audio software. The third studio, fed by the other two, houses a 40-channel Amek Mozart console.

The Steelworks, UK.
Tel: +44 1742 754644.
Soundcraft, UK.
Tel: +44 1707 665000.

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

6–9
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Fax: +44 171 240 3724.
Email: show@ibc.org.uk
Net: www.ibc.org.uk/ibc/

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Beijing International Conference Centre. Beijing. China.
Contact: China Exhibitions
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Fax: +44 171 546 8570.
Email: info@chinex.com

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Contact: MesseMünchen Gmbh
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Net: www.cinec.de

26–29
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20–24
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Email: main@admt.ru

27–31
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Fax: +91 22 215 1269.
Email: saicom@bom2.vsnl.net.in
Net: www.saicom.com/broadcastindia

4–5
23rd Sound Broadcasting Equipment Show (SBES)
Hall 19. National Exhibition Centre. Birmingham. UK.
Tel: +44 1491 838575.
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Contact: Digital Media International
Tel: +44 181 995 3632.
Email: digmedia@atlas.co.uk

20–23
20th Tonmeister Convention
Municipal Hall. Karlsruh. Germany.
Contact: Ernst Rothe. Bilungswerk des VDT
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Fax: +49 2204 21584.
Email: vdt@tonmeister.de

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Intex Shanghai. 88 Loushanguan Road. Shanghai. China.
Contact: Marcus Bernie. P&O Events.
Tel: +44 171 370 8231.
Fax: +44 171 370 8143

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

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TC UNIT·Y

SIGNAL PROCESSING ENGINE

PLUG-IN CARDS FOR YAMAHA O2R

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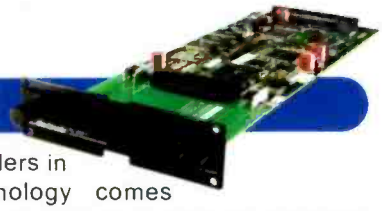
TC INSIDE



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Shooting waterfalls

I WOULD LIKE to thank you for greatly extending the information provided with your speaker reviews. While I wouldn't consider them 'definitive' by any stretch of the imagination, they are a significant improvement over anything I have seen in pro-audio journals for a few years.

However, I wonder why you have neglected to provide a full waterfall plot. You show an impulse response, so clearly you have a way to generate an impulse and measure it with sufficiently flat phase and frequency response to give an accurate and significant time domain impulse plot. If this is the case, then it should be relatively trivial in this day and age to derive the waterfall plot from the impulse response. And if it's not the case, then you shouldn't be presenting the time domain impulse response anyway.

The waterfall plot is a very convenient way of presenting a lot of information and goes as far as anything will toward displaying on paper a representation of how the speaker will sound (if such a thing is even possible). Many of the hi-fi magazines have presented them as part of reviews for quite a few years, although they haven't always been well-conducted or as accurate as they might be. It would be a good thing to see real live accurate data on studio monitors and on PA speakers rather than the smoothed and incomplete curves that manufacturers have taken to providing today.

Scott Dorsey, US

Keith Holland replies: Thank you for the encouraging comments. I agree that our reviews are perhaps not 'definitive' as it is impossible to measure everything and, even if we could, how would we display the results? The aims of this series of reviews are to measure the objective performance of a wide range of loudspeakers under near identical conditions as accurately as possible and to present the results in a way that permits meaningful comparisons to be drawn between them. The first two points are really just a matter of deciding what to measure and then proceeding with care; the final point is the one that is open to the most debate.

As the tests have progressed, it has become clear that, although the method of presentation reveals a lot of detail and differences between the loudspeakers, there is a fair degree of overlap between the various plots in terms of what they reveal. For example, the overall power response does not appear to provide any information that is not available from the directivity plots. Your suggestion that we display waterfall plots is a valid one, and I am confident that our measurements are conducted to sufficient accuracy to yield meaningful plots (the step response, acoustic centre and power cepstra results are actually all derived from the high resolution frequency response measurements). I propose to generate waterfall plots for all of the loudspeakers tested to date and see if they reveal any interesting information that is not readily available from the existing plots. If this proves to be successful, waterfall plots will be published in future reviews—probably in place of the above mentioned power response plots.

It is my experience that some people consider waterfall plots to be useful while others do not; I sit on the fence, for now...

Keith Holland, ISVR

Answering service

I REFER TO Robert Marino's interesting letter (Studio Sound, June 1998) claiming an alternative inventor for the telephone. While I have carried out some research on Bell, I had not come across this one before. I do, however, have some reservations about Mr Marino's assertions.

If they are true and documents really existed to the knowledge of the US patent office describing a device which would read onto the claims of Bell's patent, these would have constituted prior art and Bell's patent would not have been granted. However, Bell's patent was granted and so the US patent office must have been satisfied with its novelty and Bell's priority.

In the case of any invention which proves successful, a queue invariably forms brandishing prior patents claiming to represent the same thing—as we have seen recently with compact disc. Bell's patent was certainly no exception. The patent was filed in 1876 but the litigation didn't end until 1887 when the Supreme Court decided in Bell's favour.

I am sure that anyone seeking to undermine Bell's patent would have searched very diligently for prior art and if Meucci's documents were in this category I am sure that they would have been used against Bell.

John Watkinson, UK

Back to stereo

It is impossible to open a copy of *Studio Sound* or any other trade magazine without mention of DVD. Much is being written about it, all from the perspective of technical issues (data compression systems *et al*). Am I alone in seeing a potential nightmare scenario for music-only applications in what is the actual business end of this, the end-user, or Joe Punter as he is affectionately known, he who eventually pays our wages.

Stick a pin in a map at random and check out the stereo system in Joe's living room. Chances are you've got one speaker on a bookshelf, one on the floor at right angles to the other, one blown tweeter and a 50-50 chance of being in phase. All of this is before the smiley curve in the graphic.

Now let's give Joe five speakers to play with, the possibilities are endless. Even if he's a good boy, how wide does he have his left to right spread? As wide as the TV for movies or eight odd feet apart for music.

So how to mix an album for DVD release. Well I'd be daft to use the rear channels for anything serious, reverbs perhaps or the odd effect. I can't rely on even a half decent 5-speaker system to have anything other than junk at the rear, and I certainly can't rely on placement at all, real living rooms vary in size and shape almost infinitely. So let's forget those.

Now I'm down to LCR. Well these are not time aligned, chances are that they're in a straight line, in other words the centre leads in time. Even if they are on the correct circular arc there's only one point they're true for. So, like cinema systems, panning is out. There are 3 positions only available: left, centre, right. Classical mic techniques (coincident pair m-s) could be adapted for 3 channels, but lack of time alignment makes the centre channel completely unusable.

So (assuming I'm making a non-classical record) I've only got the three LCR positions to use. Check out the majority of records, vocals in the centre, drums in the centre, bass in the centre. In reality all but 1 or 2dBs of the total level is in the centre. Now we're going to put all of this in one speaker instead of two. The result, Joe's system craps out quieter than it used to with regular stereo. If compromised centre channels happen this gets even worse. We can solve this by using a phantom centre from left and right of course. Hang on a minute, that's 2-channel stereo! Oh and I get my panning back, and my time alignment. Hey this is good.

Quite possibly DVD will prove to be as popular and long lived as quad. If not we at the hot end of mixing desks need to think about these problems seriously if we don't want to end up with egg on our faces. It's likely that we could make some very expensive albums which turn out to sound like rubbish on the majority of systems.

Andy Jackson, Freelance producer-engineer, London N8, UK

NY shuffle

My name is Marc Rusch and I am the engineer for CIMP records in the US. We make only live, unedited, unmixed, unprocessed recordings using a single stereo pair of mics.

In the July 1998 issue of *Studio Sound* you ran an article on page 92 by John Watkinson called Stereo Mic Techniques. He describes a 'shuffler' circuit for monitoring intensity stereo feeds via headphones. This problem has been a source of frustration to me for some time and I have been looking for some sort of solution. I have heard of such things before but have never been able to track down a schematic for a design. If anyone can provide me with a direction that might lead to a schematic or a product I would be grateful. BTW, I should also mention that the high-quality of not only the tech sections of your magazine but the editorial section as well, is much appreciated.

Marc D Rusch, Cadence Building, Redwood, NY ■

are you ready for this?

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DAR Scalable Audio Multitrack

Meeting the expectations of film people for the uses of digital dubbers is an increasingly exact art. **Rob James** revisits DAR's dubber to discover an extended concept

ANY MANUFACTURER who manages to produce a successful design for a digital dubber will reap a rich harvest of spin-offs into the rest of the market for DAWs. A digital dubber looks deceptively simple. In reality it is nothing of the sort. As a consequence, a truly successful dubber design provides a solid foundation for almost any other application you can think of.

When I reviewed DAR's digital dubber offering, the OMR-8, I was less than impressed. There were a number of problems with the machine that needed urgent attention, notably in the area of multiple consecutive punch-ins and punch-outs. I am happy to report DAR has been far from idle over the last few months. One of the early purchasers of the machines was Granada facilities who bought them to complement their existing DAR workstations. The experience gained through the use of the machines in film and television dubbing has been put to good use and the problems I encountered have been addressed. The result is a well-designed machine that now acquits itself well as a dubber.

But this is far from the end of the story. In addition to sorting out the early teething problems DAR has taken the basic building block of the dubber and extended the concept in a number of interesting ways. For a start multiple machines can be locked together to sample accuracy using a proprietary clocking system. This kind of accuracy between machines is not really required in a film situation where the previous technology could only manage around a quarter of a frame at best, but it is absolutely crucial in music work. The theoretical maximum number of tracks is a massive 1,024, which would require 128 machines. So far no one has had deep enough pockets to specify a system of these dimensions, but if you want to get into the record books now is your chance.

The really clever stuff starts here. One machine is specified as a SAM master or interface unit with additional slave machines specified as SAM processing units. In addition to the proprietary sync connection, and conventional audio and control, the machines are linked to a network hub using (for the sake of interest) 100-Mbit Fast Ethernet. The technology employed is irrelevant to the user and can, in any case, be upgraded as better things become affordable. The point is what this 'networking' offers the user in terms of functionality. In this case audio transfers across the net are typically 10x faster than real time. This is one of the factors that enables SAM to appear as a single machine to the operator. The SAM interface unit is provided with a screen, QWERTY keyboard and mouse or trackerball. This is used to display all the tracks from all the connected machines as if they were on one piece of tape. Transport control, track arming and recording can all be accomplished using on-screen buttons in addition to the hardware buttons on the individual machines. So, SAM is a virtual multitrack with high quality 'reel-rock'.

I have no doubt many potential music users will be wondering where the hardware auto-locater has got to. DAR is well aware of this requirement and are working on it.

SAM is also a well-specified workstation. All the editing tools you might expect are provided. Not only on the normal 'segment' basis, but also a timeline that can encompass parts of segments across multiple discontiguous tracks and also cross 'machine' boundaries. Where an editing operation requires a move from, say Track 1 to Track 9, the actual audio will be copied across the network to the machine which has track 9's output. This is done rapidly and intelligently so, if the audio already exists on the target machine due to an earlier edit operation, it will not be copied again. All of this happens with little or no user intervention. The user's perception is that of editing using a single, potentially massive, multitrack machine. The timeline or 'region' editing enables a single audio event, which may be made up of several segments or parts of segments across several tracks, to be treated as a unit and copied, pasted, and so on, many times without taking up disk space rerecording audio.

DAR's Table function provides a separate recording and editing area to the main 'multitrack' reel. Material can be recorded in Table mode and pasted into the reel. Once on the 'table', material can be auditioned and edited before pasting back into the reel. Slipping tracks or segments, changing the sync of any SAM track or tracks, can be done from the front panels of the individual machines or from the master.

If all of this still leaves you wanting more, there is more. In addition to the normal 16-bit, 44.1kHz or 48kHz sampling rates, bit depths of 16, 20 or 24 bits per sample are standard and sampling rates of up to 192kHz can be specified if you can afford the convertors. Still want more? Well DAR have long been admired for the segment-based 4-band EQ on their DAWs. SAM can play back material prepared on other DAR workstations with all the EQ applied in real time. And, of course, you can adjust it using SAM.

The number of tracks that can be simultaneously displayed is limited only by the resolution of the monitor. On a modest 17-inch screen with 1,280 x 1,024 pixels around 32 tracks are comfortably viewed. I would have preferred to see a greater contrast between the blue segment blocks and the blue background, but this is purely my taste and in any event, easily changed if enough people agree. There are two main displays, the multitrack window as just mentioned uses most of the screen to display tracks with the remainder containing transport controls and a counter. The other screen is used for editing. The waveform display comes in two flavours, envelope at low magnification and sample accurate waveform when zoomed in to allow surgical precision editing.

DAR have been having a very busy time since the company was acquired from Carlton by managing director Mike Parker. As well as developing SAM the whole product range has been con-

32-track display of SAM with waveforms, fades, marks, and a region outlined as editing

verted to a new disk format, Genesis. This is based on the industry standard Microsoft / IBM .WAV format and is compatible with OMFi, AAF and the putative AES 31, not to mention BWAV. All files used internally are in standard WAV mono format, but can be exported in other formats. The disks are now formatted in MS-DOS 16 or 32 bit FAT with 2Gb 8Gb partitions. By taking the radical step of adopting the computer industry format used in 95% of PCs, DAR have positioned themselves admirably to take advantage of the rapid advances in mainstream PC technology. Fixed and removable hard drives, magneto-optical drives and even Jaz (if you insist) can all be accommodated together with Exabyte Data DAT or DLT for archiving.

Each SAM unit is a 3-unit-high 19-inch rackmounting or free-standing box. The front panel is finished in DAR's house colour of charcoal grey. The most prominent feature is the meter display. Eight bright orange bar graphs with over-mod indicators and select buttons for recording, editing and locating. Below the meters are 7 SETUP panels which are used to enter and display all important machine settings. To the right, arranged vertically, there are seven function keys. RESET resets the overload LEDs, TC ITEM changes the time display between time code, 35mm feet (units of 16 frames) and 16mm feet (units of 40 frames). MEM 1 and MEM 2 are scratch-pad locator memories. Pressing either key loads the memory with the current location. AGAIN locates to the last point at which PLAY was pressed and plays from that point. LOOP does the same thing except when you arrive at the point at which the LOOP key was pressed the section repeats until another transport control is used. The final key selects what the stereo Digital I-O addresses. Transport controls are on the right and with two exceptions are tape-recorder style. The exceptions are a reverse sync play key and the PARK key. All the transport buttons are illuminated. The chunky control wheel has good feel and is used in conjunction with three associated buttons to provide audible reel rock or audible speed control. The audio quality in both these modes is excellent. The third key is POSITION which enables you to adjust the currently displayed time code value. A fourth key in the same group enables Gate recording. The time code display also conveys other information depending on mode. Beneath the display are seven illuminated edit keys which allow marking of in and out points, ERASE, OVERLAY and COPY together with UNDO and a SHIFT key that gives access to Cut, Insert and Clean functions. An illuminated mains switch, headphone jack and volume pot and a selector switch for MIX, MONO or SELECTED TRACKS complete the front panel controls.

The expanded screen showing 8 tracks with waveforms and names, with a region marked for editing

Directory and assembly page of OMR8/SAM with real-time waveforms being generated during record

On the busy rear panel are a plethora of connectors. Sixteen XLRs take care of analogue I-O, convertors are 24-bit oversampling types. Four further XLRs deal with LTC I-O and AES-EBU I-O. All eight outputs are optionally available in AES-EBU format on a D-connector. SP-DIF is also provided on the usual phonos. The stereo AES and SP-DIF I-O are able to be selected to pairs of tracks or to carry a mix of all eight. This can be looped through additional machines to give a mix of all tracks. Further D-connectors cover biphase, MIDI or IBM compatible serial mouse, RS-422 and external VGA monitor. RS-422 provides Sony P2 protocol (9-pin) remote control capabilities. Functions remotely controllable include, track arming, chase sync control, transport commands and disk presence indication. Video reference and AES wordclock I-O are on BNCs. An IBM compatible computer keyboard can be connected to an AT-style DIN socket and there are further connectors on the panel for options such as SCSI, GPIO, RJ-45 networking, and a remote front panel. GPIO can be used to control the transport functions, plus record arming and tallies.

The gated record mode allows a threshold to be set to trigger recording, a minimum record time after a trigger event and the minimum spacing between trigger events. Two triggers closer together than this parameter setting will result in one recording covering both events. To avoid mishaps and clipped recordings the segment is actually started before the gate opens. If this sounds implausible, a bit like a time machine, it is achieved by continuously 'recording' into a rolling RAM buffer and retrospectively setting the point at which the recording starts on disk.

As well as the obvious music and sound-for-picture applications SAM is also well suited to live recording. In the event of some unfortunate occurrence, for example tripping over the mains lead 20 minutes into a recording, the most you will lose is the last 20 seconds. SAM also forms the basis of DAR's theatre system.

SAM is a textbook demonstration of a thoughtful, modular approach to digital recording. In time the innovations in SAM will find their way back onto the company's other DAWs via software upgrades. If DAR so choose, their entire future range of products could be based on SAM modules with application specific hardware and software front ends. This, then, is the realisation of the promise inherent in the successful design of a digital 'dubber'. With SAM and Genesis, DAR have effectively reinvented themselves. They have managed the extremely challenging trick of not only keeping up with the market and their existing user-base, but continuing to innovate. This can only be to the company's benefit, their customers benefit and, ultimately, everybody else's as other manufacturers get the point. ■

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SADiE 24-96

Aspirational audio recording makes 24-bit, 96kHz digital standards an essential consideration. **Dave Foister** evaluates Studio Audio & Video's 24-96 workstation

STUDIO AUDIO is, perhaps, one of the best examples you will find of a company that listens to its users. One of its biggest selling points is its policy of free software upgrades for life, which even included the major rewrite that was SADiE version 3. The frustrations of its extended gestation period were compensated for by the capabilities of the finished product, and now that v3.04 has been reached the stability is such that few users still cling to v2. The current version incorporates most wish lists in terms of project management, functionality and signal processing power, with a family of plug-ins already growing, and support for customers' problems, large or small, genuine or self-inflicted, is unrivalled. SADiE has more than kept pace with the competition in terms of software power, and although its corresponding hardware was developed several years ago it has done a remarkable job of supporting the increased workload. Now, however, that workload and the needs of users mean it's time to move on.

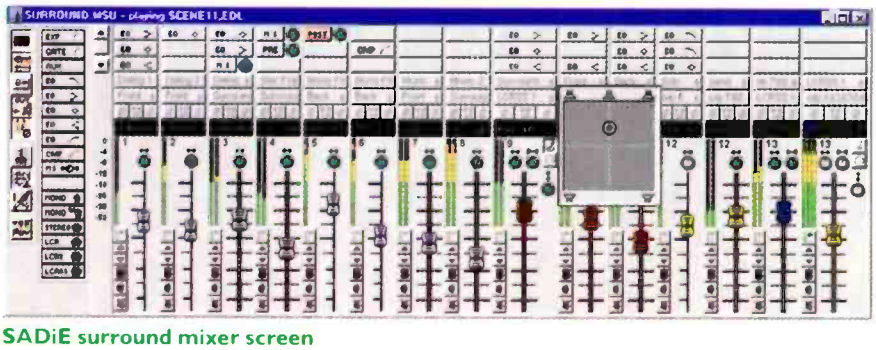
SADiE has run since its inception on dedicated hardware, the XS card, whose physical I-O consists of two inputs and four outputs. 24-bit digital sources have been supported for a long time, but analogue can only be converted 16-bit, and the sampling rates are restricted to the conventional set with no facility to handle the doubled-up rates. It is also true to say that the new software's palette of processing has outstripped the card's ability to deal with the demands, so that even a simple stereo EDL with any complexity of processing—reverb and a couple of EQs, say—can occasionally lead to error messages as the DSP capabilities are exceeded. In fact even scrubbing can cause hiccups if there is a lot of processing bolted on. All these are areas that other workstations have addressed, and the need for more I-O (for surround among other things) plus the enthusiasm for high-resolution operation has prompted a hardware redesign every bit as radical as SADiE 3.

The new system sets out its stall succinctly with its very name. The system is called 24-96 for obvious reasons, and will in fact handle 192kHz for those who can deliver it; although it would be a mistake to assume that the high-res compatibility is the only benefit. Equally important is the increase in physical inputs and outputs: now a single card provides eight of each. The standard breakout box carries these in AES-EBU and unbalanced analogue forms, and like the XS the analogue output is restricted in its headroom by the computer power rails. New for 24-96, however, is an optional active break-out box with full-headroom balanced ins and outs for all eight channels.

The XS card is ISA, and while SAV has always supplied turnkey systems in standard PC towers it has been common for users to assemble their own. The 24-96 card is PCI in line with current trends, and while it is possible to fit it into another PC, SAV now supplies rackmount computers with enough cooling to deal with three cards as well as all the associated drives. The potential for heating problems if multiple cards are fitted into a conventional case is stressed by SAV, and, indeed, the benefits in terms of ruggedness and convenience of a rackmounted PC make it appealing anyway.

Perhaps the most obvious and important benefit of the new system to existing SADiE users is the increased DSP power. It's always difficult to quantify exactly what a given amount of DSP can deliver, but a rule of thumb here is that a 24-96 system should run up to 2½ times as many simultaneous audio tracks as an XS. 24-track operation, and more, is no problem at all, with the expected trade-offs as more demanding processing is added.

Early converts to the new hardware will have been supplied with the existing v3.04 SADiE software, and it's important to realise that the software will continue to run on both platforms for the foreseeable future. It should even be possible to set up a project on a 24-96 system, put the hard drive into an XS system for further work (with the unsupported options simply greyed out), and



SADiE surround mixer screen

then stick it back into the big system for completion. By the time you read this it is expected that v3.6 will be available, and this, while still compatible with the XS card, adds facilities that exploit the potential of the new hardware. The main example of this is the addition of surround panning on the mixer.

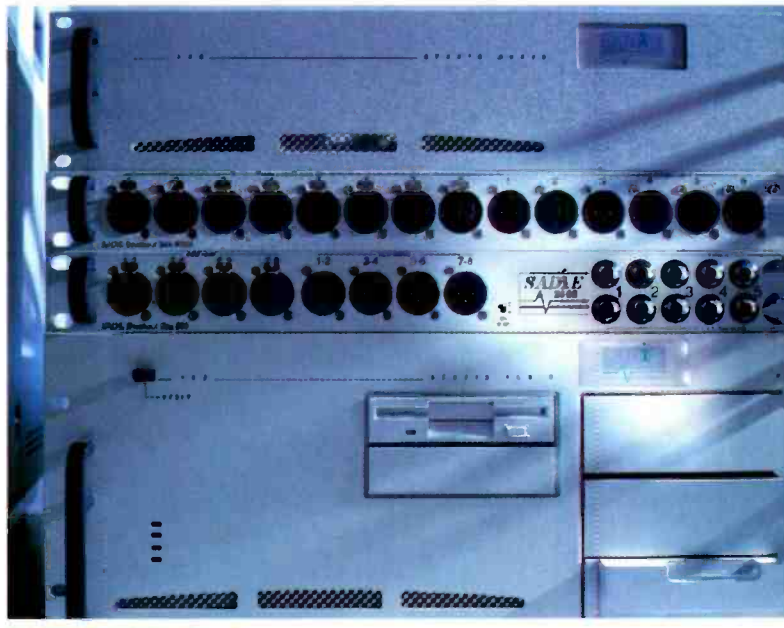
SADiE's mixer is fully user-definable, allowing as many channel strips to be assembled as required, with flexible routeing including aux sends to internal processes or external outputs. The channel strips come in several flavours, allowing many configurations of mixer to be built up, and now it's possible to incorporate various types of surround channel. The most basic is a mono channel that can be panned to LCRS, and this can be used with the existing XS card as it has four outputs. With more outputs available the 24-96 can give stereo surround, with full panning around the five channels. Separate virtual pots appear for left-right and front-back panning, and for divergence, and the whole lot can be controlled with a virtual joystick. In this mode a square grid appears with the five loudspeaker positions marked, and the source is simply dragged around the space with the mouse. Its response is fast and precise, and, of course, SADiE's mix automation covers all of this in exactly the same way as everything else, with real-time dynamic automation or snapshots. For full-blown surround mixes, various combinations of surround channels can be set up, including 5-5 channels where a premixed set of five streams passes through a single fader while spots and other tracks are mixed in from other channels as required.

Although the screen display for the panning looks very much like that used in Dolby Surround Tools for TDM, no Dolby encoding has been incorporated into SADiE on the assumption that users may well already have the external hardware for this. SADiE delivers discrete surround pure and simple, and what you do with it after that is up to you.

Another brand new feature on v3.6 is an additional editing mode called Region Editing. This is intended to simplify the job of editing multiple streams or tracks simultaneously, and is obviously targeted at surround mixes among other applications. Previously there was no easy mechanism for cutting across several streams and then trimming them identically, maintaining sync throughout the track, but Region Editing addresses this with a whole new toolbar and several new tricks.

The feature that brought most excitement to the SADiE office, and which will hearten old SADiE hands, is literally the smallest: the return of sample-level waveform editing. Version 2 was able to show full detail of the audio waveform when fully zoomed in, but the facility disappeared in version 3: clearly someone has been on a mission to restore it and now it's back. Along with the big additions come some smaller ones including faster access to a wide range of commonly-used functions from the status bar at the bottom of the screen.

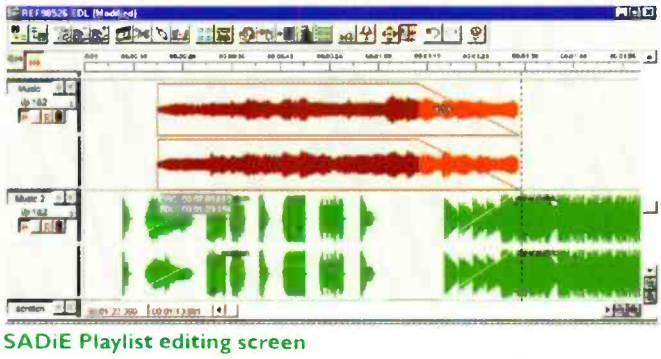
While the move to v3 entailed a little relearning, it has brought a commendable consistency to the various SADiE-related platforms and add-ons. SADiE now looks virtually indistinguishable from Octavia, and, of course, now shares its I-O capabilities. Further proof of the consistency is the fact that I didn't immediately notice that the system I was looking at had the Portia video hardware added. The new rackmount case has room inside for the Portia card, and an associated hard drive, allowing video to be transferred into the system and integrated seamlessly with the audio operations. An extra stream appears at the top of the screen for the video signal, which can be scrubbed, marked with hot spots and cut exactly like an audio track. This is not new, but the addition of the 24-96 and its surround facilities makes this streamlining even more attractive for postproduction work. The new Region Editing has particular application in this context as it greatly simplifies cutting out or adding scenes, and generally shifting whole multitrack chunks around with the pictures.



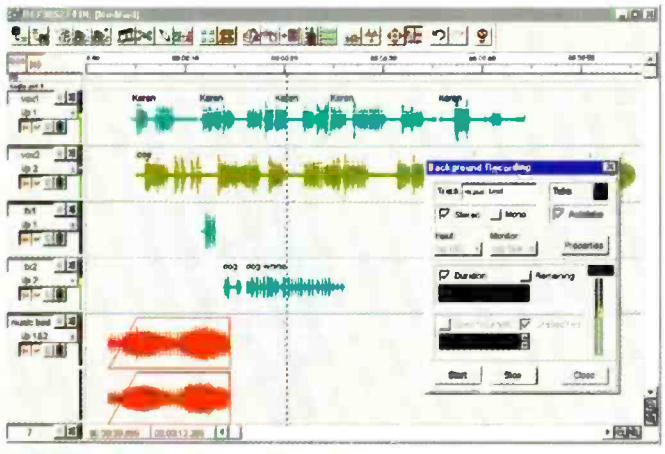
SADiE Portia half-rack

The thrust into pictures does not stop there: SAV's drive for more helpful file transfer and interchangeability systems has already resulted in SADiE's ability to work directly on Lightworks files simply by plugging in the drive with the Lightworks project on it.

SADiE's success story so far is well known. A wide user-base around the world will be further enthused by the new developments, and new areas could be opened up by them. Those applications where a SADiE system might not have been the first choice before may have to think again, as the two quite distinct growth areas of high-definition audio and sound for pictures are addressed so powerfully by the new hardware and software. ■



SADiE Playlist editing screen



SADiE background recording screen

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Sonosax Stelladat II

Long awaited and generously featured, Stelladat II comes with extravagant claims for its significance to the location recordist. **Neil Hillman** puts them to the test

COULD THERE HAVE BEEN a greater leap in audio technology since the ability to record and reproduce sound was made possible, as that from analogue to digital? Certainly, the wire recorders of the 1940s that evolved organically into the tape recorders of the 1950s, with the subsequent development in the mechanics of head design and the addition of noise-reduction systems, ensured multitrack recording reached a highly acceptable level—think of the priceless recordings that have been made this way—but the ability to record and manipulate a signal without losing the original spirit of the sound along the way remained an elusive struggle until the golden dawn of the digital revolution.

This new age brought a ray of hope, and promised the freedom to take a recorded signal to tape, process it at will, and while remaining in the digital domain experience only imperceptible degradation from these previously lossy multiple signal paths, and faithfully reproduce what was laid down on that tape.

But what of that input stage to tape, the encoding process; if what came off the tape bore little resemblance to the original sound being made who really cared how clean subsequent stages were? The advent of 16-bit, 20-bit and 24-bit, 96kHz sampling has addressed these digital downfalls, and its meritous success is the matter of some considerable debate, not least of all within the leaves of this publication.

Into a DAT market dominated by the HHB PDR1000, and to a lesser degree the Fostex PD4, then comes the second generation Sonosax DAT, appropriately, though not inspiringly, named Stelladat II and offering 16-bit, 4-track 48kHz or 44.1kHz recording or 2-track, 96kHz recording with a power consumption claiming to be far more efficient than its rivals. Do not confuse this machine with the original as the Stella DAT II has been completely re-engineered from the ground up.

With the Stelladat II, not only is it possible to produce a recording of high-quality outside the studio, the location recordist is able to in effect track-lay at source in a manner designed to please the hardest of hearts and ears in post-production: perhaps with four separate and discrete radio mics, or an MS pair with two artistes on personal mics—this is a real gain in flexibility. And with DVD courting 96kHz, Sonosax has positioned its machine carefully indeed.

Routeing aside, the Stelladat II has a very different feel from that of either the HHB or Fostex machines and looks, well, very Swiss with its stark functionality and uncluttered layout. The top face of the machine has a removable, hinged, clear perspex lid covering a deck housing, on the left, half a 4-input mixer, with in-line from the top a 3-position toggle switch to select either 12V phantom, dynamic-line or 48V powering for microphones. Below this, the input gain is a rotary knob marked 0, 14, 28, 42, 56 and 70 (dB) and below that is a 3-way selection switch for the 12dB octave LF cut filters: 80Hz, linear and 120Hz. Below this is the centre-detented PAN pot with the last knob being the LEVEL control calibrated 0-10. A toggle-switch sits below the level control pots of channels 2 and 4 as a stereo link switch and four small overload LEDs sit at the bottom of each channel strip, set to trigger 10dB before clipping.

The right-hand side of the top face houses the tape transport, and its associated control keys; small, fiddly rubber-covered buttons that also double as five function keys (F1-F5), for the software configuration of the machine. Thankfully, the transport keys are duplicated on the front face with larger push buttons. Along the right-hand side of the top face are 3 further buttons: power STANDBY, ESCAPE, this button being used while configuring the machine through the software menu, and tape EJECT, the door being heavily sprung and undamped is reminiscent in use to that of the ejector seat in the Corgi model of James Bond's Aston Martin DB5. Now that I think of it, the polished aluminium tape door itself looks substantial enough to double as a bullet-proof screen.

Those larger, rubber transport buttons are set into nickel-sized recesses with smaller high intensity LEDs sitting alongside each one, in 2 rows on the right-hand side of a wonderfully uncluttered front face that has the 2 main GAIN pots sited centrally, with a tiny grille in between them for the slate-talkback microphone. These big rotary dials are calibrated in dBs from infinity to +10dB, with 0dB marked in bold at the 2 o'clock position. Immediately to the right of the dials are 3 small buttons, mounted one above the other: MON for monitoring signals either directly or off tape, PRG for toggling between blank search being on or off, and DISP that scrolls through the three pages showing machine and ID status.

The left-hand side of the machine houses the 4 inputs on female XLR sockets, electronically balanced with RF filtering, and introducing noise of just -129dBu at 150Ω with 80dB of gain, and directly below these the 4 outputs on male XLR plugs, electronically balanced with DC protection against 48V phantom power. Also on this left-hand-side panel is the monitor selection switch

—to enable the individual monitoring of Left and Right outputs. Channels 1—4, and stereo Outputs 1&2 and 3&4 above the 1/4-inch headphone socket with an impedance range of 50Ω–600Ω; the monitor level pot is sited next to the monitor select switch and below this are 3 toggle switches that switch the MS decoder into the monitor chain, switch the headphones between mono and stereo and switch the headphones on and off.

The right-hand side of the machine is home to the digital inputs and outputs with a row of 3 female XLR sockets for AES inputs of pairs 1&2, 3&4 and time code (the preview machine was non-time-code; although a time-code version is promised for later this summer) with mirrored AES outputs on male XLR plugs below their respective inputs. Time code conforming to the Aaton TC standard for film-camera operations will also be accessible through a 5-pin Hi-rose connector fitted alongside the 2-pair optical in and outs, set below 2 BNC connectors for SYNC IN-75Ω and SYNC OUT-WORD-CLOCK. A slider switch enables these sockets to be set to either wordclock or loop. Two further connectors complete the right-hand panel, an RS422 D-connector conforming to Sony protocol for edit controllers and an accessory Harting socket for remote controlling of the transport.

The bottom right-hand corner of the machine houses a Canon 4-pin plug for an external DC input in place of the NP-1 type battery that is carried cleverly onboard in a spring-loaded recessed slot in the bottom face, making quick and easy power changes possible. There is no battery charging circuitry within the Stelladat due to space constraints, but most recordists who favour the NP-1 would have a separate charger-conditioner anyway. At the opposite end of this bottom face is the built-in 500mW monitor speaker.

The main menu offers complete access to all of the machines variables and navigating through the configuration software is very straightforward; the length of time spent in this state is due directly to the fact that the options are so comprehensively, perhaps overly, available to the user. The menu is opened by use of a small joystick on the bottom right of the top deck. Clicking on by pushing the stick down causes the signal metering LCD on the left-hand side of the front panel to show the menu options, then up, down, forward and back instructions are easily actioned by moving the joystick appropriately.

The configurations are accessed through 4 main menu headings: ID, Setup, Configuation, and Power.

The Stelladat II is certainly a heavyweight —and not just in terms of cost at around £8,000 (UK) for a non-time-code machine and around £10,000 (UK) for one supporting time code—but also in specification and physical terms. In operation, however, the depth of complexity is shed and the machine becomes transparent in the recording process, so easy it is to use; controls fall to hand logically and the reknown limiter is happily still as reliable as ever. The claim from Sonosax is that to switch from stationary recording to over-the-shoulder documentary-style simply requires the recordist to plug in a mic, pick up the recorder and run. I would ask you to ensure that your osteopath could arrange a home-call shortly after you attempted to do this. Its natural habitat will undoubtably become that of the feature-drama trolley environment or for serious location music recording. I never really would anticipate using the in-built mixer other than as track-send level controls, but I would suggest that we are seeing the emergence of a high-end market leader; indeed given the modular and robust construction of the StellaDAT II, there should be no reason why the processing chassis should not support other transport formats such as MiniDisc, PCMCIA or hard disk.

Comrades! I bring you news that the glorious digital revolution is far from over. As George Orwell's delightful pig Napoleon might have noted, 'all DAT machines are created equal, but some are more equal than others'. He might well also have been heard to grunt, '2 tracks good , 4 tracks better.' ■

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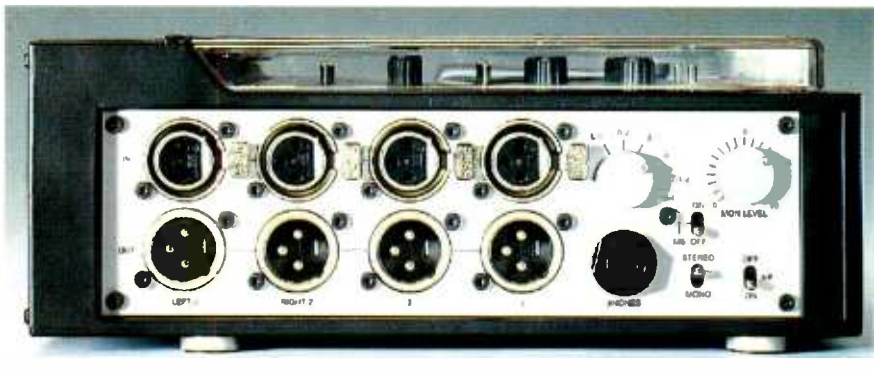

REVIEW

Eventide DSP 4500

A limited edition Harmonizer without limits, a neat science toy, or the next Ford Cortina Crusader special model? **George Shilling** meditates on the stuff of engineers' dreams

THE EVENTIDE NAME has become synonymous with the studio pitch-shifter in spite of the fact that the American company's more recent designs include many other types of effects. Probably the most widely known Eventide unit is the H3000 and its variants—these wonderful units contain a variety of superb effects and did much to establish the pre-eminence of the Harmonizer. Although two versions of this unit remain available, Eventide introduced the DSP4000 several years ago, as a truly top-flight effects mainframe, borrowing many features of the H3000 and expanding on them. Specialist versions for broadcasters (DSP4000B) and guitarists (GTR4000) have since been introduced with tailored presets. The DSP4000 has not proved as popular as the H3000 in commercial UK studios—perhaps this is because of the high price, or the specialist nature of the unit. In my early encounters with it, I found it less friendly than the H3000, with an irritating, complex, cold blue display, occasional hourglass waiting times and even a couple of software bugs.

Fortunately there are no obvious bugs in the DSP4500 Limited Edition, a special model, that brings together all the presets from all versions of the DSP4000, and includes a new library of 225 presets, by Scott Gilfix named Alchemy 101, which are also available on a memory card for DSP4000 owners. In addition, an 8"-second sampler board is included. The front panel is painted blue and printed with Limited Edition legending, and all this is included at a much lower price than if you were to buy a DSP4000, all the memory cards and a sampler board.

Physically, the DSP4500 is a bulky 2U-high device with a solid, professional feel about it. The front panel has many similarities with the H3000, but looks slightly cluttered in comparison, with buttons scattered across it. A similar large knob-cum-wheel, the keypad, four softkeys and the bright twin LED level meters are included. The LCD is much larger, but uses a smaller font. In addition, there are USER 1 and USER 2 buttons, a PATCH button, CURSOR and SELECT buttons, and the PCMCIA memory card slot with EJECT button. On the back, the two analogue input sockets are of the dual-type, which will accept XLRs or ¼-inch jacks. Two normal XLRs provide analogue output, while digital connectors are available on XLR and phono which can simultaneously be set to AES-EBU or SPDIF with Input and Output formats separately menu-switchable. There are a number of blanked panels, suggesting potential future expansion, and a space for the optional fitting of an RS232 PC serial interface or a Sony 9-pin RS-422 VTR interface. Three jack sockets are provided for Footswitch and Footpedal inputs, and Relay Output to control another unit's footswitch input. MIDI In Out and Thru 5-pin-DINs are present with extensive MIDI implementation, and a fuse-carrying IEC socket with a switchable voltage selector rounds up the back panel.

At switch-on there are reassuring relay clicks, and the unit goes through a self-test routine. The 1000+ preset programs are divided into grouped banks by type, each bank containing roughly between 3 and 30 presets in alphabetical order. However, due to the sheer quantity of presets in this special edition model, similar programs or types are sometimes spread over a number of different banks. I waded through nearly all of the presets during my time with the unit, and was occasionally surprised to find certain programs in unexpected places. Unfortunately, Eventide lags behind Lexicon and tc electronic who have made an effort to provide some sort of 'search by type' facility in their latest models, which would certainly have been useful here.

The DSP4500 comes with uprated 24-bit A–D and D–A convertors, which sounded great, boasting extremely low distortion figures. However, in a normal +4dB environment there was not always enough Input Gain to make full use of the large headroom allowance. Pressing the LEVELS key brings up the necessary settings, but with Inputs set to a maximum 0.00dB and Input Gain Trim at the maximum +10dB a 0dB tone only lights the first 4 of the 10 LEDs. Connected to a digital device the output seemed to be set with internal 0dB equivalent to -18dBFS. Sample rate is normally 48kHz, but can be set to 44.1kHz, 44.056kHz or 32kHz. A useful global parameter bypasses the mix control for 100% wet studio use. The ¼-inch jack inputs are nominally -10dBm. Impedance is not quoted, but these seemed perfectly happy to accept a signal directly from an electric guitar, with just about enough gain.

The PROGRAM key brings up a screen where you can see the current program bank title and scroll down the list of programs. Turning the knob further at the bottom of the list will tip you over the edge into the following bank. At the bottom of the screen the 4 softkeys' functions are displayed, and by pressing PROGRAM again a further 4 functions appear. Load and Delete functions worryingly share the same key, but Factory programs cannot be deleted, and User programs require a confirmation before deletion is performed. User programs can be stored in the Factory banks; although you can create new banks of your own, or save to the memory card.

When your chosen program is highlighted, pressing the LOAD softkey will instantly load the program and display the first page of editing parameters. Alternatively, the USER 1 and USER 2 keys are preprogrammed to load 'next' and 'previous' programs within the current bank respectively, no matter where you are in the menus. This could be seen as a curse or a blessing. It enabled me to quickly run through the available programs, but could end in tears if you were to accidentally press one of these after much fine-tuning of a program. No confirmation is required before program load, and this function is not immediately obvious. Armed with this knowledge you might want to reprogram the User keys. The cursor keys below the wheel takes you through the editable parameters, which can be changed with the knob. The knob feels heavy yet rotates very freely, a slight gripe being, perhaps, the lack of any variation of data entry speed. Sometimes you cannot turn it slowly enough for fine-tuning; other times you wish that parameters would change quicker. In these instances, it is often preferable to enter the exact desired number with the keypad. Most programs contain a parameter page called Info or About where a text message displays hints or useful information for the particular preset. Sometimes, softkeys can access multiple pages of the same name with repeated presses: this is indicated graphically. Like the H3000, some presets contain 'expert' parameter pages for more complex tweaks.

Preset creation and manipulation is achieved from the Patch menu, where graphical representations of the flow of audio and-or processing can be displayed and manipulated. This is where you sometimes see an hourglass while the processor computes. Users can create their own algorithms from the huge number of 'blocks' provided. Possibilities are manifold, but there are such a huge number of programs provided, that there is every chance that your need has already been fulfilled. Even some of the much-vaunted 'artist' presets by famous users are simple modifications of basic presets, where little patch modification

has taken place. A number of presets include the ability to crossfade into other presets, and this capacity is indicated by a symbol on the program page.

Let's parade some of the presets, which have benefited from help and contributions from well-known producers, engineers and musicians at the top of their respective professions.

First come the DSP Studio banks. The DSP4500 as expected performs exceptionally with its Harmonizer programs: there are many imaginative implementations of its market-leading pitch shifting algorithms. There are clever multiple-shift and delay programs for beautiful atmosphere creations or huge textures, as well as rhythmically delayed harmonised repeats. The Utilities bank includes a useful MS decoder and manipulator, and a list of all factory presets. Other preset banks available include the self-explanatory Delay FX, Chorus-Flangers, (tape-phasing programs are particularly good), Dynamics (with gates, compressors and panners), EQ and Filters, and Distortions. The Mixdown Suite bank includes useful pitch 'Whippers' for momentary tuning corrections and the H3000 Emulation bank contains 27 old favourites. There are some (rarely usable) fun things in the Bizarre, Curiosities and Multi FX banks, while Dual FX contains mostly twin mono effects which include reverb. Reverbs are grouped by size and are generally big, smooth and sophisticated in character, with the huge processing power being put to good use in creating a number of gated settings. The larger spaces are exceptionally smooth.

The Big Delay, Big Sampler and Big Loops banks make full use of 87s full-bandwidth sampling capability. Next, the Guitar presets, some of the which were a little irritating in that they required a plugged-in footpedal to work. Also, some of these programs required MIDI data input, for example to control pitch change. It would have been nice to easily be able to use the knob instead, but the creators obviously did not consider Luddite studio engineers like me when creating presets for the GTR4000. Virtual Pedalboard features 21 multieffects programs with MIDI control for the gigging musician who does not want the deafening silence that occurs when switching presets. Each of these contains a variety of effects, many of which feature graphic representations of virtual knobs for level, and mix, and so on. Notable are Trey Gunn's presets, which are characterised by heavy compression with a touch of distortion and atmosphere. Artist Bank A includes EMT 140 with a decay of up to 1000s, and AMS dmx1580S that includes the smooth Eventide Harmonizers, but lacks the warm grunge of the original device that still sounds great to me. Broadcast and Production banks include a number of sound effects, which require no input, some good, others unrealistic and a waste of time. Many of these have an overwhelming American slant. There are all sorts of telephone and radio emulation, hum and hiss elimination, and the remarkably successful Solo Zapper, which gives an instant karaoke version of the inputted mix. No, really, it works (in a way).

Finally, the real treats: Scott Gilfix's Alchemy 101 banks. These are mostly alternative takes on the DSP Studio banks, but put together with a huge dollop of imagination, and a good grasp of what goes on in studios. A chromatic tuner pops up in the Utilities Too bank, along with a metronome and the tongue-twisting Tap Tempo Template. Delay Based, Verbs, Dynamics and Filters banks all contain novel and useful presets, and there are loads of sample-looping templates. The guitar effects provided on the Preamps-Pedalboards are mostly set up with overly expansive effects, but the Preamps-Fuzzpedals and Preamps-Overdrive banks include some great settings.

Overall, there is something for everyone—collaboration between manufacturers and end-users is evidently a good thing. One problem in an analogue environment is the convertor delay, that can affect phase if you mix any dry signal externally, but this does not affect digital-domain users. Eventide describes itself as purveyors of 'neat science toys', but this sounds almost too flippant to describe the DSP4500, which is a complex tool for the serious professional. However, it sometimes seems operationally more scientific than musical and friendly. While there is no doubting the quality of the effects achievable by this box, some of the operating architecture seems a bit clunky when compared with the latest Lexicon and tc units. And at this price, why is there no dedicated remote?

The distributor was cagey about how truly 'limited' this Limited Edition might be, but it will apparently be available for approximately a year. The DSP4500 contains some outstanding audio manipulation tools. However, when investing such a large amount of money one has to consider whether this Limited Edition is possibly a last-gasp DSP4000, a Ford Cortina Crusader special model, perhaps, before the next generation of Eventide Harmonizer is released. (H5000? Anyone?) On the other hand, those blanked panels may mean this design will be around for a while, with improvements still to come. I hope so. ■

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MediaFORM 5900/Easi-DAT

The shortcomings of using a PC-based system for short-run CD-R copying are well recognised. **Tim Frost** tests an audio-optimised alternative

IN JANUARY 1997, a CD-R cost $5 and a drive in the region of $800, by the start of this year that had dropped to $1 for the discs and under $400 for the drives. These easing prices have created a new product—the desktop CD-R duplicator—which has been taken up by companies distributing reports on disc, producing short-run software projects, or creating quantities of beta-samples of software. The desktop CD-R duplicator can reliably produce dozens of copies in an hour, and is a huge improvement on attempting to make dozens of one-to-one copies on a PC.

The growing range of duplicators has spawned a number of units specifically designed for audio applications —primarily studios and broadcasters who need to produce small runs of work-in-progress and mix copies. US-based MediaFORM has been in the duplication business from the early days of diskettes and moved heavily into 'desktop' CD-R duplication several years ago. Rather than produce a completely dedicated audio copier, MediaFORM has taken the approach of developing a range of duplicators and then offering pro-audio functionality as an add-on software-hardware package that can be used with several of the systems. This way the studio can choose the duplicator specification that suits it best and add the audio functions to it.

The unit supplied for review by UK distributor Magellan is the 8-drive 5900 CD2CD tower loaded with the additional Easi-DAT tape import package.

Not having covered tower CD-R duplicators before, it is useful to outline the way the 5900 works as a disc duplicator, before going onto the specific audio functions.

On the outside, it is a PC tower with 8 CD-R drives topped by a simple control panel with a small LCD. Inside, the 5900 is a recognisable PC system. A motherboard with a 686 processor, a hard disk, and a pair of Adaptec SCSI controllers each connected to four CD-R writers.

What differentiates the CD-R tower from a PC system is the way the hardware and software has been fine-tuned just to copy data and write it to CD-R. Anyone who has used CD-R on a PC system will be well aware of the random vagaries that Windows adds to the CD-R writing function. PCs are ill-equipped for continuous and uninterrupted data flows, throwing up record mode problems and buffer-overruns and end up producing more gold drinks coasters than good CD-R copies. The dedicated desktop duplicator, on the other hand, has all these problems designed out of it and is not trying to write reports, do accounts and play *Doom* at the same time.

The 5900's Panasonic 4x drives can copy a completely full CD in under 20mins and if the master has only a few tracks on it, the copying process for 7 discs can take under 5mins.

The hard disk is an important convenience point on any CD copier, as it can temporarily store a complete image of the master CD. If a couple of dozen copies need to be made, the master CD is copied at 4x speed to the 7 remaining CD-R writers, while the 5900 simultaneously writes a bit-for-bit image onto the hard-disk. When the first run is finished all the CDs including the master are removed, and blanks put in all 8 drives, which are then written from the CD image stored on the hard disk.

The process of copying from a CD master is straightforward. Having loaded the source disc in the top drive, the LCD asks whether you want to copy the disc from a hard-disk image or a source CD. Selecting 'Source CD', the 5900 analyses the source disc to work out whether it is an audio disc, or one of the 20 other ROM, video, enhanced, or multisession variations on the CD theme. Having established the format, it automatically sets up to copy in the corresponding format and starts writing. A progress bar comes up on the LCD and when the copying is finished, the drive trays open and it softly beeps.

The whole process is that simple, and works for any type of source disc.

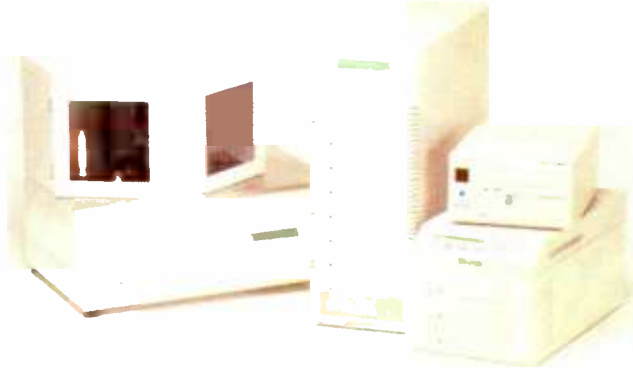
If it is essential to know that the copy is absolutely perfect, there is also a Compare routine, that runs the source and copies and executes a bit-by-bit comparison between the two. If it passes this test, then the copy is guaranteed to be bit-perfect with the original.

The unit as it stands can hold up to 8 CD-R drives, but a network kit can add up to 7 slave units, each containing 8 drives, so that a maximum of 64 discs can be burned simultaneously.

It can be attached to any PC via the SCSI connector, but this offers no real advantage; although the SCSI can also be used to import from an Iomega Jaz drive.

With the generalities of the copier dealt with its time to turn to the audio-specific functions. Even without the addition of Easi-DAT, the 5900 can create compilation masters by importing tracks from different CD sources onto the hard disk and using the hard disk as the source to write to the CD-Rs. The 5900 also allows data to be added to list of audio tracks on the hard-disk, to create an 'enhanced CD' CD-ROM XA master.

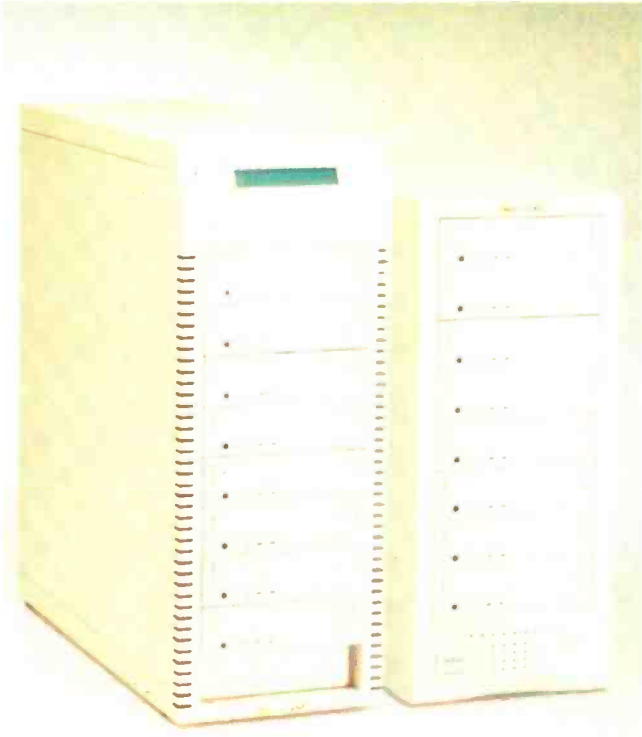
The addition of the Easi-DAT package turns the 5900 into a pro-audio CD-R duplicator that can take digital audio from any source. Easi-DAT adds a soundcard (the unit tested came with Zefiro Acoustic's ZA2) and additional software to automatically handle inputs from a DAT tape.



The card offers a pair of analogue phono outputs, SPDIF and AES-EBU in-outs using a pigtail lead fitted to the 18-pin D-connector. There is an additional analogue input option; although MediaFORM may change to a card that also offers analogue inputs as standard some time in the future.

The digital inputs can handle sampling rates of 32kHz, 44.1kHz and 48kHz. To avoid any digital signal degradation, all 44.1kHz sources are passed straight through the card and are not processed by it.

Easi-DAT adds, as its name implies, the ability to burn discs directly from a DAT master, using the Start IDs to create the individual tracks on the disc. At its simplest level, a finished DAT master is fed to the 5900 via the SPDIF inputs and the copier set up to 'copy from audio' which puts the audio and track information onto the hard disk. The menu asks for the maximum recording time and the maximum number of



tracks—with default values of 74min and 99 respectively, so it is not critical that these be entered beforehand. However, the 'number of tracks' can be used to avoid long chunks of silence being recorded on the end of a disc, if the music on tape is only, say, 55 minutes long. The trick is to put an additional Start ID on the DAT just at the end of the last track. By instructing the copier to import just the actual number of music tracks on the DAT, it will transfer all the tracks and stop when it detects the final Start ID.

The finished hard-disk master is used as the source to copy as many CD-Rs as required at 4x speed (2x is optionally selectable). Tracks can also be played back from the hard disk using a separate menu structure that apes the function of a DAT player.

If the DAT is a working copy, rather than a final master, then Easi-DAT offers manipulation of the material once it has been transferred to hard disk.

Edit-DAT's Start ID editor can individually remove, add or adjust any of the Start IDs. A global Start ID Shift function can move all the IDs back by a selected amount, useful if the tape has been made using Auto ID which inserts an ID as the music starts, but mostly places the Start ID a couple of seconds too late. Audio tracks can be deleted and the system also offers onboard track fade.

Sound quality is excellent. Comparing source to both 2x and 4x copies, the speed appears to make no difference to the sound.

In operation, the 5900-Easi-DAT package does all it sets out to do. The ease, speed and reliability of the operation comes as a huge relief to anyone used to using PCs for burning CD-Rs. In the time it takes to get a PC booted up on and get the CD-R copying software running, the 5900 can already be writing the first tracks on 7 blank CD-Rs. The Easi-DAT offers the key functionality needed to quickly transfer from tape. It does not have a vast range of edit functions, but then it does not set out to be an audio editor. To be picky, the display could be a little bigger to show more menu items at once and the nomenclature on the control buttons changed to make menu selections a little more obvious. But the only major hurdle to stopping anyone investing in such a useful system is simply going to be the £5,000 price tag. ■

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ZONAL 999

The arrival of a new high-output analogue recording tape is an event that will surely raise the pulse rate of enthusiasts. **Rob James** goes behind the scenes of its development

ZONAL is a rather unusual magnetic tape manufacturer. While less well recognised than certain manufacturers, it is the only company manufacturing tape in the UK. The company began by striping film in Acton, West London in the fifties. Nowadays it supplies the film industry with sprocketed magnetic film products across the world. The company is also popular in some areas for ¼-inch tape, indeed Zonal tape is used in considerable quantities by the BBC, and exclusively by the UK National Sound Archive (A section of the British Library). Both organisations have many examples in excess of 20 years old which are still playing without problems. Zonal also makes logging tapes for city institutions and the aviation industry, and the pancakes and shells used by cassette duplicators. But until now it has had a fairly low profile in the music arena. Previously, Zonal has only offered multitrack versions of formulations with a reliable, but pretty average, performance, but with the launch of 999, this looks set to change. Zonal 999, then, is a high output, low noise mastering tape aimed at the music business. If the relative volume of words written over the last few years about analogue and digital recording technologies are anything to go by, any company introducing a new analogue recording tape might as well follow it up with a new feed for dinosaurs before filing for bankruptcy. However, the insight behind the digital ballyhoo is there is a large number of 2-inch analogue machines in use quietly(?) making records and the number of ½-inch 2-track mastering machines is actually increasing. There are also vast numbers of 'narrow gauge' multitracks around that just need the right tape to give them a new lease of life.

Although digital technology has stolen much of the limelight in recent years—in certain applications, justifiably so—in other areas, notably multitrack music production, analogue continues to used side-by-side with digital and enjoys many perceived advantages and die-hard users. Not the least of its attractions is its ubiquity. Just as with 35mm film, a reel of 2-inch tape can be played anywhere from Calcutta to Calgary without any big compatibility worries: something digital users are still dreaming about. In many ways, analogue is still the most future-proof medium, provided the tape is still in a playable state. In 50 years time it will still be possible to construct an analogue tape machine with modest engineering resources, but the same cannot be said for any of the current digital machines.

In rather the same way that picture film technology has been improved to keep its edge over video, analogue audio tape development has progressed over the years to provide higher output, lower noise, less print-through, improved wind and better longevity. The elusive goal is improving all these desirable characteristics without compromising others. Analogue tape has a rather low-tech image, but this is far from the reality. Even in this digital age, sticking rust to plastic is both an art and a profitable science.

The manufacture of magnetic recording tape is a serious industrial undertaking: a tape plant is part chemical works, part engineering shop, and, of course, part production line. Zonal's factory is situated in leafy Redhill, Surrey where technical manager, Melvyn French describes the first stage in the process as mixing the magnetic 'paint'. The iron oxide powder, known in the industry as 'pigment', is mixed with resins, binders and other additives using ketone-based organic solvents as the carrier. These solvents are highly intlammable so the safety precautions are a major consideration and in this enlightened age, precautions against pollution are also stringent.

As in music, the mixing process for tape manufacture is highly sophisticated; it is not simply a matter of adding the ingredients in the right proportions and giving it a quick whizz in the industrial equivalent of a food processor. The magnetic particles need to be of uniform size for optimum results and are susceptible to mechanical damage. Hence the mixing is done using tiny glass balls as 'stirrers' in a pressurised machine that resembles an Archimedean screw.

The plastic tape base arrives as wide rolls of clear polyester, known as webs, that are loaded onto a coating machine. In essence, the object of the exercise is to deposit a consistent layer of emulsion onto the polyester base with all the magnetic particles aligned in the same direction. The coating process is aided by finely grooved and toleranced rollers, and the particles are aligned by powerful rare earth magnets. The longest part of the production line at Zonal (some 50m) is where the tape is dried. Again, serious precautions are necessary to avoid static induced sparks that can cause flash fires, and to enable the volatile solvents to be reclaimed.

The webs are passed through a fur-

ther coating process that adds the backing and then move on to Zonal's latest acquisition, a highly sophisticated slitting machine that cuts the webs down into the standard tape format widths. This substantial investment has been made to ensure azimuth stability. The outer edges of the web are discarded and removed for recycling by a vacuum system.

Within the required standards, the webs can be slit in whatever width combinations are required—for example, it would be possible to make seven 2-inch and two ½-inch tapes from each web, but for a top quality professional product such as 999 the 'outer' ½-inch sections are also discarded. From here the tapes continue to the packaging section and on into the warehouse. Due to careful management and 'just in time' production the finished tapes rarely spend longer than 3 weeks on the shelf.

The great secret of success with this kind of production process is quality control. Constant vigilance by everybody involved at every stage of production is combined with meticulous record keeping and checking. In addition to the more industrial aspects of process control, Zonal has a room full of analogue recorders which are used to ensure the real-world performance of each batch meets the design specification. Keith Harvey, Zonal's R&D manager after working in BBC radio dealing with magnetic media from the users perspective explains what makes the new 999 tape special.

'Conventionally the design and development of a new tape always involves considerable compromise in respect of the various key performance parameters. Improving one, say Maximum Output Level (MOL)—will lead to a deterioration in another, for example print-through. The pigment used in 999 provides big improvements in all areas—MOL, noise and print-through —without the usual compromises.'

And I thought rust was rust, and to improve performance you had to seek out more exotic compounds and processes, such as chrome dioxide or evaporated metal...

Harvey put me straight: apparently the shape and consistency of the crystalline structure of the oxide has a profound effect on performance. It is the number of particles per unit volume that is significant not the weight of magnetic material. The particles must be of near uniform shape and size and as free of 'dendrites' (lumps and bumps) as possible. The uniformity allows the maximum number of particles to be packed into the minimum space. It is possible to improve certain aspects of performance using a more conventional (and far cheaper) oxide by simply making the coating thicker, but this compromises other areas, notably shedding.

Zonal is using the same tried and tested binders for 999 as in its other products. These have been selected for outstanding 'hydrolytic stability' to guarantee long life even under less than ideal storage conditions. This means low 'shedding' and no sticky residues. Obviously, there is a lot more to it than using the right pigment and other ingredients. There are a large number of variables in the way in that each stage in the process is carried out. It is rather like baking a cake—blindly following a recipe is no guarantee of success. It is as much art as science.

To find out whether 999 will work as well in the real world as its claimed pedigree would suggest, I spoke to one of the studios evaluating it. Robin Black of Black Barn Studios in Surrey has been trying out 999 on an MTR-90 2-inch machine. 'I'm very impressed with it,' he admits. '999 seems to handle better on the MTR than other tapes and the wind is very good. It appears to be a very fine product.' Commenting on the analogue-digital debate, Black reckons 'The MTR-90 works hand-in-hand with our Otari Radar. Our clients like to use the editing capabilities of Radar while keeping the master analogue recordings intact. Paul Weller, will only use analogue for recording and we have a lot of other clients who feel the same, especially with these new high-quality tapes.'

Analogue guru Tim de Paravicini, who's EAR Yoshino company makes exotic and highly regarded valve amplifiers among other things, is even more enthusiastic. 'It holds azimuth very well, there are no drop-outs and it runs very clean. It comes back hot and should breathe new life into analogue recorders.'

This augurs well coming from a man who makes and modifies analogue recorders and other equipment for an impressive list of clients including David Gilmour's Astoria studio.

Will 999 be a success? It's early days yet but to judge from this enthusiasm, it should have a good chance. Zonal is a relatively small company with a low profile compared to their major rivals, the mighty Emtec and Quantegy, (BASF and Ampex tapes). I would like to think people in this industry are sufficiently savvy to evaluate a product properly and make their purchasing decisions on merit. Zonal has built its enviable reputation on providing the right product at the right price coupled with excellent technical back-up and support. If it can raise its profile and change its slightly stodgy image to match the bright new product, 999 will soon mean more than the UK emergency services number. ■

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AVI NuNeutron

For methodology see Studio Sound, April 1998, page 14.
See it on the Internet website:
www.prostudio.com/studiosound/apr198/r-tannoy.html

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

THE AVI NUNEUTRON is a small, 2-way passive loudspeaker comprising a 5-inch (125mm) doped-paper bass driver with a 25mm voice-coil and a 28mm fabric dome tweeter. The cabinet has a rear-facing port and has dimensions of 265mm by 140mm by 205mm deep and an internal volume of 5 litres. The crossover is described as being a computer-modelled, second-order Linkwitz Riley. The NuNeutron is specified as having a maximum SPL capability of 104dB at 1m from a 70W input (one loudspeaker driven).

The on-axis frequency response (Fig.1) lies within ±3dB between 90Hz and 20kHz except for a peak of about 4dB at 1kHz. The low frequency roll-off is seen to be well controlled, having a 2nd order (12dB per octave) slope, which is unusual for a ported design. The harmonic distortion is maintained below -40dB (1%) relative to the fundamental from 100Hz upwards, with the second harmonic rising to -14dB (20%) at 55Hz. The horizontal off-axis response (Fig.5) shows good directivity control over the important ±30 with the response maintained within ±3dB of the on-axis response, further off-axis however, the response is less good, with a broadening of the directivity evident between 4kHz and 5kHz. The vertical off-axis response (Fig.6) is dominated by the interference between the bass driver and tweeter between 2kHz and 5kHz which is inevitable in non-coaxial designs.

The step response (Fig.3) shows good time alignment and crossover design with a sharp rise and steady decay. Some 'ringing' at about 1kHz is evident up to about 5ms; this corresponds with the peak in on-axis frequency response shown in Fig.1. The low frequency transient behaviour is particularly well controlled. The waterfall plot (Fig.7) shows some evidence of ringing at 125, 500 and 1kHz, and demonstrates the excellent low-frequency control: very little low-frequency energy is present after 20ms. The acoustic centre result (Fig.2) shows that the low frequencies suffer a group delay that corresponds to a shift of about 1m relative to the high frequencies, further confirming the effect of the gentle low-frequency roll-off. The power cepstrum plot (Fig.4) shows some evidence of a reflection at about 180µs which is probably a baffle diffraction effect.

Overall, the loudspeaker performs well. The good on-axis response is spoiled somewhat by the peak at 1kHz which also gives rise to some ringing on transients. The distortion performance is good with the rise in distortion at low frequencies, perhaps inevitable in a loudspeaker of such small dimensions. A strong feature of this design is the low-frequency roll-off performance: although the response may not extend as far down as some larger loudspeakers, the smoothness of the roll-off should allow the NuNeutron to interface well with many rooms. ■

Contact
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Tel: +44 1453 752 656.

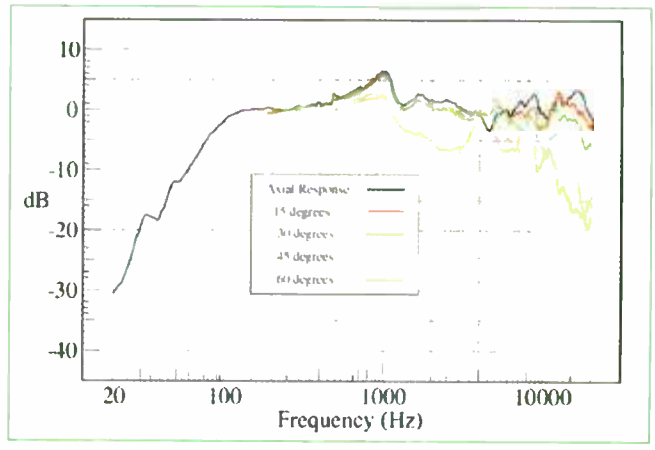

Fig.5: Horizontal directivity

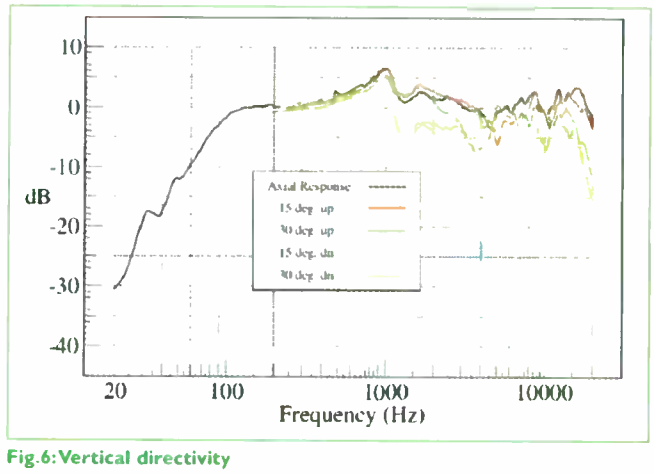

Fig.6: Vertical directivity

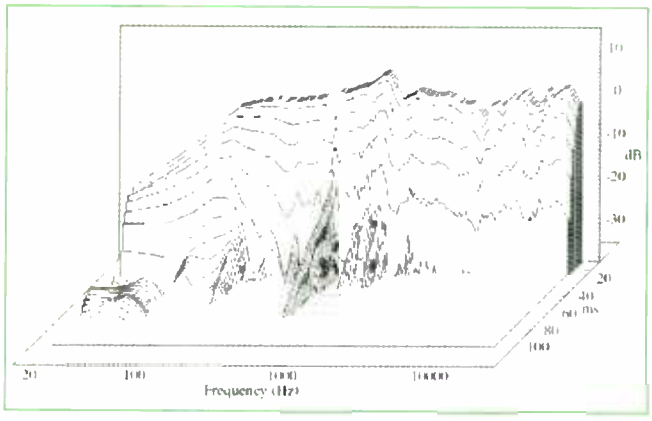

Fig.7: Waterfall chart

AFTER STUDYING the measurement results of the loudspeakers tested in this series to date, it has been decided that no information can be inferred from the overall power output plots that is not evident in the off-axis response plots. The power output plots have therefore been dropped from this and subsequent reviews. In response to requests (see letter in this issue), the power output plots have been replaced by waterfall plots which, it is hoped, will provide additional information.

There are numerous ways in which a waterfall plot can be calculated and presented. The following describes the result of much experimentation to find a method that yields the clearest and most concise picture of the loudspeaker performance. The waterfall plots are derived from the high-resolution on-axis frequency response function. The impulse response is first calculated as the inverse Fourier transform of the frequency response and a 50ms 'window' is shifted along this impulse response at 2ms intervals, the amplitude of the Fourier transform of each windowed segment is then plotted in cascade form for 'central' times from 0ms to 100ms. As with all such plots, the overlap between the time segments gives rise to redundancy in the displayed results, but this is considered desirable for ease of interpretation. The use of a 50ms time window also necessarily gives a restricted frequency resolution of 20Hz. To reduce time-domain smearing to acceptable levels, the time segments are subject to a form of 'flat-top' window function that results in 12ms pre-'tails' and post-'tails' on transients; this can be seen in the initial slope away from the main spike of the impulse response (the back-most frequency response plots).

Waterfall plots show to what degree certain frequencies or bands of frequencies continue ringing after a transient signal has been reproduced by a loudspeaker. Ringing can be seen as peaks in the response plots at later times when the other frequencies have decayed away. Looking at the plot for the AVI loudspeaker (Fig.7) as an example, the 'buttress' shape at 500Hz is evidence of ringing at that frequency. A 'perfect' loudspeaker would show ever smaller repeats of the back-most frequency response function decaying to nothing after 12ms.

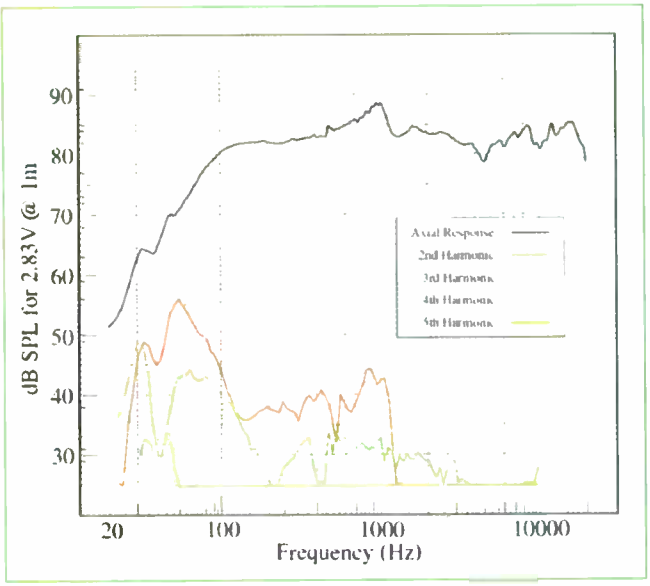

Fig. 1: On-axis response and distortion

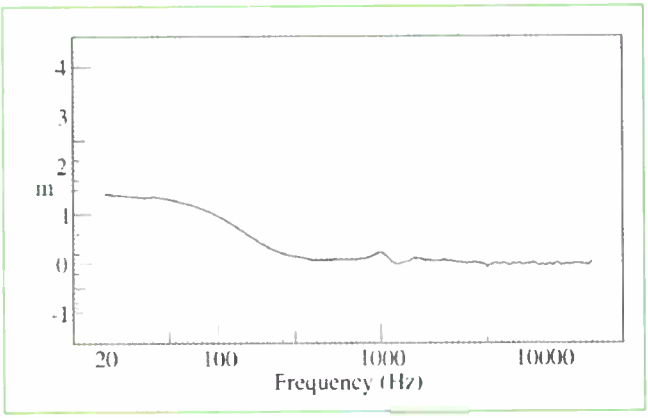

Fig 2: Acoustic centre

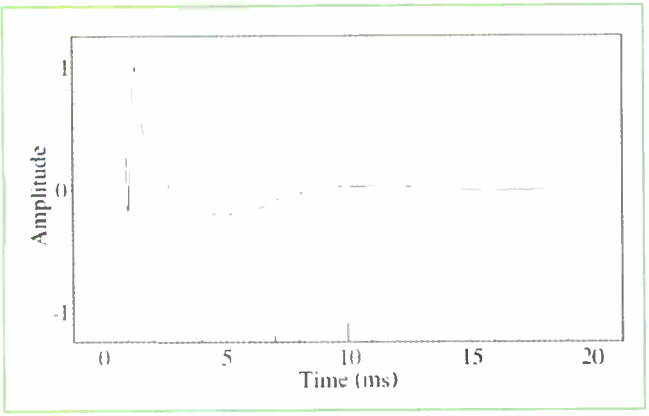

Fig.3: Step response

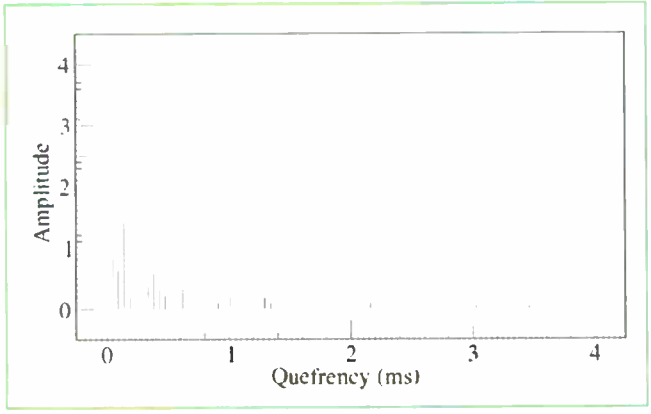

Fig.4: Power cepstrum

OUT OF THIS WORLD

L.A. Confidential	-	1997 BAFTA WINNER
Eve's Bayou	-	1997 INDIE SPIRIT AWARD
The Ghost and the Darkness	-	1996 ACADEMY AWARD WINNER
Batman Forever	-	1995 ACADEMY AWARD NOMINATED
Clear & Present Danger	-	1994 ACADEMY AWARD NOMINATED
The Fugitive	-	1993 ACADEMY AWARD NOMINATED
		1993 BAFTA WINNER
Under Siege	-	1992 ACADEMY AWARD NOMINATED

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Tascam's 24-bit DAT first

Tascam has introduced the world's first commercially available 24-bit DAT recorder. The DA-45HR is the first DAT recorder able to record true 24-bit audio data on a standard

DAT tape and uses a high-speed transport. Features include XLR balanced and RCA unbalanced analogue audio I-O, AES-EBU and SPDIF digital I-O, Word Sync I-O and a parallel control interface.
Tascam, UK. Tel: +44 1923 819630.

SPL new trio

SPL has launched three new products: a valve version of its stereo Vitalizer, the Qure equaliser, and the Transient Designer dynamic signal-effects processor. The new Stereo Vitalizer MK2-T combines valves with vintage-type coil filters for a retro sound. Qure is a dual-channel, 3-band parametric equaliser featuring a control designed to 'cure' digital harshness by introducing warmth, depth and dimension to mixes. The unit is said to selectively employ valves, coils and transistors at their optimum working frequencies. For easy recall of control settings, the unit is equipped with stepped potentiometers throughout. The Transient Designer introduces SPL's Differential Envelope Technology said to provide much greater control and flexibility in level-independent sound processing. Transients can be accelerated or slowed down and sustain prolonged or shortened, while all time constants are automatically adjusted.
SPL, Germany. Tel: +49 2163 9 8340.

V3.00 for ARES-C and CPP

V3.00 software for Nagra-Kudelski's ARES-C portable solid-state portable recorder and CPP rackmount ISDN codec variant adds features aimed at radio. Simplification of

on-screen menus can be assigned on a job specific basis. The upgrade also enables the use of bidirectional MPEG encoded uplink-downlink over ISDN with the option of mixing playback files with mic or line input. Compatibility has been improved with Studer >

AGM Digital TSS1;TSS1-D

With surround formats still in abeyance, mixing for a centre channel is becoming an imperative. **Rob James** searches for the missing link

A PERENNIAL CHALLENGE faced by film-dubbing mixers, and others concerned with multispeaker reproduction systems, is how to deal with stereo material. The stereo format attempts to create the illusion of sounds emerging from 'phantom' sound sources located between the 'real' left and right loudspeakers. Generally, this only works acceptably for listeners located in the central 'sweet spot' listening position. Off to either side these phantom images are pulled towards the nearest speaker, destroying the illusion.

In the cinema, and increasingly in 'home cinema' systems, three separate channels of sound are used to feed an array of left, centre and right loudspeakers intended to provide (among other things) a better experience for listeners outside the sweet spot.

When films are mixed using the analogue Dolby Surround matrix system, the encoding process can be relied on to bleed onto the centre speaker, not to mention the surround. However, the effect is not always predictable and often has a deleterious effect on the music balance. With digital multichannel surround systems there is no encoder matrix to help. Although Dolby do not condone the practice some mixers use the Dolby Surround encoder to produce a centre channel. Apart from this wheeze, the only option has been to use delays and filters to produce a hard centre.

AGM Digital's TSS process uses a Trifield-Gerzon, patented, frequency dependent, matrix-decoding method for feeding three loudspeakers. Michael Gerzon, a brilliant theoretician, did a considerable amount of research into how the ears and brain localise sounds. The theory is highly mathematical, based on orthogonal matrix theory, and a velocity-sound intensity theory of sound localisation. In Gerzon's words 'this leads to horrendously complex systems of nonlinear equations'. I do not pretend to understand the maths involved, but the bottom line is it appears to work. According to AGM, 'by careful optimisation, the stereo from the three speakers is balanced according to the various different methods used by the ears and brain to localise sounds at different frequencies.' Consequently, the German-based AGM operation has produced two 1U-high rackmount units using the theory to convert stereo source material for 3-speaker reproduction.

The TSS 1 and the TSS-1D Processors (equipped with a programmable centre-channel delay board) have balanced I-O on XLR connectors and a centre channel thru facility. The TSS1D also has an RS 232 interface and is supplied with a PC software package. The delay is there to compensate for the position of the centre speaker. Settings can be stored and downloaded to a unit. Once programmed the computer can be dispensed with as the unit remembers the previous settings with the power off. For convenience of installation there is a jumper on the processing board that makes the unit switchable via the RS232 interface using ASCII commands or the supplied software. Without the jumper, the unit goes into dumb mode, and the processing-delay is switched on or off by simply linking two of the RS232 pins. The unit then passes left, centre and right signals unprocessed, removing the requirement for external patching.

These AGM processors work, and they are ridiculously easy to use, what more is there to say about a box you simply plug in? Well, the programmability was added to offer ease of installation for live sound-reinforcement applications. The Windows software allows comparison between alternative settings using 8 memory locations, and an area where a sequence of two or more locations can be switched by simply toggling the space bar. In practice, subjective evaluation tends to produce slightly different optimum values for the centre-channel delay than mathematics would indicate. There is also a toggle between two

test modes. Left input routed to both left and right outputs, attenuated -3dB, compared to left input routed to the centre output with no attenuation. This allows the user to balance levels on live systems before allocating the centre-channel delay.

AGM claims to have adopted an analogue form over a DSP: 'There is no sonic penalty for having the unit purely analogue. It was decided to make it analogue and invest in making the boards small enough to enable some larger company to make a processor for domestic computer usage. It does rather a good job on a desktop PC. Apart from this, the unit boots faster than a digital equivalent and is inherently, with the right convertors, 96kHz, 24-bit ready. And yes, the delay-board option is digital.'

With multichannel sound-only delivery systems such as audio DVD looming large the TSS1 will find other uses converting stereo material when remastering. It also has a number of sound reinforcement applications. In fact units are already installed in some 20 German fixed installations, mostly handling reproduction in conjunction with a video screen.

Ultimately, if you need to convert stereo material for 3-channel reproduction this unit may be the simplest possible way of approaching the problem. ■

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Oram Octamix

Following the Octasonic, Oram's Octamix brings its modular 8-channel mixing system on line. **Tim Goodyer** figures the eights

THE SECOND ELEMENT of Oram's modular series is here. Begun last year with the Octasonic unit (eight mic preamps tucked into a 1U-high blue box), Octamix is the accompanying mixing stage (similarly tucked into a 1U-high blue box). The brief is clear: this is no modular studio mixer, as is, say, MTA's Intermix, but rather a collection of units defined by function that may be used outboard to a studio desk or in their own right, typically as part of a location recording setup. If there is a near parallel, it is Audio Toys' 8MX2, which combines the functions of both of Oram's Octamix and Octasonic, and a bit more, into half the rack space. It does not, however, offer the flexibility offered by Oram's modular approach in terms of its modularity.

With the scene set by the styling of Octasonic (*Studio Sound*, October, 1997), Octamix' front panel offers ten pots, 16 push buttons, a smattering of LEDs, headphone socket and power switch. Its rear panel offers eight balanced inputs on ¼-inch jacks (at +4dB, wired tip hot) and two pairs of balanced bus outs on XLR giving a +24dBu output with 18Hz-73kHz bandwidth. The jacks reflect the modularity of the system, allowing ready connection to Octasonic via the eight short patch-leads supplied. Alternatively, Oram's MWS offers a pair of channels of mic pre and EQ (and duplicated XLR-stereo jack outputs) with which Octamix can be fed. Although the intention is that the Octa-boxes can also be incorporated into Oram's Series-8 console.

Around the front, the first 8 pots are detented pan pots for the inputs and the remaining 2 are 41-position, stepped, output-level pots for the Master and Cue buses. Each input may be switched to either or both buses with the LEFT-RIGHT and CUE buttons which have green and red LEDs respectively to indicate their selection. A 5-LED ladder accompanies the output pots: 3 green indicating -5dB to +4dB, a yellow for +6dB and red for +8dB.

With Octamix patched in, routeing and panning could not be simpler, but both input trimming and level mixing fall to the level pots on Octasonic. And herein lies the first complication because while the preamps have an impressive +70 dB gain at one extreme, they bottom give unity gain at the other—or Inputs 6-8 do, as Inputs 7 and 8 are designated 'Hot Channels' and are calibrated for -12dB to +35dB working. Either way, you cannot attenuate an input signal down to inaudibility as you would need to for a fade. This is not the case with the output levels, however, which are legended 0-10, and go all the way up and down. In most locations settings, where levels would be preset and the mix would remain essentially static, this presents no problems and it is a trait shared by the ATI box.

The compromise being struck here is in the gain structure: Octamix dispenses with the unity gain amps used in mixing desks to provide separate trim on the channel amps and level control via the faders, shortening the signal path at the cost of having the output of the preamp sitting wide open on any enabled output bus. In the same way that Octa-EQ will offer 8 channels of MWS-style equalisation, a series of four Octafade units will provide 8 rotary faders in 1U, 8 60mm faders in 2U, 8 100mm faders in 3U, and motorised faders in 4U. There will also be a 4-joystick unit, currently being built as a special order by Oram's Custom Shop.

While it makes sense to assess Octamix primarily in terms of location recording, it is worth recognising that while the obvious use for the second stereo bus is for monitoring, Octamix does have four independent output buses and can be used for multitracking.

Of the three systems mentioned, it is relatively easy to establish a kind of hierarchy: working away from the recording studio, MTA's Intermix (*Studio Sound*, September, 1996) offers a system that can be configured to be useful in a recording studio, but may also find use in outboard-location settings. Oram's Octa-series offers a useful line in studio outboard and a flexible location system while ATI's 8MX2 (*Studio Sound*, March 1997) is a tidy, but less flexible, location box or outboard studio mic preamp. The big catch with the ATI box, however, is that it is no longer being manufactured.

I liked the Octasonic-Octamix combination (at £1,357 and £1,003 UK respectively). It takes just a little getting used to, but it is eminently functional, and passes audio cleanly and brightly. Its uses extend a little beyond the obvious if you have the time or inclination to explore them, but its strength is clearly in being part of a compact, capable and professional recording system for location recording work, whether it is direct-to-stereo classical or jazz, modestly multitracked rock 'n' roll or even location surround work—to this end, a reverb unit could be hooked in using the Cue bus and the Hot Channels on Octasonic. Oram has carefully defined the target market for this range, in terms of facilities, flexibility and audio quality, and its value is destined to increase with further additions to the range. ■

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Nagra-Kudelski, UK.
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XTA

XTA's flagship DP200 digital system controller for PA provides crossovers between drivers plus out-of-band and overload protection. The DP range also includes the DP226 speaker-management system with two inputs and six outputs, with an 8-band parametric equaliser for each input. Outputs all feature crossover filters, a 5-band parametric equaliser selectable to high and low shelving filters, a limiter and delay. Full metering is provided for inputs and outputs, with mute and access buttons allowing quick setup and gain adjustment. The DP226 can also be controlled via PC with XTA's AudioCore Windows control software. The GQ600 dual-channel graphic equaliser features 2 x 30 bands with long-throw sliders while the RT1 real-time spectrum analyser combines third-octave RTA, SPL meter, RT60 analysis and a swept frequency analyser.
XTA, UK. Tel: +44 1229 879977.

Flamingo pre

Cranesong's Flamingo is a 2-channel discrete Class A pic preamp with switch selectable 'vintage' and 'iron' sounds. Gain is

adjustable in 6dB increments up to 66dB with the help of a fine trim pot. A 22 LED VU meter monitors input.
Cranesong, US. Tel: +1 715 398 3627.

CCS Olympian codec range

Designed for stereo portable use, the Olympian audio codec offers compatibility with G.711, G.722, Layer 2 and layer 3, full duplex operation, ISDN and PSTN, many inputs and outputs and programmable functions. Features include three mic and one line in, stereo return and local mix outputs. Operation is via a menu, dials and a thumbwheel. CCS Europe has introducing its first audio codec with 2Mb/s interfaces, the CDQPRIMA 2MUX-M. The modular 2MUX-M is a bidirectional 'intelligent' audio codec that includes a 2Mb/s multiplexer and supports MPEG 1 and 2 Layer II and III, G.722, J.41 and J.57 audio coding standards. Network access via a 2Mb/s E1 interface is extended by an additional 2Mb/s drop-insert interface to allow for various network designs, while with the addition of an ISDN interface card as already used with the CDQPRIMA audio codec the 2MUX-M supports various ISDN protocols such as 1TR6, EDSS1 and NI1. The MPEG, J.41 and J.57 audio coding standards also include the capability to transport low data rate program-related signals with the audio, and this capability is supported in the CDQPRIMA 2MUX-M with various >

digital
three two e ght

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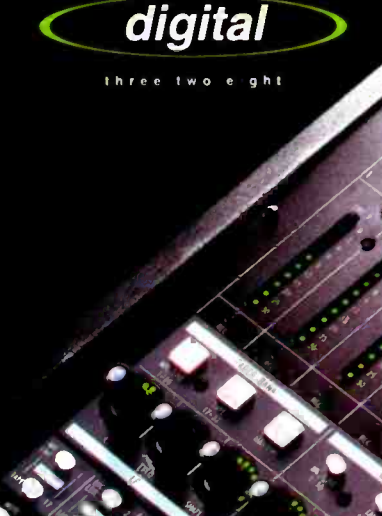

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SS

CEDAR DHX

CEDAR's budget Series X processors have helped remove noise from many people's grasp. Hush, whispers **Dave Foister**

CEDAR'S SERIES X has made a quiet splash. Facilities that for years had dreamed of having their own Cedar restoration processors suddenly had that option as the price of cleaning up plummeted. With little apparent compromise in terms of its capabilities, and few sacrifices in control functions, virtually the full armoury of Cedar weapons became available to a much wider audience.

Series X comprises three processors for the removal of clicks (DCX), crackles (CRX), and hiss (DHX). Their physical similarity shows the design philosophy that has made the range possible: they share a common board and chassis, with the different functions determined by firmware and requiring slightly differing controls poking through the same set of holes. The chassis and case are certainly not cheap items, lending a reassuringly robust feel and doing a remarkably thorough electrical job. Because of the nature of what goes on inside its equipment, Cedar's attention to EMI problems predates

the current regulations by some considerable time, and the integrity of these structures in this context would be hard to beat.

CEDAR's approach to the removal of hiss has seen, perhaps, more evolution over the years than any other of its processes. The software has developed in terms of both its efficacy, and the way it is operated by the user, becoming simpler to set up while getting better at its job. The full-blown NR3 system still retains the power of fingerprinting the noise and adjusting the resulting response, but the stand-alone units will operate without such preamble and deal with a remarkable range of noise characters in real time with minimal user intervention.

The DHX has only three adjustable controls, and, although they interact heavily, there is a simple and logical process for helping the unit identify and remove the noise. As with most CEDAR processes, this is a crucial point: the software does the bulk of the work, with the user helping to point the way rather than telling it exactly what to do. The most critical control is called LEVEL, and allows you to find the crossover point at which the unwanted noise can be eliminated. This is adjusted while the second control, called ATTENUATION, is at its most extreme, and results in some very odd sounds indeed. As the 'threshold' of the noise is reached a twittering is heard, caused by the presence of low-level wanted signals around the noise floor, and as the LEVEL control is advanced further more serious side-effects appear. CEDAR describes the main effect as glugging, and generally the HF is seriously compromised and the whole signal is modulated. This is not a problem, as at this point the unit is seriously overworking. The position to look for is the point at which twittering gives way to glugging (not a terribly technical description, but that's how the manual describes it), as this means the base level of the noise has been identified. The next stage is to reduce the attenuation to nothing, effectively putting the system out of circuit, and then to advance it until the required amount of reduction has been applied to the hiss.

The third control is the most subtle, and the hardest to describe, partly because Cedar is understandably cagey about giving away too much of the workings of the algorithms. It is marked VARIANCE, and tells the process how variable or inconsistent the perceived problem is—how noisy the noise is, as Cedar puts it. Not surprisingly, the software can do a better job of removing noise if it remains constant, allowing greater precision and more effective reduction without side-effects. If the noise is less constant the process must not be allowed to be so aggressive as it will inevitably affect wanted signals. High values of Variance therefore make the Level setting considerably less critical and help when a fast setup is required, but if the nature of the noise allows lower settings to be used then the final processing will be better optimised.

In the short time I had the DHX I was lucky(!) enough to be faced with a DAT of a live concert recording that was unusually noisy as it had been recorded about 30dB below full scale by mistake. As somebody wanted an extract put on CD something had to be done, and the DHX was the obvious solution. All the parameters came into play to remove the noise: if I asked too much of it the atmos between items could be heard faintly twittering at high playback levels, so, although the musical items sounded pristine, I had to exercise a little more caution. A brief twiddle of the VARIANCE and LEVEL interaction produced the required result: a clean signal that could have its gain cranked up as necessary without anybody knowing there had ever been anything wrong.

Armed with an upgrade to version 1.02 software—introduced to cope better with the high amplitude transients of dance and rock music—the DHX follows in the Cedar tradition to provide an uncanny means of doing the apparently impossible. The difference is that now you might be able to afford it. ■

CEDAR Audio,
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Cambridge CB1 4BN, UK.
Tel: +44 1223 414117.
Fax: +44 1223 414118.

NEW TECHNOLOGIES

< interfaces. The configuration of the system is controlled via RS232-RS485 interfaces and external PCs or controllers. The CDQPRIMA 2MUX-M supports point-to-point and multipoint connections for distribution, satellite backup, DAB, and other complex systems. The 4U unit can transmit up to five stereo or 10 mono programs complete with additional data signals.
CCS, Germany. Tel: +49 811 55160.

AKG modular pencil

Similar in presentation to AKG's C747 pencil mic, the Discreet Acoustics Mk2 is a modular system with five capsules and six installation mics—five goosenecks and one hanging module—that can be used in any combination. A phantom-powered LED ring confirms correct connection or can indicate mic activity status.
AKG, Austria.
Tel: +43 1 866 540.

Renkus Heinz total concept

Renkus Heinz Total System Concept (TSC), as the name suggests provides a complete package of complementary loudspeakers, controllers and supplementary equipment for touring and larger installation applications. The enclosures use proprietary CoEntrant Waveguide Technology, designed to provide a true broadband point source, time aligned, with a precisely controlled and predictable dispersion pattern. All the enclosures share identical compact dimensions, and boast excellent resolution, clarity and efficiency.
Renkus Heinz, US.
Tel: +1 714 250 0166.

A string of Pearls

Pearl Microphone Laboratory has three new models: the CC 22, TL 44 and DS 60 mics. The TL44 revamps the older TL4 model, but still maintains the original polar pattern switching flexibility of the original. The CC22 aims to offer a completely honest flat frequency response and very low noise. Meanwhile the DS 60 is a multipattern microphone with four cardioid outputs, 90° apart from each other. Pearl also has a microphone for fixed installation applications. The PA 58 has been developed to replace the old PA 54. The difference is a smaller microphone head at a 25cm gooseneck. Moreover, the frequency response is markedly flattened in the frequency band 3kHz-8kHz while the updated PA 58 has its electronics powered by 15V to 48V phantom power.
Pearl, Sweden. Tel: +46 42 58810. >

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Røde Broadcaster

Patiently building its range of affordable quality microphones has brought Røde into the broadcast studio. **Dave Foister** listens

RØDE'S REPUTATION, impressively quickly made, has so far depended entirely on one particular market area, albeit the broadest. All three models—NT-2, Classic, and NT-1—capitalise on the general enthusiasm for vintage microphones by shamelessly imitating them, and the result is three strictly studio microphones that manage to give the flavour of certain classics, with and without valves and frills, and always without the price tag. The impact they, along with the inexpensive eastern European imports, have had is shown by the response from the market leaders who have had to fight them on their own territory.

From the general to the particular: the new Rode is as different as it could be in terms of its breadth of appeal. Where its predecessors aim to join the ranks of the all-round classics, it sets its cap at a very specific market, namely the spoken voice, on air, but also in other situations. Called the Broadcaster (did they have to get permission from Fender?), its technical links to other Rode models are clear, but its features place it firmly in the broadcast studio or anywhere where speech recording is the primary activity.

Its outer appearance is not dissimilar to the NT-1 reviewed here in December 1997. A chunky body is capped with a coarse silver grille and mounted on a simple yet effective plastic stand attachment, locked on to it with a ring at its base. The big difference is that it is an end-fire design, using a large gold-sputtered diaphragm in non-negotiable cardioid mode. It should also be pointed out that in this domain where dynamics are often the first choice, the Broadcaster is a condenser, suggesting that perhaps it is indeed a variant on the NT-1.

But if this is the case than the variations are extensive. Several aspects of the Broadcaster's design are quite specifically aimed at its intended application, from the physical to the electronic. In the first category, there is an integral pop filter behind the grille, and in the second the bass filter is tailored for the particular proximity tip-up the expected close speaking voice will produce.

Perhaps the most obvious outward sign of the Broadcaster's role is a pod on the body containing a red LED. This is not there to indicate the presence of phantom power, and in fact has no link to the function of the microphone at all: it is to be connected to a switch on the console as an on-air indicator. Any switch will do, from a fader switch to a mute, and if suitably connected it will tell the 'talent' (how I hate that word) when they should stop coughing, swearing and farting. It does nothing at all to the microphone—there is no on-board muting or any other safety features—but placed as it is right between the speaker's eyes it should do the job of a separate cue light more effectively.

The unfortunate consequence of this otherwise thoughtful feature is that you cannot get the microphone out of the box and plug it straight in in the usual way. The connections for the indicator LED are combined with the audio output on a 5-pin XLR, necessitating the construction of a special break-out lead to hook it all up. Even the review sample was supplied with a 5-pin female plug and nothing else—I had to either improvise or make up my own cable. Fortunately the pin-outs for the audio are the same as the left-hand channel of a standard-wired stereo microphone, so I had a suitable Y-cord to hand. Of course for the application the Broadcaster is intended for this is not a problem - the microphone will be a permanent installation and it wouldn't matter if it needed a 7-pin DIN and an EDAC to connect it. The on-air LED is switched across the other two pins, open for on and short for off.

The high-pass filter switch is on the back of the LED's pod, and rolls off at 6dB per octave below 75Hz. It is awkward to get at when the microphone is hung in its clamp over an on-air mixer, which is, perhaps, intentional. Other than that there is nothing to adjust, not even a pad. This is, perhaps, surprising as the Broadcaster seems quite sensitive, producing high levels with close speech and requiring a pad on the connected preamp. The plus side is a clean punchy sound with all the presence Rode generally delivers. Pop filtering is very effective: it's easier to overload the connected electronics with a P than to produce a real blast, and the filter helps prevent even this. There is also internal shock mounting for the capsule, reducing the need for a big suspension mount that would hide the LED. This too does its job well, with as little mechanical stand sound as you could expect.

The Broadcaster is intended to be used where a big bright up-front sound is required, and it is certainly capable of delivering that with few penalties.

Other uses suggest themselves as well, but in its chosen sphere it should shine. ■

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Fax: +44 181 962 5050.
US: HHB Communications.
Tel: +1 310 319 1111.
Fax: +1 310 319 1311.

NEW TECHNOLOGIES

Wireless range

MIPRO's wireless mic systems are fitted with Japanese condenser capsules and PiloTone and NoiseLock squelch circuitry. The mics claim low handling noise and the ability to endure a 1m vertical drop without damage to the performance or casing.Products in the range include the MA101 and MA707 portable wireless PA systems, MR112 single-channel true diversity, MR312 16-channel VHF PLL synthesised diversity and 30-channel MR812.

MIPRO.Tel: +886 5 238 0809

AE200

Acoustic Energy, designer of the AE1 nearfield, has launched the low-cost 2-way AE200. This uses metal-cone driver technology for a claimed power handling of

125W, sensitivity of 89dB and a frequency response of 40Hz-25kHz.The HF employs a 25mm aluminium alloy diaphragm, ferrofluid cooled and damped, while the LF employs a 130mm chassis with anodised alloy cone thermally bonded to a 32mm high-power voice coil.

AE, UK.Tel: +44 1285 654 432.

DPA microphones

The latest additions to DPA's range of miniature mics, the 4065 is a lightweight headband unit that can be used live and in the studio. In theatrical application the problem of humidity is always present so DPA has paid particular attention to this matter, building in a double vent protection system and a drop stopper on the tube, intending the mic to be less prone to failure. Further durability is offered though use of a new kevlar reinforced cable that has enhanced flexlife and tensile strength. The capsule features a prepolarised omnidirectional cartridge element with a 5.4mm vertical diaphragm that is acoustically identical to the unit used in the DPA 4061. The headband itself is pre-bent to give a tight fixed position on any shape of head and can be easily reshaped. With a sensitivity of 6mV/P, matching the level of the human voice to the general input sensitivity of most wireless transmitters, the 4065 has a noise floor of 26dB(A) and can handle SPLs of up to 144dB before clipping.

DPA, Denmark.Tel: +45 48 142 828. >

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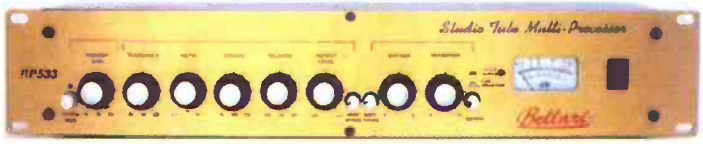
Bellari RP533

Pioneered by solid-state units, multiprocessing is now also the province of valve outboard. **George Shilling** warms to Bellari

THE RP533 is one of the cheapest microphone channels to include valve circuitry. It includes two Bellari-branded valves, but overall is as much a solid-state design as it is a valve one. It brings together processors from other units in the Bellari range in a cost-effective package. Despite the low price, this unit is, like the entire Bellari range, 'carefully hand assembled and individually tested in Salt Lake City, Utah'.

The light, yet solidly built, box is fronted with a brushed-gold metal panel. The rest of the box is sprayed black; although some internal surfaces had been missed, and astonishingly showed signs of rust. The switch gear is cheap in appearance, but works well enough. The knobs are pleasantly damped, more enjoyable to use than looks would suggest. The rear panel includes inputs and outputs on ¼-inch jacks and XLRs, the inputs accompanied by grungy little push buttons for 30dB PAD and PHASE REVERSE. This is plain daft: they should be on the front. Both jack and XLR outputs operate at the same +4dB ref level, which is unusual. Also on the back is the bonus of jacks for Side Chain In and Out, in order to access the compressor control circuitry. For so-called tube (valve) equipment, venting is surprisingly minimal, with just a few slits on the back panel. However, despite the high plate voltages, the RP533 was only slightly warm in use. The mains lead comes out from the back panel through a grommet, the case marked 120VAC despite this unit having been modified for the UK's 240V supply.



Front-panel layout is commendably clear, something not always achieved by designers of multiple-process devices such as this. The three distinct sections are marked: Preamp, Compressor-Limiter and Exciter. The Preamp section comprises a GAIN knob with a marked range of +20dB to +50dB (although the manual states +13dB to +33dB), accompanied by a PHANTOM POWER push button with LED. The compressor-limiter has knobs for THRESHOLD (-20dB to +20dB), RATIO (2:1 to ∞), ATTACK (0.5ms to 100ms), RELEASE (0.1s to 2s) and OUTPUT LEVEL (-∞ to +20dB with zero marked at 1 o'clock). COMPRESSOR ACTIVE and EXCITER ACTIVE push buttons are adjacent, (no LEDS, unfortunately), with Exciter controls comprising BOTTOM and DEFINITION, both marked 0 to 10. A switch to select either +4dB ref. Output or Gain Reduction accompanies the creamy-yellow dimly lit vu meter. A clicky rocker power switch is situated to the right.

The manual is short and somewhat vague in a few areas, but will suffice for most users, if indeed it is ever studied. The input gain seems to have been uprated since the manual was printed, and this was undoubtedly a very necessary improvement, as with +50dB gain maximum the unit can be used with lower-output mics. Even so, another 10dB of gain would not have gone amiss.

The rear-panel pad switch effectively doubles the range of gain achievable; although 30dB attenuation is excessive. 20dB would have sufficed. This enables use of high-level line inputs without overloading, if you can reach the switch, that is. All switches operate quietly, without any huge pops and clicks. I found the photocell-activated compressor slightly awkward to use, with a very hard-knee threshold. It sometimes seems very 'on or off', with the meter suddenly springing over to the left as THRESHOLD is adjusted. Also, with a minimum ratio setting of 2:1, very subtle compression is difficult to achieve. As the RP533 is marketed primarily as a valve processor, I found it surprising that the Attack setting could not be set slow enough to emulate the great valve compressors we know and love.

Indeed, the two valves would appear (from the block diagram) simply to serve as input and output buffers, playing no direct part in the Compressor or Exciter circuitry. However, I was pleasantly surprised by the Exciter section. I rarely use exciters on vocals, but this one has a very sweet high-frequency boost in the DEFINITION knob. This can be used subtly or more heavily, but even when pushed there is no obvious harshness or distortion. The BOTTOM knob added (or subtracted) a great warmth. My only reservation here was the massive boost achievable on frequencies below those normally desirable on vocals: there is no high-pass filter. Plugging in an instrument directly such as electric guitar was successful, and bass guitar particularly so, the compression and Bottom boost is well suited to the instrument.

Audio performance is very good, with a quoted frequency response up to 40kHz and a very low noise floor. It audibly performed as well as a similarly featured competitor at several times the price; although reservations about the Compressor's operation remain. Whatever, this is probably the best value unit in the Bellari range, slightly smoother in operation than the dedicated Compressor and Exciter models, and is certainly worth a listen if you are considering an optical-compressor based voice channel. It has its idiosyncrasies, and one should be wary of overspending because of the valve hype. There are some very good competing solid-state units (for this price or less) that include a high-pass filter and full EQ. But by all means give this one a twirl. ■

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Fax: +1 801 263 9068.

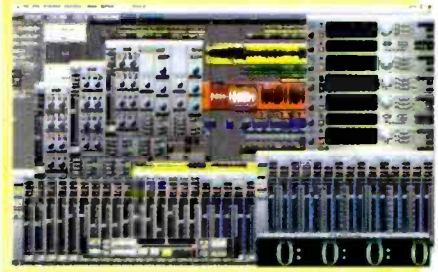
NEW TECHNOLOGIES

Soundscape adds plug-ins

Aphex Systems has announced an agreement to develop Aural Exciter Type III and Big Bottom Pro DSP plug-ins for the Soundscape SSHDR1-Plus DAW, and the new Soundscape Mixtreme 16-channel PCI card. The real-time plug-ins for the Soundscape V2 mixer run on Motorola 56301 DSP with Windows 95/98/NT front ends. The plug-ins are modelled on the top range Aphex Model 104 Aural Exciter Type III and Big Bottom patented circuitry. Also announced is the TC Dynamizer plug-in that delivers TC-quality mastering dynamic effects for the SSHDR1 and Mixtreme PCI card.
Soundscape, UK. Tel: +44 1222 450120.

Cubase goes Native

The latest version of Cubase Native Audio Processing and MIDI sequencing technology for Power Macintosh has been released. Developed in close connection with Apple, Cubase VST/24 4.0 offers 64 tracks of digital audio in 24-bit, 96kHz quality with each channel using up to four brands of EQ. Recordings can be shaped with up to four Plug-In inserts per channel, with eight aux sends addressing up to eight effects of Plug-Ins in the aux effects rack. The new group mixers offer four insert slots for effects and Plug-Ins. The mastering section offers busing to individual outputs of a range of supported digital-audio cards including the Korg 1212, Sonorus StudI/O, Lexicon Studio and Yamaha's DSP Factory. Internal resolution has been increased to 15360 ppqn,



and there are a host of new MIDI and audio editing functions including a controller editor for detailed editing of any recording parameter, a MIDI-Track mixer and Drag & Drop.
Steinberg, Germany. Tel: +49 40 2 10330.

TL Audio turns digital

Valve outboard specialist TL Audio has launched a digital division with the acquisition from Penny & Giles of the rights to design, develop, market and sell its Audio Multiprocessor System. Following the realignment of P&G earlier this year, the company felt the Audio Multiprocessor range would be better served by a company specialising exclusively in the development of audio products. TL Audio is already shipping the new digital units.
TL Audio, UK. Tel: +44 1462 490 600.

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Reader Response No. 038

Lexicon PCM81

Complementing the PCM91's dedication to reverbation, the PCM81 embraces a wealth of effects. **Dave Foister** meets a flexible friend

IF A JOB'S WORTH DOING, it is worth doing twice. Lexicon's PCM91, reviewed last month, constitutes half of a new pair of processors designed to build on the industry-standard status of previous models, and to bring effects from the top-end boxes to a wider audience. Where the PCM91 concentrates exclusively on reverberation, with a huge palette of the sort of stuff that made Lexicon's name, the PCM81 covers the whole gamut of effects processes while still centring on the famed Lexicon reverbs.

The two boxes are nearly identical in appearance, with only a slight variation in the pattern of the blue and black livery to distinguish them. The control layout is exactly the same: a new bright blue display is partnered by two rotary encoders for parameter selection and adjustment, with input level control and metering the other side. The actual roles of the encoders are determined by the operating mode of the unit as selected by a bank of 12 push buttons, most with integral green indicator LEDs.

The default mode is the one for choosing factory programs, of which there are hundreds. So much ground is covered by the presets that there's something for every occasion here, with only the odd tweak to the timing necessary to make it right for the job. It is immediately apparent that the quality of the reverbs, which are available on most of the configurations, is not in the least bit compromised compared with the PCM91: although the detailed programming potential is reduced in the interests of manageability. While there are some reverb-only programs, most work in conjunction with some other effect, if not two, so even with the complex editing matrix there's simply not the room for quite so many parameters.

Editing a program follows the same procedure as on the PCM91. Each program has a matrix of parameters, grouped as logically as possible into rows. UP and DOWN buttons, which you might suppose are for nudging values, are actually for stepping through the rows, and within each row the SELECT encoder chooses a column, or a specific parameter. This is then controlled by the ADJUST encoder. Navigation is easy; the only difficulties are (a) knowing what variables to expect for a particular setup, and (b) finding them, when a program may have as many as 50 variables. Of course it is all in the manual, which is spectacularly fat, and there is a limited amount of on-screen help available by holding buttons down. For those who do not want to get bogged down in the nuts and bolts there is a simpler editing mode, that allows access to a single row of the most important parameters.

These will vary enormously depending on what the chosen preset is doing. The range of algorithms is very comprehensive, including all the obvious delays, flanges and choruses, all the way down to the simplest of delay line setups for generating such effects the old-fashioned way. It also includes banks of tuned resonators and some of the smoothest pitch shifting you are likely to hear. Both of these can be controlled from a MIDI keyboard to set multiple pitch values in real time, just part of a very extensive MIDI implementation.

Editing factory programs is made easier by the facility to patch several parameters to one control, allowing more intuitive adjustment of an effect's colour than could be achieved by getting at all the relevant values separately. This is used especially well in some of the parameters which appear by default on the Adjust encoder: like the PCM91, the PCM81 always has one important setting live on this knob as soon as the program is loaded. This is even more appropriate on a multi-effects box such as this, and a good example is a preset to simulate the accidental stopping and restarting of an analogue tape machine. Here the knob toggles between stop and play, causing the pitch to drop quickly and ramp back up again as though the tape op has leaned on the machine. Other presets rely on a footswitch to do things like switch the speed of a rotary speaker simulation or freeze the input signal.

Both units have the option to tie the effects in with the tempo of the music, and this is even more relevant here with delays and other time effects than it is with reverbs. The working tempo can be entered numerically, tapped on a button or driven from MIDI, and parameters can then be set in note increments rather than milliseconds.

This is not new; neither is MIDI-controlled polyphonic harmonisation; nor are many of the features on the PCM81. The point is that capabilities like this, that others shout about or focus whole processors on, are taken for granted here, and are all present in the same box alongside some of the best reverbs in the business. There won't be much you can think of that the Lexicon cannot do, and it can do so much at once, and to such a high level of quality and controllability, that the resulting combinations may well be unique. I challenge you not to enjoy it. ■

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< to combine the performance of the company's PowerLight series with the value of the MX series. Four initial models are being shipped delivering 500W, 800W, 1000W and 1500W into 2Ω. Proportional response clip limiters optimise limiting based on programme material to preserve dynamics and user-selectable low-frequency filters increase useable power and protect speakers. Features include Speakon and touch-proof binding post outputs and LED indicators for major functions.
QSC, US. Tel:+1 714 754 6175.

Quested passive 3-way

Quested has launched the UD1 3-way passive monitor with dual proprietary 200mm bass driver, a 75mm mid-range and 28mm soft-dome tweeter. The speakers have their own integral stands with all drivers housed in individual chambers. The crossovers use polypropylene capacitors and air-cored inductors, and triple gold-plated binding

posts permit multi-amping. The QSB118 18-inch subbass cabinet is designed to compliement the HQ210 active system and is powered by an AP700 amp. The driver is double-cooled using vented pole piece and an external radial chassis.
Quested, UK. Tel: +44 181 566 2488.

It's the Flinstone

Amptec has launched a digital mixing console called the Flinstone 200D based on the earlier Stone D001 console. Using the same traditional user-interface, is configurable by means of different I-O modules, features sample-rate convertors on every digital input, and under the bonnet uses 24-bit A–D/D–A convertors, and 32-bit floating point DSP. The Flinstone >

Ronald Prent

on BASF tape

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BASF

Thermionic Culture Phoenix

It is increasingly rare now for valve outboard to appear with a true heritage. **George Shilling** on the rising of a retro compressor

VIC KEARY has been involved in pro audio since the 1950s, when studios often had to customise and build their own solutions. This environment has ultimately led him to the happy position he now occupies as research and design guru of Thermionic Culture. Over the years, he has set up a number of recording studios, most notably Chiswick Reach in the early 1990s, famous for its extensive valve equipment. Since then, he has had time to pursue his own design ideas for outboard equipment featuring valve circuitry. Vic Keary's Thermionic Culture partner Jonathan Bailes helps with design and oversees manufacturing.

The appearance of the Phoenix is striking. It is a hefty 3U-high rackmounter, with startling square vu meters. The upper casing is fashioned from bare steel sheeting punctured with a mesh of large holes, through which the major components are clearly visible. These comprise: six valves (of three different, relatively uncommon types); three huge transformers; four potentiometer shafts (for service adjustments) two massive capacitors; and the aforementioned meters; all mounted on a bare metal base section. It is a measure of the design philosophy that huge shafted potentiometers are used for trims rather than the tiny trimpots found more commonly. The smaller electronic components and wiring are enclosed within the base.

These are accessed by removing the bottom metal panel, this reveals familiar electronic components, but, astonishingly, no PCBs. Components are either attached to the casing or mounted on one of three solder-tag strips.

This method of construction is almost unheard of in these days of computer design. The rear of this section includes the IEC mains socket, fuses, and two pairs of XLR connectors for input and output. The rough casing construction could be from a school metalwork class, but has a rugged appeal.

The black front panel includes separate plastic-knobbed damped pots for each channel for GAIN (input level), ATTACK, RELEASE, THRESHOLD and OUTPUT TRIM knobs, and BYPASS toggle switches. There is also a LINK toggle for stereo operation, or as the manual usefully suggests, side-chain control of one channel by the input to the other. The controls are free from any markings of conventional calibration scales. All knobs are simply legended from 1 to 11—yes, the Spinal Tap joke has truly been done to death. To the right is a large POWER toggle switch, accompanied by a torch bulb with a green cover to indicate power on. There is no illumination of the odd-looking vu meters. These meters are of Indian origin, and were reportedly spotted in a nuclear reactor control room in a certain Eastern nation. Thermionic Culture reject a large number of these due to sticking needles, but after one initially paused a couple of times they were fine.

Their representation of the compression characteristics is not entirely accurate, as their movement is somewhat lethargic. This is no great problem, and one soon became accustomed to their nature. It should be noted that the Output Trim pots are placed after the electronics. They can be safely used to lower the output level for -10dB operation.

Loosely based on the Altec 436, this latest development of the design features a wider-ranging ATTACK control than the original prototypes. There is no Ratio control, as the design features a 'Vari-Mu' soft-knee circuit that ranges from <2:1 initially to >20:1 at 20dB compression via a pleasant curve. This makes setting up a doddle, the lack of conventional calibration forcing one to work by ear. Juggling the GAIN and THRESHOLD knobs enables a wide range of compression amounts, and the ATTACK and RELEASE controls are extremely wide-ranging, providing a variety of useful settings. It is remarkably difficult to make things sound bad with this unit. The Phoenix has a very musical character, and certainly brought one of my mixes to life, controlling the bottom end beautifully, making the vocals more rich and juicy sounding and generally making the track sound more dynamic. I took an instant liking to it, and forgave its lack of calibrated legending straightaway. On vocals the Phoenix is truly a joy to use, suiting all vocalists I tried it with during my brief review period.

Use with drums and bass were equally enjoyable. Noise levels were extremely low, and frequency response extremely wide. If I have any criticism, it is possibly that the sound of this unit is slightly too smooth and sweet, lacking a little raw 'graininess' of character found in certain older valve compressors.

The Phoenix is priced as a professional product, distancing itself somewhat from cheaper valve units flooding the lower reaches of the market. Taking into account the many years of practical experience behind this product, it seems like very good value. Indeed, the Phoenix compares favourably with competitors that are much more expensive. Construction is undertaken by hand by a team of just three people, which makes each one inherently more special than any mass-produced identikit box. The Phoenix is from a comfortingly old-fashioned Home Counties cottage industry: note the proud use of the word 'England' on the front panel. Its old-fashioned valve warmth brings a magical glow to any signal. ■

Contact

Serious Audio, 96 Queens Road, Watford, Herts WD1 2NX, UK.
Tel: +44 1923 442121.
Fax:+44 1923 442441.

< aims to be the perfect digital substitute for an analogue broadcast console, and without some of the more high-end features of the Stone is priced attractively. Maintaining the geological theme, the Boulder OB1 is an analogue production console designed for OB vans and TV or radio production studios. The Boulder features stereo and mono inputs with 4-band EQ, HP filter, direct and clean-feed outputs on every input module, 10 aux sends, 8 mono subgroups, 8 VCA groups and 8 mono matrix outputs, 2 stereo and 1 mono program output, 2 monitor and 1 communication module. Other special broadcast features include: compressor-gate on every I-O module, fader start, I-O transformers, and programmable channel on/tally output.The console is prepared for moving fader and snapshot automation, based on the Stone snap shot system, specially adapted for broadcast use.

Amptec, Belgium. Tel: +32 11 281 458.

Shure unveils UHF system

Shure has unveiled a new UC wireless UHF system that is designed to be a flexible and affordable contracting tool offering frequency agility, interchangeable microphone capsules and extensive LED metering. The main elements are a diversity receiver and choice of body-pack or hand-held transmitters.Occupying only half a rack space, the system's receiver is equipped

with adjustable squelch control, group and channel switches for frequency changes, a 2-band adjustable EQ section and Shure's proprietary MARCAD (Maximum Ratio Combining Audio Diversity) circuitry.

Shure, US. Tel: +1 847 866 2200.

AD's PICO mixer

UK broadcast specialist Audio Developments has introduced its new PICO mixer, the AD245. This keeps the facilities and usability found on its predecessor, the AD145, but features the low noise microphone amplifier originally designed for the AD140 series of mixers. A hybrid of this component has been produced for the new PICO, which offers powering for 48V and 12V Tonader microphones. While this new feature has been added, the main selling point of the original mixer, its simple but effective EQ section, has been retained.Other features include protection against radio frequency interference on the inputs, a balanced transformer, a high-pass filter to guard the transformer against low frequency saturation, limiters on the main output, switchable PPM or VU meters and a choice of frame sizes (six, eight or ten inputs).

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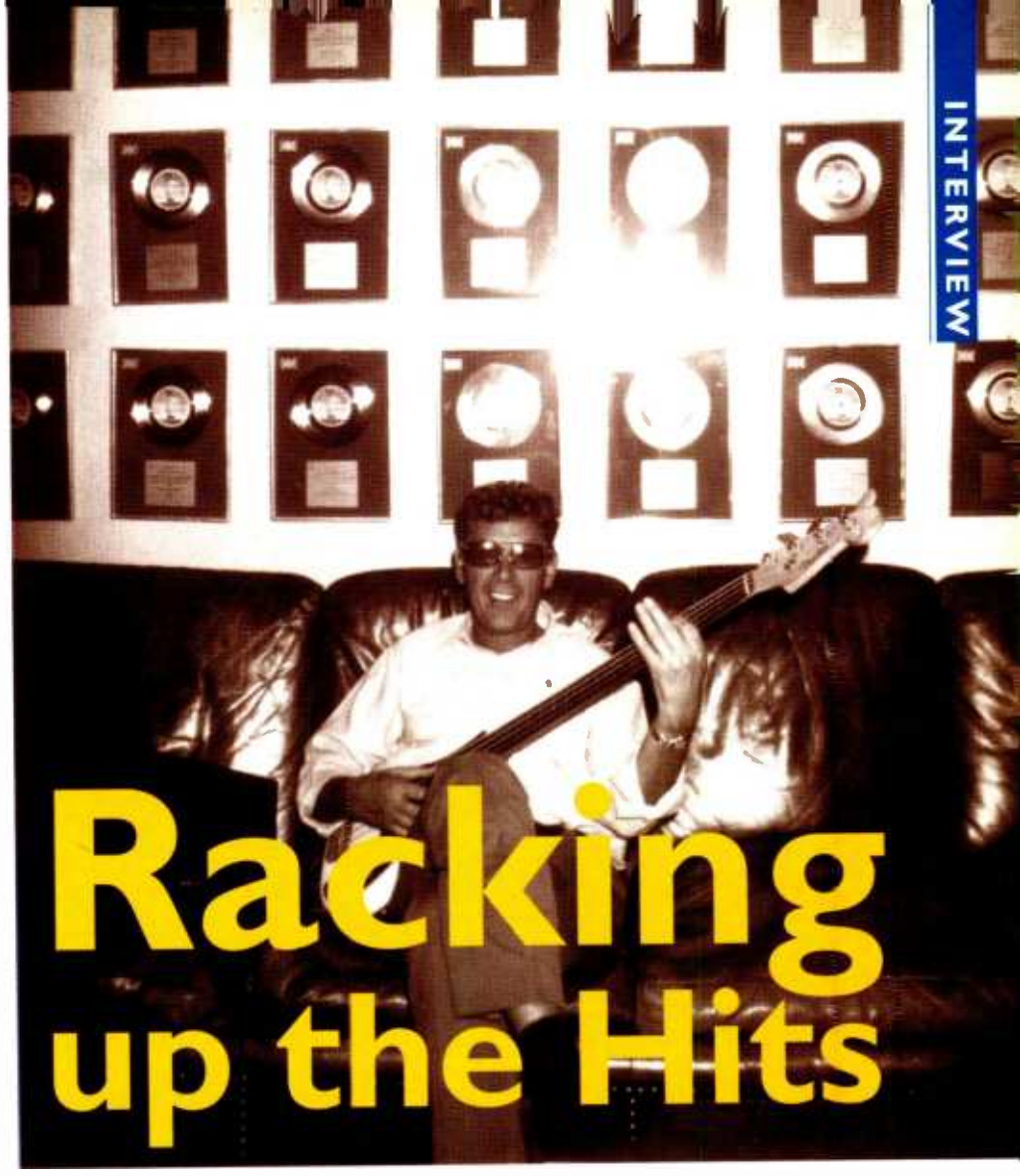

Racking up the Hits

A key player during the British music scene of the sixties, Mickie Most has recently returned to work **Richard Buskin**. makes the most of an interview

'THE FIRST THING you have to understand about the music business is that there are no rules,' proclaims Mickie Most. 'If I was to say to you a few years ago that a group from Manchester called Oasis was going to do The Beatles and sell millions, you'd say, "Oh come on, that's all been done". You never can tell what's coming next. It could be anything, but normally it's a progression from America with regard to what is dictated in terms of beats per minute. American black music is the only black music. British black music is really white.'

In the beginning, it was American black music that inspired Mickie Most. Artist, producer, publisher, record label exec, and studio owner, he has pretty much done it all during more than 40 years in the music business, and it was the blues and R&B that first drew him in before rock 'n' roll encouraged him to pick up a guitar.

A native of North-West London, Most grew up listening to the R&B sounds that were broadcast on radio by the American Forces Network during the early fifties. Then, once he'd learned to play the guitar, he performed semiprofessionally in clubs around Central London's Soho district, before a pairing with a schoolfriend saw the disingenuously-named Most Brothers recording for Decca during the mid-to-late fifties.

Mickie's penchant for making singles as opposed to albums probably dates back to this era, when albums were basically too expensive for him to buy. 'They were for people who had money to burn,' he asserts. 'In fact, one of the only albums that I bought in those days was the first by Elvis Presley, and I still play that today, so I've certainly had my 30-bob's worth out of it.

'Most of the albums would just have one or two songs that you liked and the rest was throwaway, so we were really hot for singles and we lived in this world of juke boxes. That was our entertainment. Any cafe that had a jukebox and a pinball machine was Las Vegas for us in the fifties, and I don't think I ever grew out of that.'

Meanwhile, success was still more than a step away for The Most Brothers in 1958 when the real Most decided to take a right turn and head for South Africa. In light of the rock boom that was then taking place in Britain this might have been viewed as a questionable decision, but it was made for purely personal reasons, and, as things turned out, happened to be quite fortuitous in the professional sense too.

'My girlfriend was from South Africa, and when she returned there her family said I would have to follow her if I wanted something more permanent,' Most now recalls. 'They thought I wouldn't bother but I did, and they then made it clear that I would have to spend four years there as they didn't know me, which seemed sensible. Well, during those four years I had a lot of hit records as a singer, and I also produced the records as there weren't any producers in that very small market. That's how I learned to produce.'

Indeed, having installed himself at the vanguard of the burgeoning rock 'n' roll scene in South Africa together with his band The Playboys, Mickie Most was a teen sensation there during the late fifties and early sixties. Nevertheless, once his 4-year probation was up he headed straight back to England, and it was in 1963, while in Newcastle during a package tour alongside The Everly Brothers, Little Richard and Bo Diddley, that Most visited The Club-A-Gogo and saw a band on stage called The Animals.

'I immediately knew that this was what I'd been looking for since I'd arrived back in England,' he enthuses. 'At that point the music scene was really bland, with people like Eden Kane and John Leyton, but The Beatles, and their like, were also just starting to hit, and so I was fortunate to be in the right place at the right time.'

Not just fortunate, but also and astute. At a time when there was only a handful of major record companies controlling matters, Most, himself, opted to pay for the artists that he signed and subsequently produced.

'The record companies didn't like the idea of you doing things that were outside the norm,' he says, 'but I just signed the groups to myself and I financed them, offering them a royalty and a deal, and then it was up to me to make this deal work. Fortunately I had already been recording with EMI, and EMI were interested in what I was doing. They had a label manager there working for Motown named Derek Everett and he liked what I was doing. The first record that came out was a hit, the second record was No.1 all over the world, and after that I never had a problem.'

The first Top 20 hit by The Animals was a record entitled 'Baby Let Me Take You Home', on the strength of which they secured a tour supporting Chuck Berry. Now, one of the numbers that they performed regularly on stage was 'House Of The Rising Sun', so in mid- >

< tour Most decided to bring the band down from Liverpool to London overnight and record their extraordinary performance of this song.

'They got on the sleeper and I picked them up early in the morning along with their drum kit, amplifiers and all their gear,' he recalls. 'We were booked into Kingsway Recording Studio for a 3-hour session from 8 until 11, and by 8.15, take two, I said, "That's the one".'

So, what to do during the remaining two hours and 45 minutes? Well, that was easy. Why not make an entire album?

'That consisted of songs that they wanted to record, really,' Most replies. 'Songs that they'd rehearsed and played many times as part of their repertoire, so I said, "Okay, go for it". We did everything live, straight to mono, and that's how it all started. After that, for me, it was a case of hit after hit. The next one was 'Tobacco Road' with The Nashville Teens, followed by 'I'm Into Something Good' with Herman's Hermits...'

The last of which was a million miles away from the earthy R&B sensibilities of The Animals. Mickie Most, don't forget, was into R&B and rock 'n' roll, but, as time would soon tell, he was also equally into capitalising on his uncanny knack of spotting commercially viable acts and matching them up with exactly the right song material in order to produce hit records.

'I had this Goffin & King tune, 'I'm Into Something Good', which was really catchy, and I really thought it needed somebody youthful-looking,' he explains. 'Herman's Hermits' management had called me many times and asked me to take a look at them, so I said, "Send me a photograph", and as soon as I saw the photo I envisaged Peter No-one as a young [President] Kennedy. I quickly went up to Bolton, where the band was playing, and they were doing all of the pop R&B stuff such as 'Mother In Law', 'Poison Ivy' and so on. I'd brought 'I'm Into Something Good' with me and they fitted it really well, so I told them to learn the song

by Sunday and we would record it then. That's basically what happened and it was as simple as that.

'I used to spend every other week in New York or Los Angeles, scouring around places such as The Brill Building for material. I had all of these appointments set up for me, so when I arrived there on a Sunday night I'd have my schedule and then from Monday morning to Friday evening I'd visit publishers and listen to tunes. On Friday night I'd return to London, record the material the following week and then go back to the States the next Sunday. That's all I did for five years.'

It sounds straightforward. However, while a lot of people think that they can spot a hit record, most potential hits never even make it into the lower regions of the charts. So, given his impeccable instincts, what is Most's criteria when trying to identify the kind of material that prompts record-store cash registers to ring? Well, to quote the man himself, 'There has to be that bit of magic. The song, the recording; the whole thing has to add up in my mind to a hundred, and when I hear it I go, "That's it!".'

Does that explain the hit-maker's method?

'At other times I might hear a great song, but the arrangement isn't doing it for me, so I'll rework it and often that'll turn out to be what is needed. I just seem to have the ability to do that, I don't know why. There again, there are also times in your life when you're wrong—perhaps it was the wrong timing for the record; it came out too early or too late, we didn't pick up the airplay, whatever—but if it doesn't succeed, it doesn't succeed, and making excuses is a negative. You just have to say, "Okay, I goofed, I've got to try harder next time".'

As the sixties moved on, Most found himself working with a range of different artists, from Lulu and Donovan to progressive bands like The Yardbirds and heavyweight talents such as Jeff Beck. The whole record business was rapidly evolving, rock music was being taken more seriously and priority was increasingly being given over to album sessions, so in MM's case weren't the sessions starting to take quite a lot longer?

'They certainly all took longer than >

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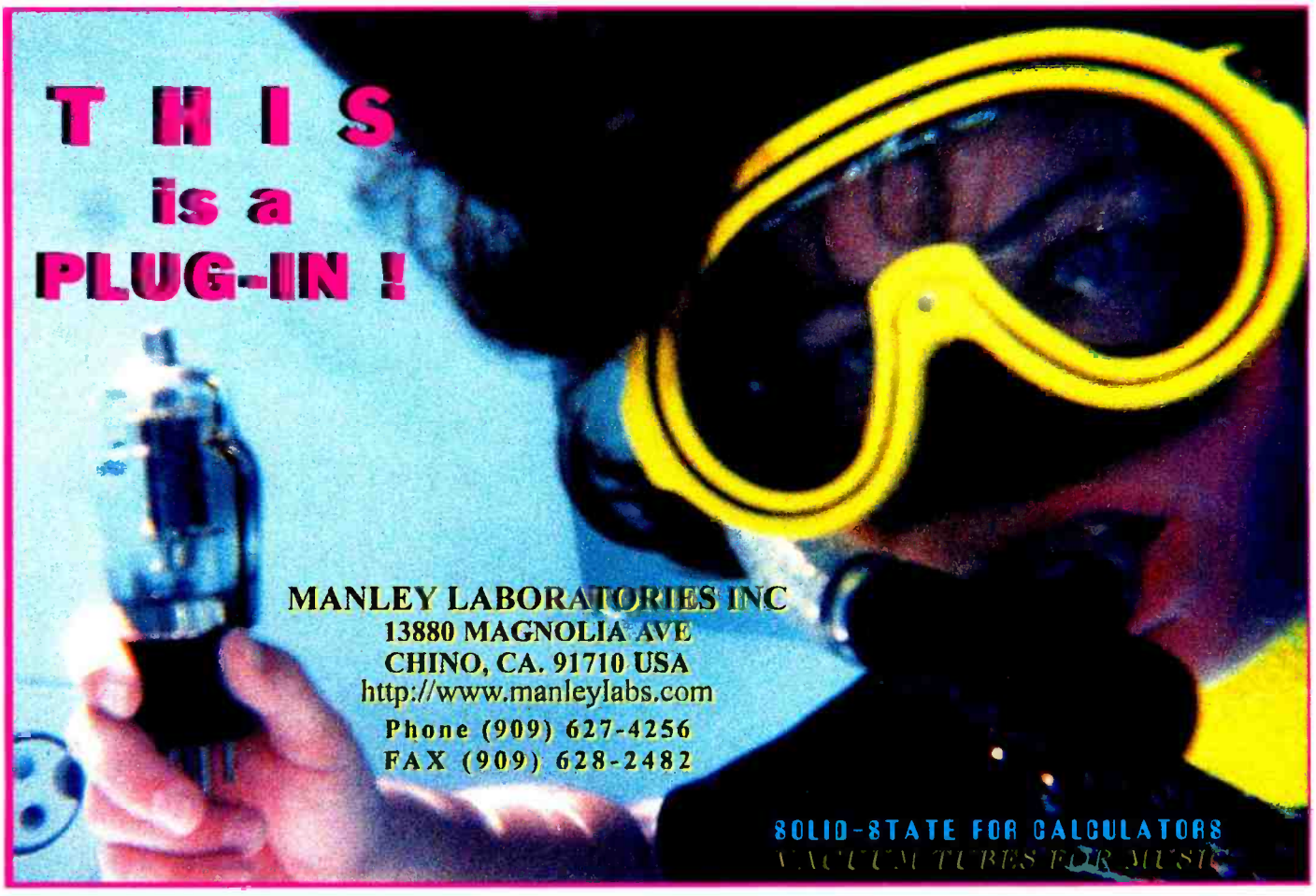

< 15 minutes,' comes the laughing reply. 'I mean, that was just one of those freak things, but still, as far as I'm concerned, once a performance is on tape it's just pointless to keep going on. With ['House of the Rising Sun'] I realised I'd got it, and I must have got it because the record's sold millions and millions for more than 30 years. I'm sure if I'd spent another two weeks doing it I would never have improved on it. In fact, it would probably have got worse.

'As we moved towards the late sixties things were changing technically. We went from mono to stereo, 4-track to 8-track to 16-track, and things obviously began to take longer, but I personally like to be in and out of the studio. I just can't keep things up for that amount of time; it's too long. Also, as you get older you don't want to waste so much time doing something that you used to do in three hours. We used to do a whole single and maybe a spare B-side in a 3-hour session—certainly we'd get the master done—and even all of those Donovan records, some of which were quite complicated, were done in three hours.

'Having said that, I think there are records that suit the lengthier sessions. A lot of the dance and rap records suit it, because you've got programmed material and once you've got the vibe right on the program it remains right. So, you can do overdubs, and so on, and maintain the energy, but I do believe that when you're performing group music the recordings sound and feel better when everything is played together. If they're good players and they're well rehearsed they should play in time, and that will produce a better feeling. After all, most people buy records because they feel and sound right. Musically they don't understand, and why should they? They're not musicians, but instinctively they know.'

Even though Most had brought Epic Records a lot of success over the course of about five years courtesy of his work with the aforementioned Donovan, Lulu and Yardbirds releases, by the end of the 1960s CBS' main man, Clive Davis, felt that the single had already had its day. So did many of his industry colleagues, yet Most did not agree, maintaining that it should still be regarded as the flag-waver for the album. This was the attitude that he adopted when forming his own RAK Records label in 1969; if others were willing to abandon the singles market then he would aim to fill their shoes, and the result was that the first 27 records issued on RAK were all Top 50 hits. Thus commenced Mickie Most's cycle of success during the 1970's, during which time he discovered acts such as Suzi Quatro, Mud, Smokie and Hot Chocolate.

'I was recording an album—that never got finished—with Jeff Beck at Motown in Detroit when I first saw Suzi Quatro,' he recalls. 'The manager of a group called Cradle invited us to see them in our spare time, and they were pretty good, but it was the bass player who caught my attention. She was not singing at the time, she was just standing at the back, but she played very well and I thought that she had something. So I told the manager that I was not interested, but that if the group didn't make it, and it should break up, then give her my number in London, and I'd like to talk to her.'

Well, as luck would have it...

'Sometime later she phoned and said that the group had broken up, so I sent her some money, a contract and a plane ticket, and that's how it happened.'

Hot Chocolate, meanwhile, appeared at the RAK offices in the form of songwriters Errol Brown and Tony Wilson during the early seventies, and Most promptly used a couple of their compositions to good effect with Herman's Hermits and Mary Hopkin. However, when Brown and Wilson then turned up with a song entitled 'Love is Life' he suggested that they record it themselves. Some session musicians were quickly brought together in the studio, but it took a lot longer to come up with the sound that the producer was looking for.

'We had to create a sound, a Hot Chocolate sound, because there wasn't a group, just two writers,' he says. 'I kept

trying things, and it was almost like being a chef really, introducing different ingredients, throwing them away and starting again. Eventually I got this kind of organ-guitar thing going, this riff, and that's what they got known for.

'Anyway, that record was a very big hit. The second and third weren't so big - they were a bit too Carribean, too calypso-ish, and I told them that although the songs were all very pleasant I didn't think they were going in the right direction. I said, "You've got to write something really black, otherwise there's not much more I can do", and the next song that they wrote was 'Brother Louie', which was a black record. That got them back on the path, and then we had all of the big stuff that followed, like 'Sexy Thing' and 'Everyone's A Winner'.'

Meanwhile, it was a riff that Mickie Most had heard on an American blues record back in the mid-fifties that would eventually lead to him converting a drummer into a front man. Cozy Powell was the sticks-man and 'Dance With The Devil' was the record that he had a smash hit with during the mid-seventies, by which time the riff had been floating around in Most's head for the best part of 20 years.

'It reminded me of the Coronation Street theme,' he says. 'It had the same notes. I'd known Cozy since he'd worked on the Jeff Beck album in Detroit, and we got on really well, so I suggested turning this riff into a drum-based song. We worked on the arrangement, we recorded it, and, amazingly enough, it became successful as well.'

In the autumn of 1976, with a stack of hit productions to his name, a successful record label up and running, and the profits to show for all this, Most decided to circumvent the then-crippling British tax system by investing in a recording facility, RAK Studios, in North-West London. Not perhaps the easiest way to make money, but nonetheless an appropriate move for someone in the business, and, given our man's entrepreneurial instincts, a sure-fire earner all the way.

'It's made money every year for 22 years now,' he says. 'So, on top of the tax avoidance, it's been a very, very good decision. We bought the building for about £350,000 when the market was on the floor—and we're talking about 60,000ft² in St. John's Wood, a hundred metres from Regent's Park—so it was really a bargain. Then, when we sold the record company, which owned the studios [to EMI], I bought the studios back for a couple of million, and now I think they're worth about £7m.

'When I sold the record company it was because music was turning in a way that I didn't understand. You know, the kind of stuff that Duran Duran and the like were doing, it was starting to move into areas that I didn't feel I could >

< contribute to. You see, before sampling and synthesised sounds, when we were in the studio we relied heavily on the rhythm section, and then if we wanted to sweeten the tune up a bit there were only three or four things we could do: we could either use strings, brass, reeds or voices. There was nothing else, other than the percussion, and so if we wanted to make a sound that didn't exist in those days we'd have to do so through echoes and mixing sounds together. We did that a lot with Donovan and with The Yardbirds: putting amplifiers in cupboards and microphones in the toilet, and tape running all around the studio for those long delays as we didn't have Lexicons. That kind of thing was interesting.

'Eventually, however, the recording process got bogged down with so many preset sounds that it became very difficult to make a decision. At the same time dance was getting in there as well with all of those 125 beats, and while the people who were taking Ecstasy could probably see the light, I couldn't. After all, I was getting past my sell-by date now to be spending my evenings dropping this stuff and leaping around until five or six o'clock in the morning. I'd already spent 30 years of my life in the recording studios, and so just before I was 50 I retired from seriously making records.

'I took a long time off and I just sort of dabbled, dealing with our publishing company and overseeing the updating and installation of new equipment in our four studios. I would hang out with a load of musicians who came in everyday for three months, made their album and disappeared before another lot would come in, and it was great to do that because I didn't feel like I was missing anything. Instead I was here without having to do any of the work!'

Recently, after an 8-year hiatus, Most produced a new artist named Tee as well as an album and two singles by a group of three girl singers called Jamaica. So it is that his work has come full circle in a sense, trying to develop raw talent and once again come up with a sound.

'It's interesting,' he says, 'because it's a battle. It's a big challenge, and I've always loved a challenge.'

Today, as the producer of records that have amassed more than 200 million sales, Mickie Most lives up to his *Sunday Times* billing as one of the Top 200 most successful people in Britain, collecting cars, riding his motorbikes and residing in a palatial house in Totteridge, North London, that boasts 8 bedrooms, 9 bathrooms, a 40m indoor-outdoor pool, fully fitted gym, sauna, tennis court, five-a-side football pitch and 4½ acres of parkland. Not everyone in the music business finds themselves in this position, but then not everyone has Most's acumen.

'We had a deal with MGM,' he says, looking back to his earliest distribution deal. 'They had Herman's Hermits and The Animals. Then I had a further deal with Epic for the next five artists that I produced. You see, EMI distributed MGM's records in England, and the president of the company came over and said, "Hey man, why haven't we got The Beatles?". So, Len Wood, who was running EMI at the time, looked down the charts and said, "Well, have these ones: The Animals". It's funny how these things start.

'I'm Into Something Good', 'Tobacco Road' and 'House of the Rising Sun' were all number one hits in America, yet six months before I'd played these records to a lot of companies in New York, Philadelphia and Los Angeles, and not one of them would take any of the three, saying they didn't think they were right for their market. It wasn't difficult to hear that these records were in for a shot, so I was laughing really. It didn't depress me at all, but the only thing was that I expected these people in America to know. I couldn't believe how much they didn't know, and nothing has changed. They're clueless, and the proof of this is that if they knew what was going down they would all be multi-millionaires, wouldn't they? I mean, the guys in A&R departments would be riding around in Lear jets if they got it right all the time, but they're not.' ■

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Photo: Dominic Chalileau

The power of Three

Continuing its operatic association with football, the 1998 World Cup closed with a performance from the Big Three: Pavarotti, Carreras and Domingo. **Kevin Hilton** witnesses France's other victory

FOOTBALL is a schizophrenic phenomenon. Pele, one of its greatest practitioners, dubbed it 'the beautiful game', but it has also been responsible for some of the ugliest scenes ever witnessed, both on and off the field. It is heavily associated with the mob mentality, which has produced the football chant, one of the more moronic human achievements.

Bizarrely, when organisers wish to celebrate the climax of a major soccer tournament, they call on three of the finest voices ever heard to perform some of the most intricate music ever written. Schizophrenia isn't in it.

Carreras, Domingo and Pavarotti are keen football fans—supporting Barcelona, Real Madrid and Juventus respectively—and first performed together in 1990 at the Baths of Caracalla, Rome as part of the Italia 90 World Cup tournament. The operatic equivalent of a super group aroused huge interest, involving live television and radio broadcasts, a live video, and a CD. When the trio performed in Los Angeles as part of the USA 94 pageant, the TV coverage attracted a worldwide audience of 1.3bn, making it the most watched global music event ever.

Until Friday 10th July 1998...

With the Eiffel Tower as an enviable stage prop, it was billed as the concert of the century, with a projected audience of 2bn people within a week of the event. On an elaborate, specially designed, stage in the Champ de Mars, the big men of opera performed a new programme of arias and songs, specially arranged by composer Lalo Schifrin, who is best known for his film and TV work, and played by l'Orchestre de Paris, conducted by James Levine, artistic director of the Metropolitan Opera.

The 2½-hour show was beamed by satellite to 99 countries, 75 of which were relaying it live; other broadcasters were either constrained by the time difference, and so had to record it for later transmission, or decided on later broadcast (the choice of the UK's BBC, which edited the concert for a 2-hour slot prior to its live coverage of the World Cup Final on the Sunday). The technical infrastructure necessary for this was additionally used to record proceedings for subsequent video and >

Photo: Kevin Hilton

Photo: Richard Haughton

< CD releases, with a possible DVD in the offing.

While the live performance is the core of this event, it is the global broadcasting and recording rights that pay for it. In preparing for the performance, the technical teams had to take the audience in the Champ de Mars into consideration, but never forget that the audience in their homes and the requirements of the TV stations were the prime concern.

Since the 1994 Los Angeles concert, the *Three Tenors* shows have been produced by Hungarian-born Tibor Rudas and his Rudas Organisation. The Paris show, the first time the trio has performed in France, is part of their ongoing 1997-98 tour and cements the relationship the mercurial producer formed with Pavarotti in 1982. The Rudas Organisation negotiated contracts with the broadcasters, hiring in specialists to co-ordinate the vision and audio production for both the world-wide broadcast and the subsequent sell-through versions.

In direct control of audio was sound engineering director John Pellowe, a former Decca engineer who first worked with Pavarotti in 1975 as a sound assistant. When the tenor began to play arenas, Pellowe was brought in to share front-of-house mixing duties with Jimmy Locke in the mid-eighties, taking over completely approximately three years ago. Since then he has handled all live sound and recording matters for Pavarotti, leaving Decca at the end of 1996 to go independent.

In his role as sound engineering director, Pellowe had to arrange personnel and facilities for front-of-house, recording and broadcast. For the latter two, Pellowe worked in conjunction with Toby Alington, as broadcast audio producer, and his Audio Logistics production company. Pellowe's initial plans to handle recording and broadcast mixes in one mobile studio were discarded in favour of three trucks: locally based Le Mobile Son for the main broadcast mix, performed by Pellowe; Euphonia as the hub, engineered by Alington, taking on functions that would usually be handled by the sound desk in the vision scanner; and Manor Mobile 1 to make the main CD-video soundtrack recording, mixed by Alex Marcou.

Splits from the stage went to both Le Mobile Son and the Manor (each with an SSL console and Soundtracs

All other photos: Kevin Hilton

Stage and API submixers respectively), with the former sending the main Dolby Surround encoded mix and the latter a safety stereo mix to Euphonia. This Raindirk-equipped truck, best known for its classical work, additionally took the 14 audience mics and a safety mix from the front-of-house system, also sent to the main vision scanner, which took the final surround mix from Euphonia and sent the transmission feeds to the international distribution vehicles.

This configuration ensured that, even in the event of hideous disaster, something was going to go to transmission and to tape. Toby Alington explains that the various audio trucks and their personnel were hired in by Audio Logistics to carry out specific functions.

'The hub audio functions can be carried out by the sound desk in a scanner,' he explains, 'but these don't have the quality of monitoring or the type of console that would usually be used for music. I mixed the final transmission output and Alex [Marcou] concentrated on getting everything on multitrack to be remixed later, effectively leaving John [Pellowe] to get on with the live broadcast mix.'

A live outside broadcast of this size and magnitude, which was the first time the tenors had performed together outside of a purpose-built amphitheatre or arena, will always bring problems, particularly in terms of extraneous noise. Alington illustrates the specific issues of this event by saying, 'There was bound to be something as we had an ambient stage, a crowd of 800,000, more satellite uplinks than had ever been used on such an event, microwave connections, multilaterals, unilaterals, lots of RF flying around and, right on top of us, the Eiffel Tower, which is a backup transmitter for Radio France.'

Buzzes, tizzes and earth problems preoccupied the technical crews in the days leading up to the performance, even taking up the Thursday, which had been designated a rest day. Eventually the problem was minimised, with efforts made to ensure that the recording feed going to Manor 1, seen as the highest priority, was totally clean; while there was still some noise on the broadcast output, Alington says, 'It was very acceptable and within broadcast tolerances.'

The CD-video signals were recorded onto 72-track digital (two 3348s and two 3324s), with backups made in Le Mobile Son on two 3348s and in Euphonia on 16-track digital (4 DA-88s), time-code DAT and non-time-code DAT. While there was some confusion in the run-up to the concert as to whether it would be offered in Dolby Surround—a situation caused by the Rudas Organisation not returning the relevant paperwork until the day before—it had always been the intention to use the format, which was one of the reasons for choosing Le Mobile Son, as it is surround ready.

On creating a surround sound mix of a full orchestral performance, Alington says, 'With music, the ten- >

dency is to keep the whole stereo image across the front. It doesn't translate well into stereo or mono if you take the instruments into the mid-field—the orchestra belongs along the front. What we put into the rear loudspeakers was the audience, ambience and very carefully set up reverbs to simulate the live experience. You have to be subtle but it can add a huge amount, making the listener feel as though they are there at the concert.'

Alington's Audio Logistics orchestral credits include Carreras in Red Square, the LPO, Warsaw Symphony Orchestra and the Bolshoi Ballet, while among other clients are Bjork, David Bowie, NHK, HBO and VH1 Europe.

Front left-centre-right monitoring was on the recently released B&W 805s, which were installed into Le Mobile Son and Euphonia, replacing the incumbent Genelecs in the case of the former.

'They proved ideal,' comments Alington, 'because they are small enough to put into a truck but they sound like big monitors.' (See sidebar)

The audio was postproduced at the Penthouse in Abbey Road Studios by John Peffed notes or coughs and splutters. Any audio edits required were carried out at Classic Sounds in Kilburn; this work was mixed onto DAT and then laid back for mastering to picture at Tele-Cine. All this had to be carried out in the week following the performance, as Pellowe was due to mix a Carreras performance in Germany on 21st July, while the CD had to be ready for release on 17th August.

The Tenors were in good voice, the setting was impressive and the fireworks spectacular but there is the nagging thought that this phenomenon cannot continue for much longer—at least not with the same participants. At least five more such concerts are planned, so look out for the Three Tenors Millennium Party, coming soon to a TV near you. ■

Surround is increasingly being used to enhance all types of music and performance. In the case of a high profile event like the Three Tenors in Paris, the idea is to give the listener-viewer at home a sense of being there.

Toby Alington of Audio Logistics, who assembled the final broadcast Dolby Surround mix for the Tenors, comments, 'I love working in Surround, especially for this kind of show, where you can recreate the environment for the audience at home. You have to be pretty careful and subtle with classical, or indeed any, music and I always leave the instruments on the front left, centre, right, adding ambience, reverbs and audience in the rear.'

Although Le Mobile Son, where audio engineering director John Pellowe mixed the main broadcast feed, is equipped for surround, Pellowe wanted to use something other than the truck's Genelec monitors. Audio Logistics brought over three of the new B&W Nautilus 805s, a small domestic loudspeaker, which were used for close-field monitoring. A further three 805s were installed into the audio hub truck, Euphonia.

The 805 is a smaller (40cm high) version of the 801, which is installed in Abbey Road's Penthouse studio, the surround sound suite where Pellowe and Alex Marcou remixed the concert for CD and video release. 'The most noticeable difference between the 805 and previous B&W small monitors is the bottom end,' says Alington. 'They really do sound like big monitors.' Both Le Mobile Son and Euphonia were fitted with Dolby SEU4 and SDU4 surround encoders and decoders.

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From Kashmir with love

One track off the *Godzilla* soundtrack album saw Sean 'Puffy' Combs in LA linking up with Jimmy Page in London via EDnet audio and satellite video. **Richard Buskin** talks to Paul Logus, Bill Smith, Bruce Maddocks and Toby Wood about the project

IT IS THE STANDARD Hollywood formula: make the blockbuster movie, lace it with classic songs performed by contemporary artists, and then supplement the box-office receipts with a soundtrack album. In this case we're dealing with *Godzilla: The Album* and a track that captures attention as much for the way in which it was recorded as for the finished product itself: 'Come With Me', which features axe hero Jimmy Page reproducing his virtuoso guitar parts from 'Kashmir', one of the standout cuts on Led Zeppelin's 1975 opus, *Physical Graffiti*, while producer Sean 'Puffy' Combs does his thing by rapping new lyrics over the top.

1:Tom Morello and Paul Logus fighting over his double-neck Gibson. 2:At Puffy's studio during the mixing session. Jimmy Page(left) Puffy Combs and Paul Logus(right). 3 Logus rehearsing the band at the set. Page was not then in town so Paul Logus had to sit in

Since Combs was in Los Angeles and Page was in London on the day that they worked together, both EDnet and satellite links came into play for a 2-way audio-video hook-up between Capitol Studios and CTS. Yet, for a track that took a total of six weeks to record, that was by no means the whole story.

Paul Logus took care of all of the recording and mixing, and prior to Page's CTS session he recorded the basic drum pattern, a bass part, a rough string part and the guitars of Tom Morello from Rage Against the Machine at the Record Plant in New York. All but the string patch were live, a beat from a drum machine having kicked off the proceedings before Mario Wynans duplicated this on a real kit.

'We'd listened to the original record [of 'Kashmir'] and Puffy had tapped out what he wanted for the drums,' says Logus. 'We recorded that with a machine for about two minutes, Mario played to that, and then I did a quick mixdown to stereo and dumped it into Pro Tools and looped it. That's what ended up in the song.'

In the case of Tom Morello, he played a Les Paul, a Strat, a Telecaster and a double-neck for their various sounds, yet Combs decided that classic Zeppelin was what he was aiming for. This paved the way for Paul Logus, a die-hard Zeppelin fan, to be a little creative.

'We were using two different Marshall cabinets, and two different Marshall heads, and for the miking I used a 57, a 421 and an 87. I also had other room mics up from when we'd been recording the drums, so I just turned things on to pick up different ambiences. In the end we had two stereo tracks of Tom's playing; one is a little more close-miked with ambience mixed in, while the other is just ambience hard-panned left and right. He was playing in a big room, and when you stack those up you have a very big, very Zeppelinesque sound. In fact, a lot of industry people who heard it actually thought that we had used a Zeppelin loop, and that was kind of funny because that was exactly what we were trying to avoid. We were trying to be as original as possible, and so we did everything from scratch.

'The same thing applied to the drums. Puffy had asked me to make them sound like Led Zeppelin drums, so I gave them the same sort of treatment, doing a lot of ambient miking. Mostly that consisted of a pair of mics at the other end of the room—maybe 15 to 20 feet away from the kit—and then a D12 and a U47 about two to three feet away, with the U47 aimed right at the middle of the kit while below that the D12 would be aimed at the kick drum. To me, most of Zeppelin's stuff sounds very ambient, depending on the track, of course, and putting up the room mics and compressing the hell out of them was the starting point, while adding mics that were closer to the kit, but not really tight-miked, produced more definition, and a bit more low end.'

Meanwhile, Logus taught Tom Morello the Zepellin bass part, which was played with a Fender Jazz through an Ampeg B15. Combs also recorded a rough guide vocal, and Logus then mixed everything down into stereo pairs and made a slave that could be sent over to CTS in London via EDnet.

'We went over to Capitol because they have the ability to transfer a pair of tracks at a time locked to SMPTE,' Logus explains. 'They had three EDnet tracks open, but only two of them were digital. The third one was like a dedicated SMPTE track, and we later had problems resync'ing the multitracks when we transferred back...'

Meanwhile...

'The tape that I had consisted of reduction mixes of everything, like a stereo track of drums, stereo track of guitars, stereo track of strings, a bass track and a vocal track. That got transferred via EDnet to England, and the reason we did that was that there was a delay involved in the audio. We were running video at the same time, and so what I wanted to do was, if we needed to punch anything in, just have them [at CTS] do the recording while we could communicate with Jimmy over the EDnet lines and over the satellite picture in order to alleviate any technical problems that might occur.'

It isn't clear whose idea it was to use a satellite link-up for the video, but there can be little doubt that this decision cancelled out any cost-effectiveness gained by accessing EDnet. Perhaps 'ironic' would be an apt description of the situation.

Toby Wood was the engineer capturing Jimmy Page's performance in CTS' Studio 2 on a Neve Capricorn,

Puffy Combs

using a Studer A827 tape machine with an A800 standing by as backup.

'There was a 2s delay,' says Wood. 'Our ISDN was only 1s there and back, so because of that we actually sent a talkback audio—in fact, all audio—via ISDN. We didn't use the satellite channel.'

'What we had was a feedback delay problem and it sounded kinda crazy,' adds Logus. 'We didn't have time to really fix it, and the cost factor was ridiculous, so we kept the EDnet lines open for the whole session and we communicated through the EDnet. Still, because of the delay with that the picture was still running ahead of the audio, and that was a little funky.'

'Jimmy's overdubs just entailed 15 or so takes,' says Toby Wood. 'I ran a very loud monitor mix for him in the studio, because he didn't use headphones. We hired in a couple of Martin Audio wedges and played the backing track through them. He was playing a Gibson Les Paul, and that went through a Fender Combo, recorded with a couple of dynamic mics placed right up against the speaker. We knew we were going to end up with spill, but Jimmy said that he liked the sound of the room, so I just put up a 149 for an ambient track—out of reach of the monitors that he was using for foldback—along with a mixture of 421 and an 87. I stuck those through a TLA C1 compressor-mic amp, using only the compressor side along with the Capricorn mic amp.

'After we'd done the takes we put them onto a multitrack, made a copy of that multitrack and sent it off to Capitol. They needed to mix and prepare a tape for orchestral overdubs, so I sent all of the individual guitar takes back over to them in stereo pairs via ISDN. That way they could put them on their slave reel and get on with whatever they needed to do. Throughout we maintained the same tracking that they had out there in LA, so when they eventually got our tape everything was in the same place as they had been used to working to.'

Nevertheless, when the Capitol engineers locked their tape machine to that at CTS via the SMPTE track over EDnet, it became clear that there was a considerable sync problem relating to the guitar tracks.

'What I ended up doing was to find an offset by ear and synchronise the machines off of the 2-track mix that I had on both of my tapes,' says Logus. 'After that the guitar tracks locked up perfectly, and later I found out that the reason that had happened was because there was a delay in the processing with EDnet that I hadn't been aware of. It's nothing that you can calculate, it's different for every session, and a dedicated SMPTE line does not get processed in the same way that a digital audio EDnet track does. That's where the time difference came in, because we were bouncing the audio through the EDnet digital lines and locking the machines through the unprocessed EDnet lines.'

Logus still asserts, however, that overall the session was 'the most flawless and technical one that I've ever been involved in', while Bill Smith, a freelance, former Capitol engineer who has worked on hundreds of EDnet projects, points out that the approach this time around was somewhat easier than normal.

'When we're doing EDnet with studios in other cities the delay times vary, so when we get up on the air we'll normally send audio to them and they'll take that audio and simultaneously route it straight back to us,' Smith explains. 'We figure out what the delay time is between the two cities, and we've got a little formula worked out where we convert the delay time into a SMPTE number. We've also created a reduction slave reel from the master tape, and we'll lock up the slave with the master whereby the slave is in advance of the master. We never listen to that slave here in Los Angeles, but we send it down the line to the other city where the artists listen to it. They'll react to it, play along with it and the audio will come back down the line to us, while our original master has caught up.

'So, their signal and what they're doing falls onto our master tape in perfect sync, yet the disadvantage is that punching in is a pain. You need to unlock the machines, clear out the offset, copy what you've just done from the master to the slave, put the offset back in, get the tapes back to where they should be, and then you play them what they've just done off of the advance tape and when they react you punch in on the master.

'With Jimmy, on the other hand, we just sent CTS a 2-track mix and SMPTE down the line, and they recorded that onto their multitrack and had Jimmy play along. They actually ran the session from CTS and punching in was much easier, so for us most of the day consisted of just hanging back and listening.'

Paul Logus had 15 Jimmy Page guitar parts to choose from later at the

PHOTO: CHRIS FLOYD

Jimmy Page with an old friend

Record Plant, comprising rhythm tracks, free-form tracks and solos, as well as one with the trademark Echoplex sound that is being included on a remix.

'I could see the Echoplex as I was watching Jimmy on the screen, and so I asked him if we could have a track of that,' recalls Logus. 'He said, "It's a bit noisy, don't you think?" and I said, "Yeah, but that's your sound! We like it!" So, he was kind of funny about that, but he did it."

Page's gold-top Les Paul was in fact fitted with a Trans Performance L-CAT (Computer Assisted Tuning) system, the DTS1, which was the first version of this device. Another of his Les Pauls has a DTS2, that features a second row of presets for concert tuning, but apparently this had already been shipped for an imminent tour.

In all, Logus incorporated a couple of Page's rhythm tracks, an Echoplex track, and a solo, into the master reel at Record Plant. The actual 'Kashmir' riff was then looped for Combs to use wherever he wished throughout the track, and thereafter the bass part was fine-tuned, guitars were layered, some Combs vocals were recorded and synth string parts were played by Jay Dub as a guide for orchestrator Jeremy Lubbock. The orchestral recording subsequently took place on a Sony Studios scoring stage and sub-mixed to a stereo track.

'As we recorded things we generated slave reel after slave reel,' Logus recalls. 'So, to keep things more concise I ended up doing submixes much in the same way that I had for the EDnet session. As a result, through much of the recording process we were basically dealing with just one 24-track tape.'

While the mix itself only took about half a day, Paul Logus and his colleagues spent a lot longer assembling all of the different versions of the song that were requested by either the film company people or record company personnel.

'We printed like a hundred different versions or more,' he states. 'It took us almost a week to print everything. It was crazy, but we had a lot of fun.' ■

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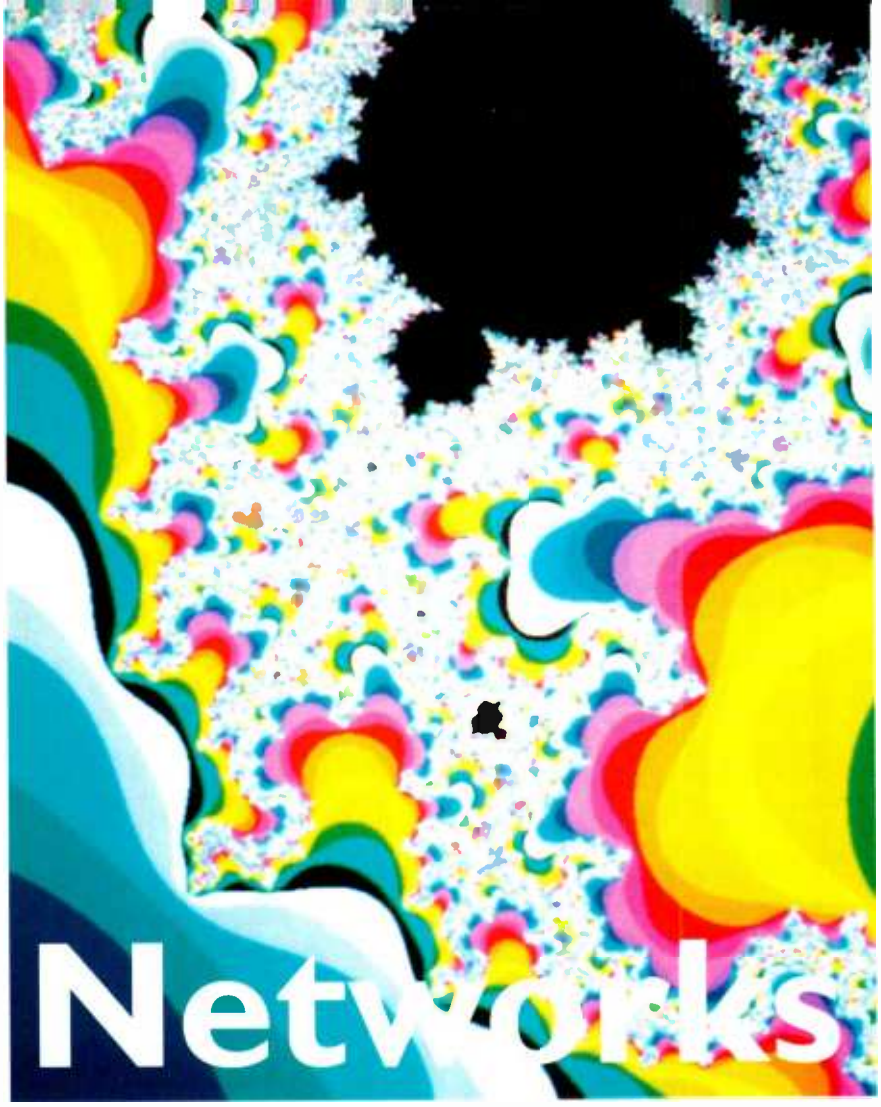

Networks

In the first of a series of articles, **Rob James** examines the background to digital-audio networking and identifies the key questions in choosing the best system

IN THE EARLY NINETIES, I was involved in a heated debate over the merits of networking as a means of 'improving' postproduction sound processes in television. With the technology available at the time the only viable answer was eventually shown to be removable storage and 'sneaker-net'—to physically move material between the various stages of the post-production process. We are now eight years further on, technology has developed and things are not quite so clear cut. What is clear from the multitude of product announcements is that the manufacturers believe they now have viable products to offer. Rather than jump head first into a discussion of the relative merits of what is on offer, let's look further at the context. We can avoid a detailed discussion of the various connection technologies and topologies of networking in this article partly due to space but mostly, simply because they are not particularly relevant to the professional end-user.

To get the buzz words out of the way, you will hear Ethernet, 10 or 100 base T, ATM (Asynchronous Transfer Mode) Fibre Channel and Firewire bandied about by some vendors of networking solutions. It is a highly seductive notion that specifying, say, a fibre-channel network will guarantee success. Unfortunately life is not that simple.

Simply asking what technology a network is based on is almost invariably to ask the wrong question. The questions that need to be asked before considering the introduction of any new technology concern accurately identifying the problem and determining whether any of the proposed solutions actually solve it. Of course, assessing the effect this would have on your bottom line is important as this is not merely a question of cost reduction or justifying the raising of rates. The reality is far more complex—improving quality, speed and convenience to clients, or even just changing their perceptions can produce better profit figures even if there is no obvious direct cost reduction or opportunity to raise rates.

In order to achieve success, it is essential to assess your existing and future requirements accurately and to try to ensure all the hidden costs are taken into account. Further questions that need to be asked concern the changes that will have to be made to procedures and ensuring the proper evaluation of all the implications and costs? Reliability, security of material and continued support also need to be taken into account, as do concerns over longevity. Next, training and staff implications should be addressed - who will maintain the network, take care of the housekeeping, repair and archiving? Consider this: if losing the contents of one DAW can be highly inconvenient, embarrassing and time consuming, then 10 workstations are sharing the same material from the same server, the survival of your business in the event of a disaster has to be in question.

The keys to success in the introduction of networking to real-world processes are initially overcoming the physical limitations of bandwidth and connectivity, process analysis and system design, and, arguably most importantly, proper management to control and minimise risks.

At its simplest, networking can be taken to mean the physical connection of two or more devices, and the transmission of data and-or instructions between them.

There are many ways of achieving this. Conventional 'office' data networks are one possibility, but are far from being the only one. From one viewpoint, a facilities house already has a variety of 'networks' carrying analogue and probably digital audio, video in any of several flavours, time code, various control signals, and, of course, telephones. Picking up the phone to a central library and asking a runner to bring a project to the room is a valid networking solution. The control is the phone, the transport, hands and feet. There are obvious disadvantages—the time taken to find the relevant project and physically move it, the risk of damage in transit and the fact that access is restricted to one operator at a time unless the project is copied. If the job in hand requires access to multiple projects things get trickier still. All these factors are just as significant when considering a more hi-tech approach.

Because of the computer industry antecedents of many audio workstations, there is a temptation to look at the technology and methodologies employed in that industry and extrapolate them into audio and video. With conventional office networking, the volume of data is comparatively small, the time taken to move it relatively unimportant and synchronisation as we understand it, irrelevant. However, moving even a single high-definition image or high-quality audio files over a conventional office network (or worse, via the Net) can be mind-numbingly slow in operation. Worse, it is comparatively easy to clog up a conventional network in this way, slowing things down for all other users. With current technology the only viable answer for most applications is a >

< dedicated network for audio (and possibly video) use.

The currently fashionable networking models for audio fall into two main groups. The first is based on connecting existing stand-alone workstations to a central server to allow material to be moved between them, at whatever speed the connecting technology can manage, for later use from local storage. Depending on the volume of material and the technology employed, this could be significantly slower or faster than real time. The second model has all significant storage centralised and therefore requires full real-time connection bandwidth appropriate to the intended use of the workstation. This might involve 4 channels of real-time audio for a simple radio-news editing station or 100 or more for a complex sound-for-picture station.

One desirable goal is to provide multiple access to the same material for multiple concurrent users in real time. With currently affordable technology, the realistic bandwidth limit is around 120 real-time channels of audio on any given network at any one time.

Because of the limitations, most practical network solutions use a combination of central and local storage, the object being to provide real-time auditioning with selected material transferred to the local drive(s).

Radio broadcasters have been looking for a complete networked system for acquisition, programme production and transmission. In a news context, this would allow external feeds to be recorded into a central server with editing workstations given access to material appropriate to them, editing and packaging it and returning the completed items via the network for transmission direct from the original recordings.

In feature film postproduction, the use of a team of operators employing multiple rooms working on the same project is the rule rather than the exception. This is partly due to specialisation with different people working on individual aspects of the tracklaying, dialogue, Foleys, spot effects, backgrounds, music, and so on, but these days it is also due to cash flow. Megabuck investment in productions results in the compression of schedules. The faster a film is completed the smaller the gap between expenditure incurred and the backers beginning to see a return on their investment. This can result in very large teams of people cutting and tracklaying a movie. One room might only work say, on the effects for a couple of reels. To add to the project management complexities this way of working obviously introduces there are the inevitable picture recuts. Gone are the days when the picture cut would be 'locked' before tracklaying commenced. Even if we assume the original audio and video files remain inviolate and all the satellite workstations merely produce lists of instructions as to what operations should be performed on the material, when it should be played and from what outputs—in the form of EDLs, there remain the problems of version tracking, general management and housekeeping-archiving of material. As an example, this could mean a room working on effects tracklaying with a networked DAW would copy the required guide dialogue, and so on, and possibly picture across the network into local storage. Effects are auditioned and selected from the central server and the selected effects copied to local storage for further editing, treatment and tracklaying. Specially recorded effects could be recorded into the local DAW and uploaded onto the central server to allow access by other operators. Alternatively, recording could be restricted to the server. This is one way to ensure all material used in a project is logged and available to all.

The right sort of questions, here, concern the types of access restriction, ensuring all the contributors to a project are working to the same version of the picture, how easy it is to change the version of picture everybody is working to, and how easy it is to enable multiple access to the same material. But perhaps most ominously, you have to ask: 'Is it safe?'

One obvious candidate for a networking solution, a sound-effects library in a major facilities house, can be used to illustrate some of possible alternative approaches. The luxury approach is to provide a complete set of effects in each room, with a printed catalogue or perhaps a PC running a database programme to aid in locating specific effects. This is expensive both in the cost of the material and also physical storage space. The pre-PC approach to managing this requirement was to have a central library with printed catalogues where occasional users could browse for effects, visit the library to audition them if required, then order transfers to the appropriate format. High-volume users would have a subset of the main library available for instant use. A 'perfect' IT solution to this requirement would allow the required number of work areas instant access to any required effect via a user-friendly search engine along with logging of effects used for statistical and, potentially, billing purposes. Perhaps the most elegant of the currently feasible networking approaches is to install a central server containing the library on a fast-access RAID-Winchester disk storage system which would allow simultaneous access for a reasonable number of concurrent users. Experience has indicated a ratio of around 3:1 is about right to keep costs sensible and avoid frustrating delays - with average use a system capable of supporting 10 concurrent accesses would be suitable for up to 30 users. Users could be 'prioritised' such that high priority users would get access at the expense of others at busy times. Effects would be auditioned and downloaded direct to a variety of manufacturers DAW's via the network in a format appropriate to each. To achieve a decent performance from this kind of system requires terabytes of fast central storage to take care of the hundreds of hours of material, a high-speed network to support the concurrent access and, critically, a common file format for the various manufacturers DAW's. Leaving aside the last requirement which could, of course, be overcome by using products from one manufacturer, this is still an expensive option.

An alternative, more cost-effective, solution employed by several major facilities is to use ordinary PCs to communicate with a dedicated server and network. The server also controls CD juke boxes and an audio router. The CDs can be arranged so there are multiple sets of the most popular effects and single sets of the less popular ones suitably distributed across the juke boxes to achieve optimum access 'hits'. The audio output is piped into the room via the audio router and conventional tie-lines. Selected effects can be immediately recorded into the DAW or marked for batch recording. This approach eliminates file incompatibilities problems. Variations on this theme employ either Winchester hard disk or magneto-optical drives to store the most popular effects for faster access leaving the CD juke boxes for the less frequently accessed effects. This avoids the necessity of having multiple iterations of popular CDs. The latter system is not perfect but has been proven over time to offer substantial operational advantages over the previous manual library systems and to be cost-effective. Most importantly, clients like it.

The messages here are simple ones.

Even when the bandwidth limitations have been solved, as they surely will be with the inexorable rise in speed and capacity of all things to do with computers, there will remain the management problems.

The introduction of networking into a facility can provide tangible benefits. But this will only apply if you follow a rigorous approach to process analysis. Do not be seduced into thinking a conventional data type network is the only solution. Many other variations are possible and may be preferable for a variety of reasons. There may well be several perfectly valid technical approaches to the same problem, but at widely varying cost, even from one vendor.

In a subsequent article, we will examine what some of the major DAW manufacturers are offering. ■

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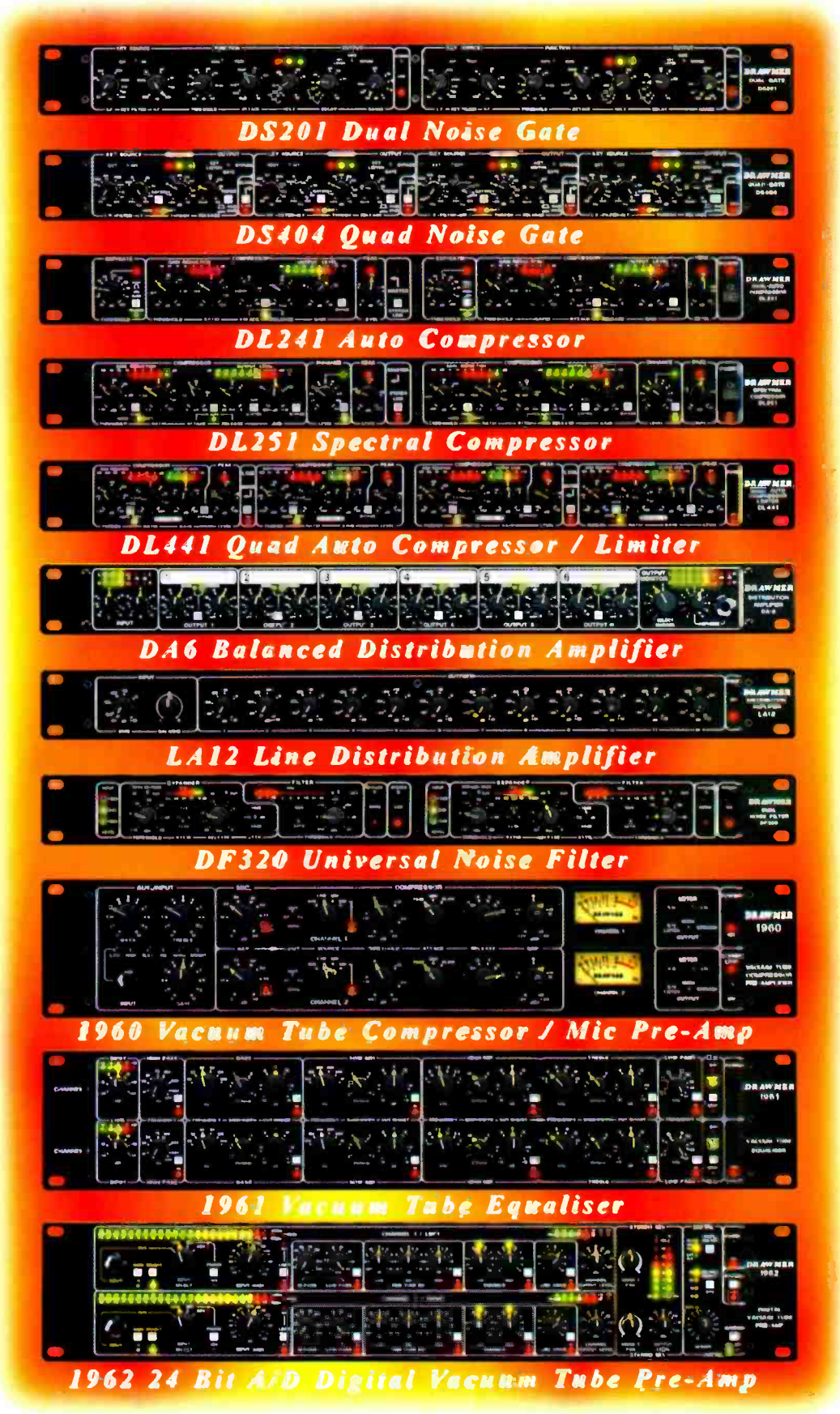

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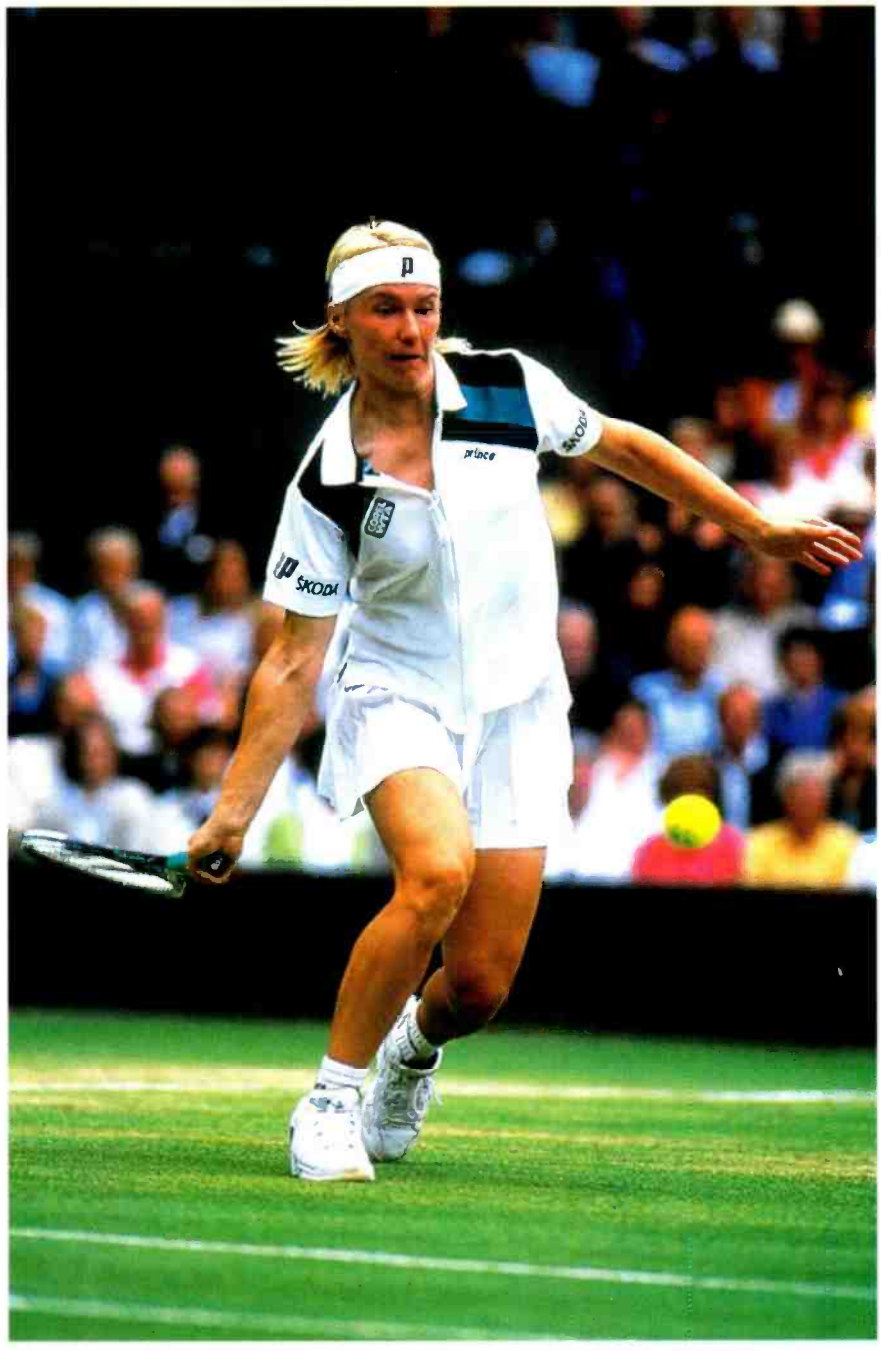

Photo: Mike Hewitt at Allsport

Wimbledon 98

This year's BBC coverage took an unexpected move into surround sound and provided the court for a little experimentation. **Kevin Hilton** looks on

MORE THAN BEING a middle-class district of South London, the name Wimbledon means two weeks of tennis, play interrupted by rain, champions, underdogs, people in silly hats and over-priced strawberries. It is now a truly international competition, something underlined by the worldwide media interest. In recent years a new media centre has been built at the All England Club, which has been at its present location since 1922, and seen the whole range of broadcast innovations: higher definition pictures, colour and stereo sound. This year the event went one step further: Centre Court and Number One Court, scenes of some of tennis' greatest encounters, were presented in surround sound, adding an extra dimension to the coverage.

Not that anyone without a Dolby-equipped decoder or TV set would have noticed—the BBC has displayed an ambivalent attitude to surround sound in general and Dolby in particular. Some inside the Corporation have objected to the matrixed nature of Dolby Surround-Pro Logic, which, technicians have said, can create artefacts in the audio signal (something dismissed by Dolby); the proprietorial nature of the systems also bothers the largely non-commercial organisation. Combined with this is the BBC's position that it is not offering a surround-sound service of any description: while surround content will be broadcast if it is mono and stereo compatible, it is not publicised.

Which is not to say that the BBC has not been experimenting with various types of surround. Rather, it has contributed to the project to develop MPEG 5.1 multichannel audio and produced some of its own programmes in DPL. Since the 1990 Broadcasting Act, the nature and structure of the Corporation has changed, seeing the position of craft managers disappear, leaving decisions on what technology to use up to the technicians directly involved. In the case of Wimbledon, sound supervisors from the BBC Resources Outside Broadcast department proposed the use of surround sound to BBC Production as an added extra that would enhance coverage and enjoyment of the event, at least to those with the necessary equipment.

'The producer, Martin Hopkins, liked the idea,' explains Bill Whiston, one of the sound supervisors who championed the scheme. 'In some ways it is experimental, but it's not quite an experiment because we have played with surround sound before. We thought it would be a good opportunity to do it here at Wimbledon because we are in the same place for two weeks and this has given us the chance to play around with different ideas.'

Whiston acknowledges that surround sound from the BBC is something of a secret service, but says that work has continued on what can be done with the technology even if not many outside the Corporation are aware of it.

'We haven't made a big noise about it, but in much the same way we started stereo services surreptitiously—nobody shouted about stereo, the technicians were interested in doing it and thought that they could. There has been interest in surround within the BBC for some time, but Dolby Surround has its limitations.'

Despite these limitations, which Whiston says include its stereo front pair and mono rear format, it was decided to see how far the Dolby system could be used to bring an extra dimension to the coverage of Wimbledon.

'We wanted to see if we could add anything over and above what is normally done with football coverage, where the convention is to sling a 635 mic over the crowd and put the output into the rear loudspeakers,' Whiston explains. 'We wanted to be more adventurous. If you look at movies, they >

< stick all sorts of things into surround and in that way we wanted to give a sense of the atmosphere of Centre Court, which is a very special place.'

The standard miking configuration for past stereo-only Wimbledons has been a main stereo mic at the near end of Centre Court and a single 416 at the far end. Other court mics are placed in the stands, giving the operators a background ambience to fade to when sounds are too drastic on the court itself; for example, on the occasions when the covers are brought out to protect the grass from the rain that is as much a part of the tournament as arguments with the line judges. 'We have been able to play with this setup as long as we don't jeopardise the output and can improve the coverage,' Whiston says of adding surround. 'There are some additional mics at the back of the base-line pointing at the crowd, which comes right down to the edge of Centre Court, so we were trying to recreate that kind of feeling through the sound.'

Mic feeds for Centre Court were fed into one of BBC OBs' Type 8 scanners, Unit 1, which, like Unit 3, the truck covering Number One Court, is equipped with a 48-channel Calrec Q-series desk. Dolby encoders were removed from OBs' two big trucks, the Central Mobile Colour Control Room and the Digital Mobile Colour Control Room, and installed into Units 1 and 3, along with two pairs of Jamo domestic loudspeakers as the rear monitors, driven by Quad 520 amplifiers. Of the 19 courts at Wimbledon, only Centre and Number One were mixed in surround this year; the other main venues were straight stereo, with the lesser courts in mono. Within the media centre there is a sound control room; although it is still referred to as the 'sound hut', for the show courts, apart from Number Two Court, which has its own scanner. Output of the three sound mixing booths and the commentary technical areas within the media centre are sent to Unit 3 for final transmission.

Given the standing of Wimbledon, not only in tennis terms, but also in the general world of sport, the expectation would be that Centre Court is an enormous amphitheatre of the racket. This is not the case; it is smaller than Number One Court and is described as an intimate venue, something underlined by the crowd being so close to the action. Whiston says that the diminutive nature of the court brings its own problems.

'The dynamic range is enormous. One of the problems of surround is that, if you are going to give a true surround mix, you still need to hear the commentators above the roar of the court.'

To ensure that the prescient observations of the commentators could be heard over the increasingly hysterical reactions of the spectators, Whiston risked annoying his wife by taking a VHS copy of the first day's coverage home and cranking it up on his hi-fi system to hear how the mix performed.

'What I had to bear in mind,' he says, 'was keeping the commentators not just above the stereo mix, but also above the surround-sound mix.'

To fulfil his self-imposed brief of doing something different with surround to the standard one mic for the rear approach, Whiston created a group on his mixing console specifically to feed the surround processor, which he called his Dolby Playground. Into this he mixed different combinations of microphones in a conscious attempt to do something different from the commonplace method of achieving surround.

'What we've added to the back loudspeakers are splits of the audience mics,' Whiston says of assembling the surround mix. 'We didn't want to add the court mics to the surround mix because we thought that would be confusing for the viewer. What was going into surround were the audience-crowd mics and whatever was in the playground, with a couple of splits of crowd mics to both front and back. We noticed that we were getting a 90° coverage of people, which was probably because of the slight delay that is built into the Dolby system.'

With the extra dimension of surround, Whiston says that particular effects were created, some that could be used creatively, others that had to be got used to.

'With the extra mics we could get to the point of almost picking out couples in the audiences,' he says, 'getting back-of-head chats, which all added to the general murmur that was useful to bring up when the players were sitting down between games. However, this was only added to a certain degree; murmur was all it was, there was no detail. There are also some very interesting effects that we've noticed when monitoring in the scanner. The addition of surround gives a different impression of height and with the left and right coverage of the people round the edge of the court, you can hear individuals calling out to encourage the players.'

Much the same approach was taken by the sound supervisors in Unit 3 on Number One Court, with individuals experimenting with different elements on different days. Whiston says that no mention was made by the international broadcasters, confirming that the addition of surround did not affect the main audio output. While the experiment was deemed to have been successful for Wimbledon, Whiston does not think, given the present structure of the BBC, that it signals a Corporation-wide acceptance of surround sound. So, for the time being, surround continues to be a secret service. ■

London – At the world famous Abbey Road Studios. "Our GX8000s move between the studios and on location for use on a wide range of projects." Neil Aldridge, Chief Engineer.

Nashville – Mastering the new George Strait album at Georgetown Masters. "Everyone was astounded when they heard it at 88.2kHz compared to 44.1kHz. Its reliability has also been amazing." Denny Purcell.

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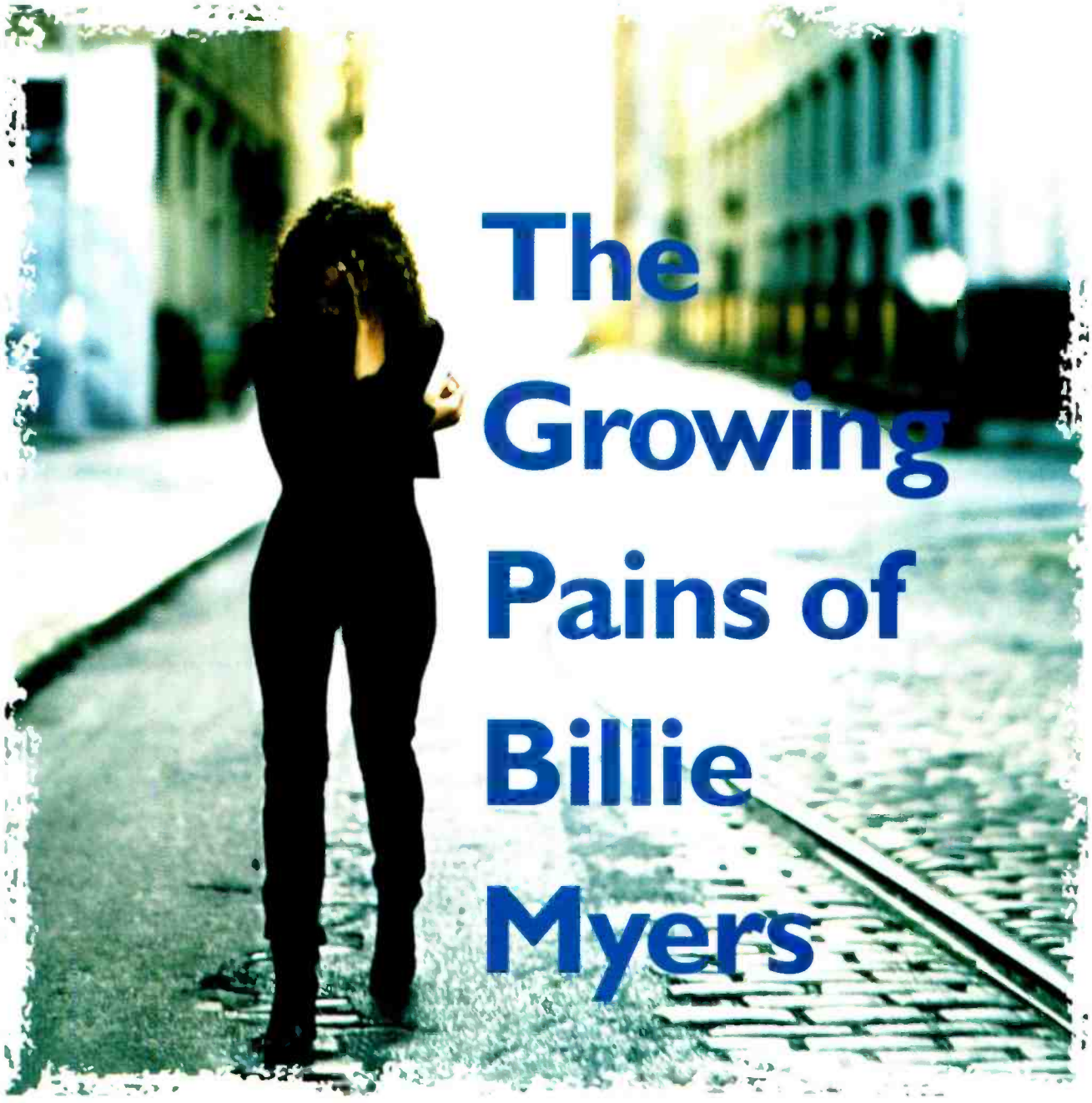

The Growing Pains of Billie Myers

Spread a little happiness... **Richard Buskin** talks with Billie Myers' engineers about the techniques and advantages of remixing in 5.1 surround.

TO MY MIND there is little doubt about it; 5.1 surround is no longer a novelty, but the sound of the future. The only question is, when that future in music will be?

'Things are moving quicker than we expected, but I still imagine it'll be about ten years before surround replaces stereo in the home,' says DTS [Digital Theatre Systems] production manager, Peter Lewis, while I'm being treated to a demo of different 5.1 remixes and comparing them with their stereo counterparts. A-B testing a number of disparate tracks, what stands out is not only the obvious advantage of having extra channels, but also the variety of ways in which different engineers choose to use them. At one end of the spectrum there is a Bonnie Raitt live recording that provides a perfect sense of the venue dimensions and room sound, while at the other end a Boyz2Men studio performance has each of their voices coming out of different speakers while a percussionist wanders from side to side at the rear.

Still, the main purpose for my sitting in the home setup of Ric Wilson, a Northridge, California-based engineer who specialise in 5.1 remixes, is to listen to the surround version of Billie Myers' debut CD, *Growing Pains*, as remixed by Charles Dye. An engineer for the past 15 years, Dye has worked with Gloria Estefan, Jon Secada, Jon Bon Jovi and Lynyrd Skynyrd, among others, and he recorded *Growing Pains* at Criteria in Miami as well as at The Gentlemen's Club that he runs.

Kenny Aronoff drummed and Hugh McDonald played bass at Criteria, while Eric Bazilian's guitars, the keyboards and all of the vocals were recorded at The Gentlemen's Club using a 48-track 24-bit Pro Tools system that Dye refers to as his 'Swiss Army knife for audio'. However, it was at Del Rey Beach, Florida, in the studio, named Audio One —owned and operated by engineer David Frangioni and his wife, singer-songwriter Jenna Drey—that Dye did the 5.1 remix. Catering to 5.1 as well as regular stereo work, Audio One houses 80-inputs of Yamaha 02R console with 20-Bit ADAT machines, Tascam DA88s, and a loaded Pro Tools 24 system, as well as an assortment of outboard gear that includes names such as Lexicon, tc, Eventide, Drawmer, Focusrite, Fairchild and GML. Apogee 24-Bit AD8000 and Prism 2024 convertors are

used for monitoring.

Frangioni, who over the years has engineered on projects by Aerosmith, Elton John, Chick Corea, Bryan Adams and The Rolling Stones, prides himself for always having been on the technological cutting edge. Consequently, after being commissioned to handle the first three songs to ever be mixed in DTS 5.1—as recorded by Seal, Boyz2Men and Shoeless Joe—he was immediately smitten with the format and decided along with his wife to tailor a mixing suite to its requirements. Audio One opened its doors on the 1st March, 1998.

'If you have a great 5.1 room 2-channel work is no problem,' he says. 'However, a great 2-channel room does nothing for 5.1 necessarily. I mean, you can't just drop in the extra speakers. That is the wrong environment to mix in, and that's because of the live-end/dead-end design that a lot of control rooms have these days. The live end produces reflections for the engineer that help recreate the home environment and therefore improve the monitoring of stereo, but you can't have a live end in a 5.1 room because it won't give you the balanced frequencies coming from the rear as well as the front speakers. You must have a room with absorbent materials almost all the way around, and so that's how the room here at Audio One was designed from the beginning with 5.1 in mind.'

It was, in fact, Frangioni who originally approached Charles Dye to take care of the 5.1 remix for *Growing Pains*. After all, it was not the first time that the subject of surround had been broached. The two men are good friends, and, according to Dye, Frangioni had been trying to sell him on 5.1 for quite some time.

'To tell the truth I didn't really think it was going to take off,' Dye now admits. 'I remember David telling me, "Charles, I swear, once you mix in 5.1 you'll never want to go back. That's what everybody's saying," and I thought, "Yeah, right." It sounded like such hype. I wasn't going to believe it for a second, but after my first few days mixing "Kiss the Rain" [the first single off the album] I was going, "Wow!" You can do all of the things in 5.1 that, as an engineer, you have always attempted to pull off.

'Out of two speakers you're always trying to extract space. You're trying to juxtapose a small space against a large space and nearness against distance, and with 5.1 that's simple. If you want to put something in a big hall and make the listeners feel like they are there, it's easy, because you've just reinvented the environment of their room, whereas whenever you're listening to stereo you're not a part of the experience, you're just observing it. 5.1 puts the listener into the song and into the music, and I realised that during playback it affected me emotionally in way that had never happened before with other mixes. That in turn also allows the mixer to affect the listener to an even greater extent emotionally.'

Sample the passage about halfway through 'Kiss the Rain' where the real drums come in as a prelude to the guitar solo. In 5.1 the effect is far more dramatic, the listener virtually being immersed in the thundering sound of drums and guitars that emanates from all angles. Nevertheless, while it is easier with five speakers than with two to make a quick mix in which all of the musical elements are immediately discernible, there is also a danger with 5.1 of losing some of the impact that can invariably be attained in stereo.

'It's like the rhythmic drive that you get when you have the hi-hat panned in the middle as opposed to it being panned to the side,' says Dye by way of example. 'That's even more diluted when you have crap coming at you from behind. I mean, a mono mix coming at you from one speaker can have a very strong impact, but as you start to pan things out to the side those elements no longer have drive. I'm sure people learned that in the early days of stereo, and that's obviously why we've ended up with the bass, the kick and the snare in the middle.

'Being that I'd worked on this [*Growing Pains*] record I already knew how things were supposed to sound in stereo, and so I would have to concentrate on making sure that the various elements didn't lose their drive. Now, one way of doing that, of course, would be to put all of the driving elements back where you always have them, in the front, but that would be really boring, and a real waste of the rear speakers. I have a basic principle: the guy at home is paying the same amount of money for those rear speakers as he did for those front speakers, so I'd better put music that justifies it back there. I mean, if I just give him ambience he's going to be really put out at having spent $500 on rear speakers that have reverb in them.

'As a result I still keep the kick and the snare up front, but in the case of Billie's material the production lent itself to having elements come at me from more than one direction. For example, "Kiss the Rain" has two kick drums >

In Del Rey beach, Florida acclaimed recording engineer Charles Dye (seated) and Audio One studios owner David Frangioni (standing) are taking singer-songwriter Billie Myer's CD Growing Pains, where only a few others have gone by remixing it in 5.1 digital surround sound at Audio One. Dye recorded Myers' mega hit debut album, which includes the hit 'Kiss the Rain' last year

< and two basses, so one kick drum is in the front and one is in the rear. There again, a number of her songs have drum loops, so in the case of all but one song I found that it worked to have the kick and the snare up front and to put the drum loop in the rear. With that equal energy you would actually feel a kind of pushing-pulling sensation from the rear to the front. It wasn't so pronounced, unless maybe you solo'd up the drums, but in the music you'd notice at times what was going on in the rear and at the front.

"That, therefore, is the general rule of thumb for me. It's easier to get that driving sound out of two speakers, so now I just need to place the elements and EQ them and compress them in such a way that the original intent and impact is still there. It's not going to sound the same, but it's not supposed to sound the same. It's 5.1, not stereo."

For that matter it's also not quadraphonic. Nevertheless, going into the *Growing Pains* project Charles Dye did study some old quad mixes as well as a number of newer 5.1 efforts in order to hear how other people worked with extra channels, before then coming up with his own method of placement.

"If it didn't occur to me right away where to put something, then the pattern I came up with was to just spread the left and right out," he explains. "In other words, imagine hard left and right

being rear left and right. That was pretty much how I made a number of decisions. I placed the hi-hat immediately at your left ear—I have always done drummer's perspective, so my hi-hat is always in the left. However, my hi-hat is never hard left and so that is why it wasn't in the rear left this time around; it was panned immediately between the two left speakers. At the same time the tambourine was pretty consistently panned between the two at right.

"Basically, therefore, the idea here is that you're just opening the field up 3-dimensionally, and so that was one of the rules that I used; although there were one or two occasions when it did not work. On "Opposites Attract", for example, I put the loop in the rear as I always had and that destroyed the

groove. It's a lighter, more acoustic song, and that's probably why, so I ended up putting the loop in the front and I just made sure that I had other elements in the hole back there.'

Still, had he known during recording of the Billie Myers album that it was going to be remixed in 5.1, Dye points out that he would have recorded the drums differently, using a quad microphone for the overheads and four mics instead of two for the room sound. That, however, would be it. 'I hear a lot of talk about 5.1 recording but for the most part we don't record in stereo either,' he points out. 'I don't know what people are talking about. Most of the time we record in mono, not stereo, often just to save the tracks.'

Meanwhile, another of the rules that Dye formulated for the Myers project related to the fact that, from the start, he had a firm intention for the remix to be aggressive in its use of surround. Not for him just some reverb in the rear; on this record there would frequently be an instrument taking centre stage while coming from behind the listener's head.

'My idea was that if something would have been panned in the centre then the logic would probably still apply to do that,' he says. 'On the other hand, if it would have been underneath the lead vocal and we would have had to do special tweaky EQ to it or pan it at 0.5 one way or the other, I'd put it in the rear. That creates the lower volume as well as the somewhat tweaky EQ insofar as it isn't masking the lead vocal, while keeping it centred in the mix. For the most part you're saying that anything in the centre—whether it be front centre or rear centre—is an important element, a lead instrument. In other words, by putting it in the front you're asking that most attention be paid to it, while putting it in the rear means it's a melodic lead instrument underneath the volume of the vocal.

'Meanwhile, since the lead vocal was coming from the front, in most cases I would have the backing vocals coming at me from somewhere over the listener's shoulders. Not always the rear speakers, but sometimes panned a little further forward. For instance, if there were two sets of background vocals then I would have some come from the front, panned out wide, and the others coming from the rear. However, there were also a few cases where the backgrounds needed to be loud and in your face, and so at those times I found that the best place for them was immediately left and right, split halfway between the front and the rear, and immersing the listener.'

Overall, the results in the case of Charles Dye's 5.1 remix of the *Growing Pains* album amount to a sound that is not at all gimmicky, but simply richer in texture, and, not too surprisingly, more all-encompassing than the still-relevant stereo version that had served as his reference.

'I would never create movement unless I already had it in stereo,' asserts Dye. 'The delay on Billie's voice on "Kiss the Rain" pans back and forth—relating to the sound delay on the international phone call that's mentioned in the song—but for the most part I'm an autopan hater. I like my mixes to have placed instruments, so that your ear is taken from one position to another when an instrument starts or stops playing or goes to a particular pitch that brings it out. For me the same rules apply to 5.1.

'That having been said, half the fun is that there really are no rules right now, and that's somewhat scary because it places the burden on you to establish the rules at this point. I'm very conscious of the fact that this is the mix of Billie's album that everyone will be listening to in five or ten years, and so you can be sure that I wasn't about to do anything goofy.'

'One thing that we have to keep in perspective is that music dictates how we use the technology, not the other way around,' adds David Frangioni. 'Given that we have 5.1 speakers, full frequency response in the five main channels, an extraordinary low end to put into the subwoofer and 20-bit fidelity to play everything back, it's now up to us as to how we use it. The extra channels beyond the traditional two can be used in any way that you feel is best suited for that music, and what I think makes Billie Myers' album so well suited to the format is that there's so much production. When you have a pop album that has great songs and tons of production it lends itself really well to a 360 soundfield, and that's exactly what *Growing Pains* is all about.' ■

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Tokyo Opera

Modern audiences place great demands on established opera houses. **Zenon Schoepe** reports on a new building

IN THE SAME WAY that emerging markets are leap-frogging generations of technology, countries responding to the changing entertainment demands of their populations are investing heavily in new facilities. Taking the example of opera, it is ironic that the established strongholds of the art are continually struggling with the sums of up-keep, modernisation, and upgrade of old theatres while new markets think only of what would be best. Countries not bound by this inherited tradition, and the need to preserve, can approach the requirements of a modern audience with ground-up builds, as in Tokyo with the recent completion of its New National Theatre. The centre of operatic operations in the Japanese capital moved from its previous home of 20 years near the city's Supreme Court to an all-new rather plush and western-looking complex last year. The attraction and popularity of opera in the country cannot be overstressed, and cleanly satisfies the entrance requirements of decadence, difference and 'imported' that it shares with the other favoured Japanese pursuits of golf, fly fishing and exotic single malts.

The old facility was designed from the outset to handle traditional Japanese drama, and essentially adapted itself to the peculiar demands of opera, while the new complex is dedicated in its entirety to opera. The new building was opened on 10th October last year with more fine finishing work completed in February. Tokyo previously had no theatre dedicated to opera, and neither for that matter has anywhere else in Japan; although it has to be added that plans for such a complex were first hatched some 30 years ago. The theatre is booked solid well into 1999, seats 1,800 and it is reassuring to hear that the Japanese also complain about opera being too expensive for ordinary folk.

The in-house system is Meyer, and acoustic plans were laid down by a specially appointed technical committee with representatives from the numerous disciplines involved. Senior sound engineer at the theatre Kunio Watanabe agrees that the biggest obstacle in the presentation of opera is the need to hide any sound reinforcement from the purists. Four rotating stages precluded the positioning of speakers on the sharp end of proceedings, and the Meyer system is secreted craftily in the proscenium and the ceiling with some hanging boxes.

The same purists may be moved in a similar way by the inclusion of an AMS Neve Capricorn for the theatre's front-of-house duties. Capricorns have become a popular choice for theatre recording duties—such as the two at Germany's WDR—and live broadcast production, but the New National Theatre Tokyo is the console's first time in FOH colours. However, the most fascinating feature of the theatre's desk is the novel implementation of a motorised meterbridge, that, at the touch of a button, folds back giving an unobstructed view of the stage from the control room. Watanabe claims the desk has no obvious operational benefits for opera work, but qualifies his comment by pointing out that the technical committee was adamant that it should opt for a digital console in this theatre.

'It was judged the best sounding digital console and that's really what decided it,' he says. 'It performs the function of front-of-house desk, but we also make live recordings here. One of the points in Opera is the complexity of output routeing required and the Capricorn is certainly capable of that.'

The control room runs Akai DD1000s for manual effects firing; although a future software release for the Capricorn promises to integrate this function into its snapshot capability.

Slung over the front of the control room window are six Genelecs speakers arranged as pairs allowing monitoring of the various speaker outputs. The readiness of live sound for digital consoles is a much discussed topic even though the technology is frequently employed in live broadcast which is subject to very similar one-take pressures. Although the New National Theatre is a comparatively controlled environment to life on the open road, Tokyo may well be giving a pointer of an application of technology that is still to come. ■

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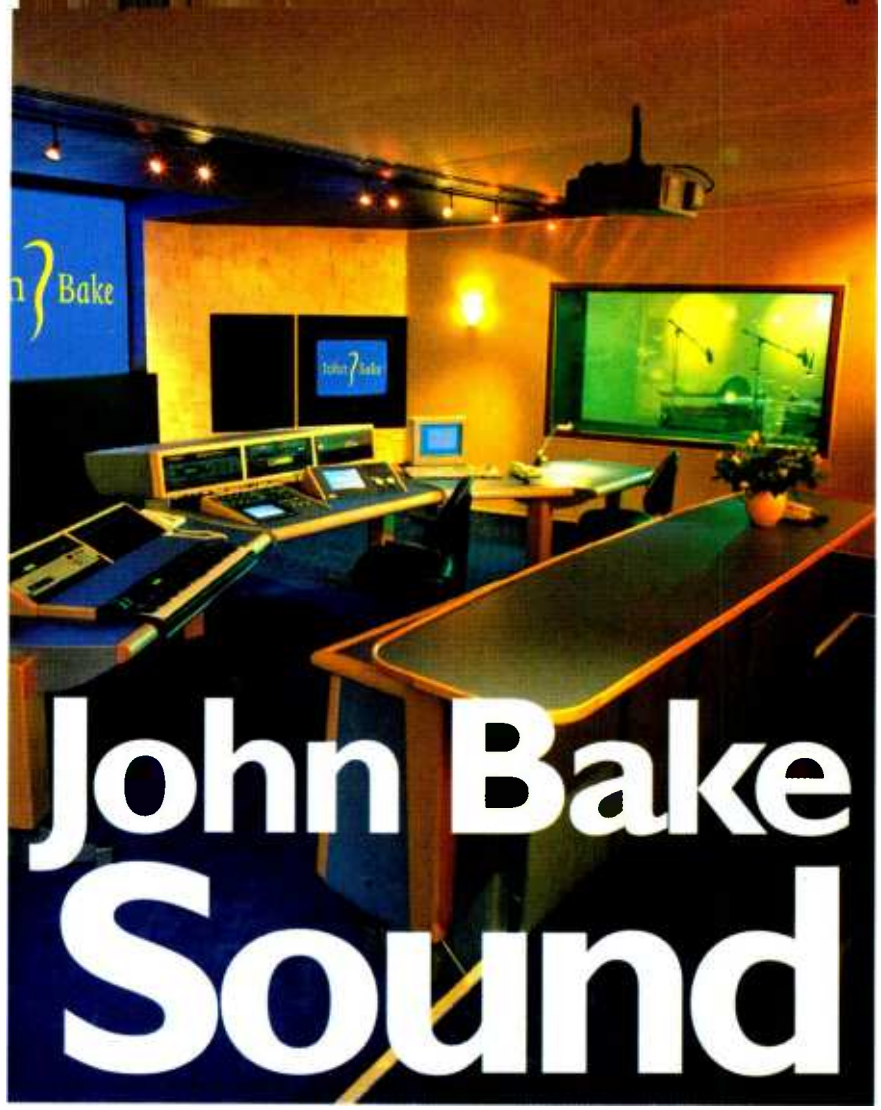

John Bake Sound

Post may be evolving, but the lot of the clients is changing dramatically. **Zenon Schoepe** reports on one facility's view of future roles

John Bake Sound, a leading audio postproduction house based in Amsterdam, recently completed its relocation from the centre of the city to new premises on the city's ring road. While the fact that the changeover was accomplished over a single weekend with prewiring and commissioning being completed at the new site to the point were only the desks and DAW's had to be transplanted is interesting, yet more intriguing are the plans for the new complex. Owner John Bake has some 1100m² at his disposal and three AMS Neve Logic 3s already sit in three large comfortable control rooms with associated studio areas. A film mixing theatre is being constructed, and a number of preproduction rooms and offices are planned in the upstairs area.

The clients have followed him out to the new complex—not just through loyalty, but for practical reasons. The city centre may have once been the creative hub of the area, but parking is now almost impossible, and Bake claims that clients can now get to him faster than before by using the ring road, particularly as many have now also moved out from the centre. The new facility is situated at one of the major intersections of the ring road, and benefits from good tram, train, and metro connections. It is just around the corner from the still fairly new futuristic 'big beetle' Ajax football stadium. This has attracted a wealth of investment (in the region of £1bn, UK) that is bringing the formerly neglected area into one of the up-and-coming parts of the city.

It is also not far from Holland's premier film mixing studio Metasound (*Studio Sound*, July 1998) and indeed the commercials stronghold of Vonk Sound (*Studio Sound*, September 1997). The area is a veritable audio hot-spot.

Bake is a dedicated Logic 3 user having added a third with the move to the new premises. 'It's greatly underestimated and very clever technology. It's also very compact and you can sit right in the middle of the room's sweet spot,' he says.

Ask if he's likely to install a bigger Logic for his film mixing theatre and his answer is entertaining.

'I'd like to install three Logic 3s,' he laughs. 'I'll have to think about how we could do that, but I really like the concept. It would also help us because we're used to working in a particular way that transfers to all the studios here. It's why I like working on the AudioFile, when you've learned it, and have it set up for yourself then every function is a hand movement, you don't have to think about it, you can concentrate on the creative side. A few years ago they changed the RECORD button and for a long time I was getting my moves wrong and that demonstrated to me how much energy you waste when you have to continually think about performing simple tasks. Our customers pay us for the creative part, and I don't want to have to think about the technical side when I'm in a session particularly when they don't care about which system I'm using.'

The facility's first toe in the water with picture is a Power Mac running Adobe Premiere, and there will be more to follow. 'The boys here are starting to think video, and in their spare time they work on it, and, occasionally, the customers ask us to do little bits for them,' explains Bake. 'It's how these things start.'

However, an audio-visual room for multimedia is on the cards to tackle what Bake describes as a requirement to look at more disciplines.

'It's not just audio anymore as picture is coming in to the audio studio,' he continues. 'Our core business will remain audio and upstairs we're going to build four preproduction rooms also with AudioFiles and we're excited about AMS Neve's StarNet networking. We knew it would be here eventually so we've built it into our plans.'

The work is mostly commercials for TV, but radio also crops up with the remainder taken up with what Bake describes as 'audio visual work at the very highest level because we're too expensive for anything else.' More attention is to be paid to corporate audio visual work and jingle writing, with the development of in-house writers and facilities. Eight staff are employed and Bake is fully aware that the planned new rooms will put pressure on his ability to staff up. 'Finding new people is one of the most difficult aspects of expansion because they have to fit in with the group here, and, of course, our customers.'

In the past Bake has started one new member of staff at a time so they have time to fit in with the studio and not the other way around. 'We are successful at what we do, and that balance, and the way we work should not be upset. Never change a winning team,' he says.

Bake himself started off in music recording and made the seemingly unusual switch to audio for cartoons 16 years ago. 'I like all aspects of audio providing there is a creative part,' he explains. 'What also drove me was the fact that the music market wasn't doing so well at that time, and even though I had made a good living from it at the beginning prices had started to come down—MIDI was coming and MIDI broke the market here.'

He moved over to commercials four years later which happily coincided >

< with the wider adoption of video as a handling medium in Holland. Around the same time he bought what was Holland's first AudioFile. 'Sometimes the technology pushes you in a certain direction, and I still remember what it was like then—it was a crazy thing to do because it was an enormous amount of money, but it was the start of a new direction for everyone. The commercials market wanted high quality at high speed and the price was not so important; at the same time there was an increase in the number of commercials being made.'

The studios at John Bake Sound all lead off from a large, central, glass-roofed atrium that serves as the reception and relaxation area. It's all very informal, but there is an unmistakable feeling that this is a place of work.

When you enter a studio's anteroom, from which you access the control room and live area, you are confronted by a mass of polished tubing, gauges and taps behind glass that supply air under pressure to heavy-duty 'balloons' (for want of a better word) that the floors are 'floated' on. Pressure can be increased or decreased to change the damping of the floors.

Acoustics for the complex are the result of a collaboration between Bake and Alex Bolster whose handy work can also be seen at Wisseloord. Studer monitors are used throughout for the front wall of the full surround systems which Bake qualifies for the high quantity of vocal editing work. These are supplemented in the low end by Philips-designed 1000W stereo subwoofers with small Genelecs for the rear. The control rooms are large.

Bake is adamant that, although it is widely thought that low frequencies are not directional enough to warrant such a stereo arrangement, he and his engineers can decypher a clear performance benefit in the 60Hz–70Hz region.

'The majority of room acoustics are relatively straightforward and easy to perform, the last 20% is expensive, but it's also the area in which the high quality aspect is captured.

'You have to remember that a good and experienced engineer can work in a bad control room if he understands acoustics and what is happening in the space,' adds Bake, 'but it's a hard way of working. The rooms here are really special.'

The film mixing theatre is large, and, although due for completion at the beginning of next year, is already shaping up nicely with decisions still to be made on the equipment list.

Bake owns the building, and its his money that has been invested. 'It's the way I've always thought, it might take you longer to expand, but if you start with good basic foundations it will happen,' he says. 'I've always been cautious, but I'm always watching for trends and then it's a matter of changing at the right time. I know that I will have to move towards video, for example. Picture and audio are converging and in five years time I think we will be working, and thinking in a different way. Producers are now working at home with their PCs for preproduction, and that will be coming to me via ISDN and ATM. What won't change is the demand for creativity because even if everyone will be able to buy an editing system in five years time, the extra that we can bring to a commercial will still be required.'

However, with his earlier comments about MIDI impacting on commercial music studios, can he not see something similar threatening for post? 'Perhaps, but they will still need to listen in a good environment,' he replies. 'They won't have that at home, we have rooms with big screens, and Dolby, and DTS. My planned preproduction rooms are a way of keeping my customers here because I'll be giving them the tools right here.' Bake detects that the attitude of his clients is changing, and they are, as he puts it, 'becoming more homely, and less rock 'n' roll'.

While there is a reported move towards home working, Bake believes his clients want to go home to spend time with their families and not to work there. 'Everybody knows that when commercials clients are in the studio with their engineer, 80% of the time they're not doing anything—they're waiting, but they have to be there to decide if something is good or not,' he says. 'But give them a PC, and the opportunity to do their email, make phone calls, and do a little on the Adobe, then they achieve more in a day here than they would otherwise, and they're free to read a book in the garden in the evening.

'The future for a facility like this is that we will have to perform more time management for our clients to let them work more efficiently. Time is money for them in the same way that it is for us. Creative people are expensive and frequently they are not patient.' Bake adds that allowing the client to get on with other things during a session has the positive benefit of letting the engineer get on with job without having someone breathing down his neck.

'Clients achieve more and in this way when they leave here they feel that they've finished their work for the day, and that's important for everyone. It's a matter of lifestyle, and we want this for ourselves, and for our clients.' ■

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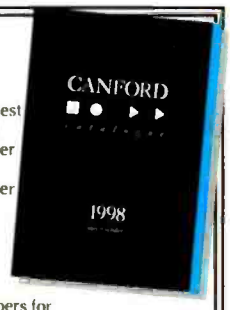

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Dynasty

Whether it is nature or nurture, certain family lines seem more suited to running recording studios than others, writes **Dan Daley**

LAST MONTH'S COLUMN discussed the changing nature of the educational aspects of the pro-audio business. With the entry into the US market of the broadly international School of Audio Engineering, and the interesting response of its largest putative competitor, Full Sail, which announced in July a joint venture with the MARS retail chain to provide pro-audio education classes in what will eventually be nearly two dozen stores across the US, it is quite apparent that the business continues to transition from an apprenticeship-based paradigm to one in which formal training is becoming the norm.

Looking back on the organised counterattack against home studios mounted in Los Angeles by the quickly assembled posse comitatus known as HARP (Hollywood Association of Recording Professionals), the notion of stopping the proliferation of personal recording studios now seems ludicrous, like trying to bail out a flood with a teacup. But even more than the effort itself was the mind-set, one that also in hindsight could be looked upon the way members of the medical professional today regard the use of leeches as a curative in a preceding era. What were people thinking?

Add that together with an industry that's increasingly beholden to public shareholders instead of the misfits in the basements and garages who exchanged a social life for a soldering iron and created all this stuff in the first place and you have a very changed landscape from the pleasant little cottage industry of yore. Based more of the science of marketing than the science of technology, pro audio today is far from warm and fuzzy, and as the ads get literally more salacious as manufacturers use sex to sell gear, a thin but very hard crust forms over the whole enterprise.

But one of the great myths of the cottage industry model that this is all supplanting never seemed to get started in the studio business except in a few rare instances. While other industries built dynasties, the Carnegies and Mellons of pro audio bailed out for the most part long before the eldest son was of age. And that's presupposing that the progeny of this business's pioneers would necessarily have had any interest in following in Dad's footsteps in the first place.

The modern recording studio was born in the 1950s, when independent engineers struck out from the record labels for whom they worked and which owned the few large facilities there were at the time. In the 40-plus years hence, there has been ample time for at least two generations to have come of age, with the earlier one having passed what it started to the next. But it simply didn't happen that way. It's rare to find family-run recording studios. The best example that comes to mind is at Hit Factory in New York City, where Ed Germano continues his Godfatherly role in the facility's operations but where his son, Troy Germano, has taken on the mantle of prince and heir apparent. At the Village Recorder in Los Angeles, which is celebrating its 30th anniversary this year, Julie Hormel, daughter of facility founder Geordie Hormel (himself the unwilling scion of an industrial meat dynasty), struggled to keep the studio open and running as her father prepared for retirement. That dynastic progression was short-lived, however; once the younger Hormel had assembled a new team to operate the studio and equip it with a new round of gear and vision, she returned to an acting career.

But there's an obvious shortage of this sort of royal primogeniture in the studio business, and its absence underscores several things about the nature of the industry. Long before personal studios threatened the community's world order, it was clear that studios were not universally regarded as family heirlooms. They were businesses that could be and were mostly bought and sold before many of their owners got too attached to them.

But it may simply be that this is not a business that lends itself to familial cultivation. It's

The three-year itch

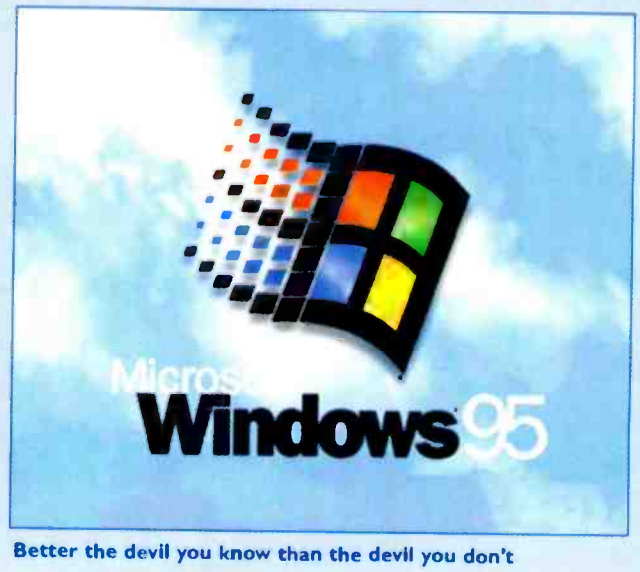
They called Windows 95 'Macintosh 88'. Now it is the turn of Windows 98, but this time the joke may be on you writes **Barry Fox**

SINCE LATE JUNE I have been telling anyone who is interested not to risk the upgrade from Windows 95 to Windows 98, unless they felt very brave, very lucky or have a cast iron reason for needing the new operating system.

Initially Microsoft's line on my situation was quite simple: Windows 98 'unlocks the power of a PC' and the problems I had discovered were one-off anomalies, special to me. But gradually the sheer weight of complaints and warnings is turning the tide. If you use Windows 95 PCs for office or studio work, and your PCs ain't broke, don't even think of trying to fix them with Windows 98. Wait at least six months, probably a year, until the very substantial teething problems have been resolved. Despite the advance publicity, Windows 98 does not provide visible support for ISDN, and it does not come with software capable of decoding MPEG2 video from DVD. So a DVD-ROM drive needs MPEG2 decoder hardware before it can show movie material. The support for Universal Serial Bus connections is only of use if the PC has USB connectors. Older PCs don't. Most important, upgrading from Windows 95 to 98 can cause a disastrous crash.

Windows 95 relies on a 'registry', an electronic index which is automatically updated as the user adds new programs or hardware devices to a PC, and then removes them. Windows 98 also relies on a similar registry which the upgrade software builds by converting the records stored under Windows 95. If the registry is large and complex, from heavy use of the PC over a couple of years, it may —through no fault of the user—contain inaccurate entries which the automatic pre-upgrade check does not find. Windows 98 then cannot build its new registry so tries to run an automatic 'plug-and-play' check on all hardware and software working in the PC. This can take many minutes and may fail. Not only 'power users' are at risk. Anyone who has installed trial software, form cover-mounted CD-ROMs, and then removed it, is at risk. The Windows 95 registry is a fragile beast that can easily lose track of reality and corrupt. But even while Microsoft's Technical Support Team was admitting the problem, Windows Product Marketing Manager David Weeks was still saying that no trouble with the registry had been 'flagged' to him.

Users who successfully load Windows 98 risk being rewarded by the discovery that whole rafts of other devices, such as modems, ISDN terminal adaptors and multimedia cards, no longer work. Most important for studios,



Better the devil you know than the devil you don't

not even really a business that lends itself to families, despite its recent Disneyfication. Aside from the more obvious aspects of the lifestyle that accompanies the artistic end of the industry, even the less outrageously inclined do it for reasons more pathological than just bringing home a paycheck. Much like that legendary New York City pastime of pondering where baby pigeons come from since they're never seen, when you think about the number of hours and their placement on the clock that the typical recording engineer puts in, you find yourself wondering where planned procreation fits in with the scheme of things. And even when miracles do occur, The Life simply doesn't look all that appealing viewed from the progeny's perspective.

The flip side of this is that those who do follow their parents into the industry most likely do so for the same reasons their forebears went into it: they discovered it on their own, seasoned, certainly, with some genetic predisposition. Jason Bonham had every reason to become an accountant or something, anything, other than a drummer (and a very good one, at that). He saw The Life and its consequences better than any cheap biography could ever retell it. But the bottom line might be that, whether or not there is a genetic link from one generation to the next, there is a shared sense of purpose that connects them. We're supposedly all the same under the skin. I think, though, that those in this business are just a little bit more so.

Philips and software company Adaptec are now warning that anyone who has fitted a CD recorder to their PC, should remove Adaptec's driver software, Direct CD, before upgrading to Win 98. I know from first-hand experience that it can cause a full-scale crash.

This warning applies to millions of users round the world. Most of the 5 million CD recorders sold so far were made by Philips, often badged for other vendors such as Hewlett-Packard. And most are bundled with v2 of Direct CD. New version, Direct CD 2.0A, is now available and will solve the problem on some PCs. But those that are more than a couple of years old may have inadequate BIOS software, and still crash or freeze. Adaptec now promises v2.5. And so it goes on.

Both Philips and Adaptec blame Microsoft for the mess. They say Microsoft had promised that software drivers that worked under Windows 95 would also work under Windows 98. Quite simply some of them don't. Microsoft is now issuing a Service Pack of extra software. Gateway is giving a fix-it ROM to owners of its PCs. Dell advises against upgrading Latitude notebooks. New warnings are posted daily. The only thing missing is an apology from Microsoft.

I have solved all my problems with Windows 98 by expunging it and re-installing Windows 95. This proved as tricky as getting Windows 98 up and running. But at least I can now use the devices, like my CD-Recorder, that Windows 98 refused to recognise.

The simplest course of action of all is to stick with Windows 95. You have been warned.

Last orders

One of the qualities of leadership is to make decisions, right or wrong—but it is better if they are right, writes **Kevin Hilton**

But many that are first shall be last: and the last shall be first.
The Gospel According to St. Matthew 19:30.

I AM NOT THE MAN for quoting from the Bible—in general there is little in the Good Book that could be seen as relevant to broadcasting; although 'How the mighty are fallen in the midst of the battle' (Samuel 1:25) could be seen to apply to Silvio Berlusconi; and 'Your old men shall dream dreams, your young men shall see visions' (Joel 2:28) could sum up the hold television has on the younger generations.

The wisdom of Matthew 19:30 began to resonate with me a few weeks ago when I was considering how the UK was willing to take the lead in some European matters, but was steadfastly behind everyone else in others. After the nation felt utterly slighted at the beginning of the 1970s when General de Gaulle said 'Non' to the idea of the UK joining the Common Market (as it was then), a great proportion of people now think that 'Grocer' Heath made a big mistake in ensuring that the General's successors said 'Oui'.

As the Common Market became the European Economic Community, and then the European Community, or the European Union, depending on the time of day, Britain embraced the idea of being part of Europe while not joining in fully. It has been like watching someone who goes to a bar after work when they are not really sure whether they should. They don't take their coat off, they keep refusing drinks because they say they have to go soon, but they still don't leave; they just hang around the doorway nursing the same glass all evening.

In some respects the European Broadcasting Union (EBU) was a technological precursor to the EC-EU. When it comes to this kind of European union, Britain has been very keen to be seen as a leader, while other members have been less certain about the issues involved. Okay, so the UK was not wild about PAL+, but it did forge ahead with MAC, and now is at the forefront of both digital television and digital radio.

Wanting to be seen as an innovator or enthusiastically embracing new ideas are both laudable, but there does seem to have been an unseemly rush in the UK wanting to start digital services. Perhaps observers in Sweden feel the same way because that country has either mirrored the UK or been just a few paces behind. There always has to be a leader, but matters are not helped when that leader goes striding off down the street, leaving everybody else either struggling to keep up or getting lost along the way.

One of the problems caused by the hurry to establish these new technologies has been the lag between broadcasters, domestic equipment manufacturers and retailers. It was only in July that five manufacturers unveiled in-car digital radio receivers, outlining a July-August launch schedule, and giving details of pricing. Services have been running in the UK and Sweden since September 1995, with other countries introducing pilot schemes in the years since then.

During a press conference at the BBC, the one question nobody asked was 'Why has it taken so long for these receivers to appear?' I cornered a representative from Grundig who said that it was all down to money. This particular company has spent £100 million developing its new product, an amount it has written off. Despite this, its new receiver still costs £499; this is the starting point, with prices rising to £1,099 for other brands.

Prices have yet to be announced for digital TV set-top boxes, yet confusion has spread even before they have hit the market. The EBU and national regulatory bodies have forced broadcasters like BSkyB and BDB to guarantee that their approved receivers will give access to all digital signals, avoiding the situation of the benighted consumer not only having to buy two boxes, but to find somewhere to put them both.

This move was welcomed by the Digital TV Action Group (DigiTAG), a forum under the auspices of the EBU that provides information for broadcasters, manufacturers and retailers. The group's president, Herman van Wijk, says that its ideal is to create a unique European decoder-receiver, but he acknowledges that the already structured nature of the TV market will have to change to allow the take-up of digital TV. Van Wijk welcomes moves by the EBU, saying that unilateralism by broadcasters could isolate a technology from later harmonisation.

'We understand the drive... to be the first on the market and to create a fast take-up,' van Wijk says, 'but our aim is to make sure that there is no gate-keeper, no master of the set-top box, because that could make the spread of digital television impossible.' Perhaps this explains the British obsession to be first this time. They missed an opportunity with the EU and so sulk in the corner; this time they have a chance to make everyone play with their ball and nobody else's. Pas.

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M-S techniques

M-S techniques have a vital role in stereo audio systems. **John Watkinson** reveals that M-S can be used in microphones, signal processing, metering and transmission

THE M-S TECHNIQUE is a fully reversible way of mapping conventional L-R (Left-Right) stereo into an alternative spatial representation. Fig.1a shows that the two stereo signals, L and R are passed through a sum-and-difference unit that produces two signals, M and S. The M (or Mid) signal is the sum of L and R, whereas the S (or Side) signal is the difference between L and R. The sums and differences are divided by two to keep the levels correct.

M-S signals can be returned to L-R format by passing them through another sum-and-difference unit. However, keeping the signals in the M-S domain has advantages.

The result of this sum-difference process can be followed in Fig.1b in which it is assumed that the L-R input came from a pair of crossed eights. A new polar diagram is drawn that represents the sum of L and R for all angles of approach. It will be seen that this results in a forward-facing eight, as if a monophonic microphone had been used, hence the term M or Mid for the sum signal. If the same process is performed using L-R, the result is a sideways facing eight, hence the term S or Side. In L-R format the acceptance angle is defined as the angle between the nulls whereas in M-S format the acceptance angle is clearly between the points where the M and S polar diagrams cross.

In sound radio, the M-S technique is used in FM radio to produce a mono-compatible stereo broadcast. The L-R input is converted to M-S and the M signal is broadcast normally. The S signal is transmitted on an additional subcarrier that a monophonic receiver will reject. However, a stereo receiver will handle both M and S signals. The stereo decoder contains a sum-and-difference unit to return to L-R format for reproduction on conventional stereo speakers.

In audio production the apparent width of the stereo image may need to be adjusted, especially in television to obtain a good audio transition where there has been a change of shot, or to match the sound stage to the picture. High-quality stereo preamplifiers may contain a width control so that excessively wide stereo recordings can be restrained.

In M-S this can be done by manipulating the difference between the two channels; in other words subjecting the S signal to variable gain as shown in Fig.2a. Following this a second sum and difference unit is used to return to L, R format for monitoring. The S gain control effectively changes the size of the S polar diagram without affecting the M polar diagram. Fig.2b shows that reducing the S gain makes the acceptance angle wider, whereas increasing the S gain makes it smaller. Clearly if the S gain control is set to unity, there will be no change to the signals.

Setting the S gain to zero results in monophonic reproduction because L and R must be identical. Thus a width control can replace a stereo-mono switch with a variable control having the conventional stereo and mono settings at the extremes of travel.

On a practical note, it is often necessary to use a stereo microphone with a fishpole. In some shots the microphone will be above the action, but in a close-up it may also be used below shot. Inverting the microphone in this way will interchange left and right channels. This normally requires plugs to be exchanged or two pan pots to be turned. If an M-S microphone is used, the fishpole operator can operate a simple phase-reverse switch in the S channel that will reverse the channels. Using balanced signals a passive phase-reverse switch is possible and can easily be built into an XLR barrel connector. Mixers having an M-S width function can easily incorporate an S-invert function.

While M-S stereo can be obtained by using a conventional L, R microphone and a sum and difference network, it should be clear from Fig.1b that M-S signals can be obtained directly using a suitable microphone. In M-S microphones the S capsule is always an eight so that a central sound source will be in the response null and produce no S signal. A variety of responses (other than omni) can be used for the M capsule which naturally faces forward.

The M-S microphone technique has a number of advantages. The narrowing polar diagram at high frequencies due to diffraction is less of a problem because the most prominent sound source will often be in the centre of the stereo image and this is directly on the axis of the M capsule. An image width control can easily be built into an M-S microphone because the S signal is directly available.

A favourite mono microphone can be turned into an M-S microphone simply by >

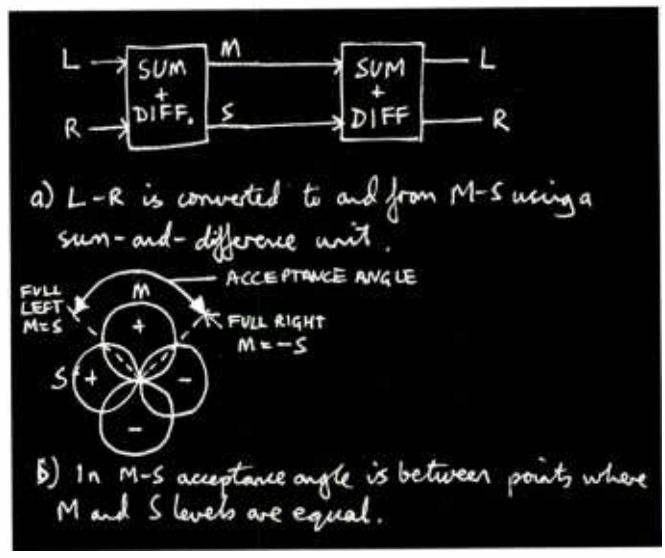

Fig.1

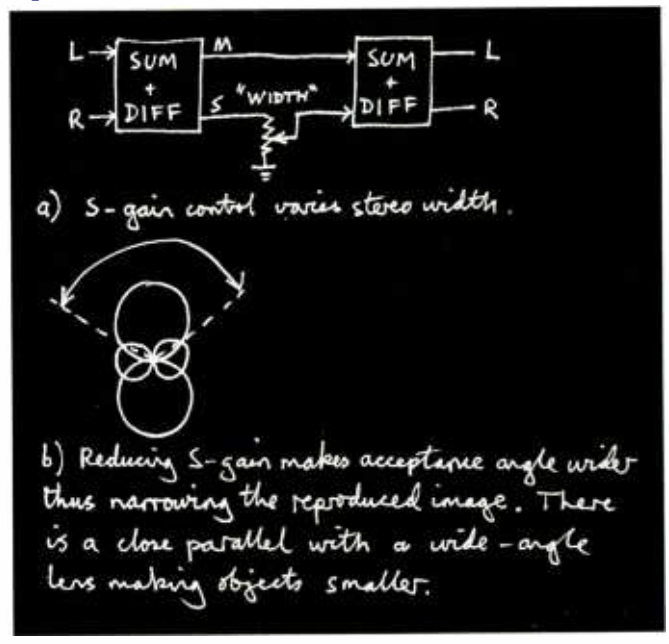

Fig.2

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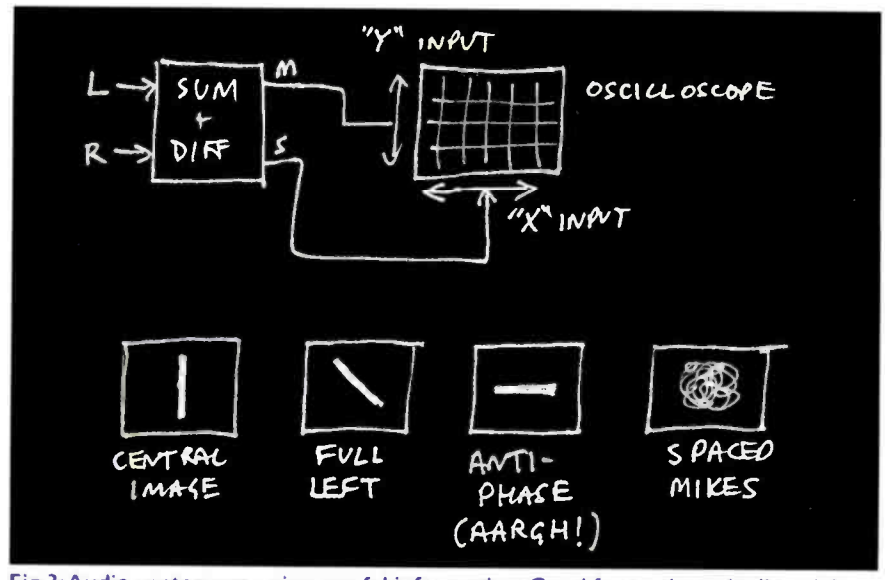

Fig.3: Audio vectorscope gives useful information. Good for testing mic directivity

< mounting a side-facing eight above it. This works surprisingly well provides a few basic precautions are observed. Mixing transduction types—moving coil and condenser can be problematical because a condenser mic measures displacement, whereas a dynamic mic measures velocity. It is possible to get unwanted phase shifts between the two signals when mixing mic types.

In stereo systems is it important that the left and right channels display the same gain after line up. It is also important that the left and right channels are not inadvertently exchanged, and that both channels have the same polarity. Often an indication of the width of the stereo image is useful. In some stereo equipment a twin PPM is fitted, having two needles which operate coaxially. One is painted red (L), and the other green (R), so that pilots have no difficulty remembering which is which. In stereo line-up tone, the Left channel may be interrupted briefly so that it can be distinguished from the right channel. The interruptions are so brief that the PPM reading is unaffected.

The broadcaster's nightmare is the inadvertent out of phase condition that causes a mono FM radio to get pretty quiet. Unfortunately the twin PPM gives no indication that the unacceptable condition exists. A better solution is the Twin-Twin PPM which is two coaxial PPMs, one showing L-R and one showing M-S. When lining up for identical channel gain obtaining an S null is more accurate. Some meters incorporate an S gain boost switch so that a deeper null can be displayed for line-up purposes.

When there is little stereo width, the M reading will exceed the S reading. Equal M and S readings indicate a strong source at one side of the sound stage. When an antiphase condition is met, the S level will exceed the M level. The M needle is usually White, and the S needle is Yellow. This is not very helpful under dim incandescent lighting that makes both appear yellow. Exasperated users sometimes dismantle the meter and put black stripes on the S needle. In modern equipment the moving-coil meter is giving way to the bar-graph meter which is easier to read.

The audio vectorscope is a useful tool that gives a lot of spatial information; although it is less useful for level measurements. If an oscilloscope is connected in X, Y mode, so that the M signal causes vertical beam deflection an the S signal causes lateral deflection, Fig.3 shows that the result will be a trace that literally points to the dominant sound sources in the stereo image.

Visual estimation of the width of the stereo image is possible. An out of phase condition causes the trace to become horizontal. Nonimaging stereo from, for example, spaced microphones causes the trace to miss the origin because of phase differences between the channels. An unsuitable combination of microphones in a 'home made' M-S coincident mic rig will also be revealed by a vectorscope display that misses the origin. In fact this should be the first test applied to any coincident mic Directivity switches on mics can fail so the polar pattern is not what you think. The audio vectorscope finds this problem at once.

A conventional oscilloscope is not ideal as a vectorscope because it has linear deflection, whereas the amplitude of audio signals is logarithmic with perceived level. The result is that at low levels the trace collapses to a spot and constant gain tweaking is necessary.

A proper audio vectorscope will contain logarithmic amplifiers or even compressors so that the spatial attributes of the signals are displayed over a wide range of levels. These units may display on a CRT or even on a 2-dimensional array of LEDs. Units have also been seen that synthesise a video signal containing the vectorscope picture. This can then be keyed into the video signal of a convenient picture monitor. Some units also provide L-R and M-S bar graphs or virtual meters in the video. ■

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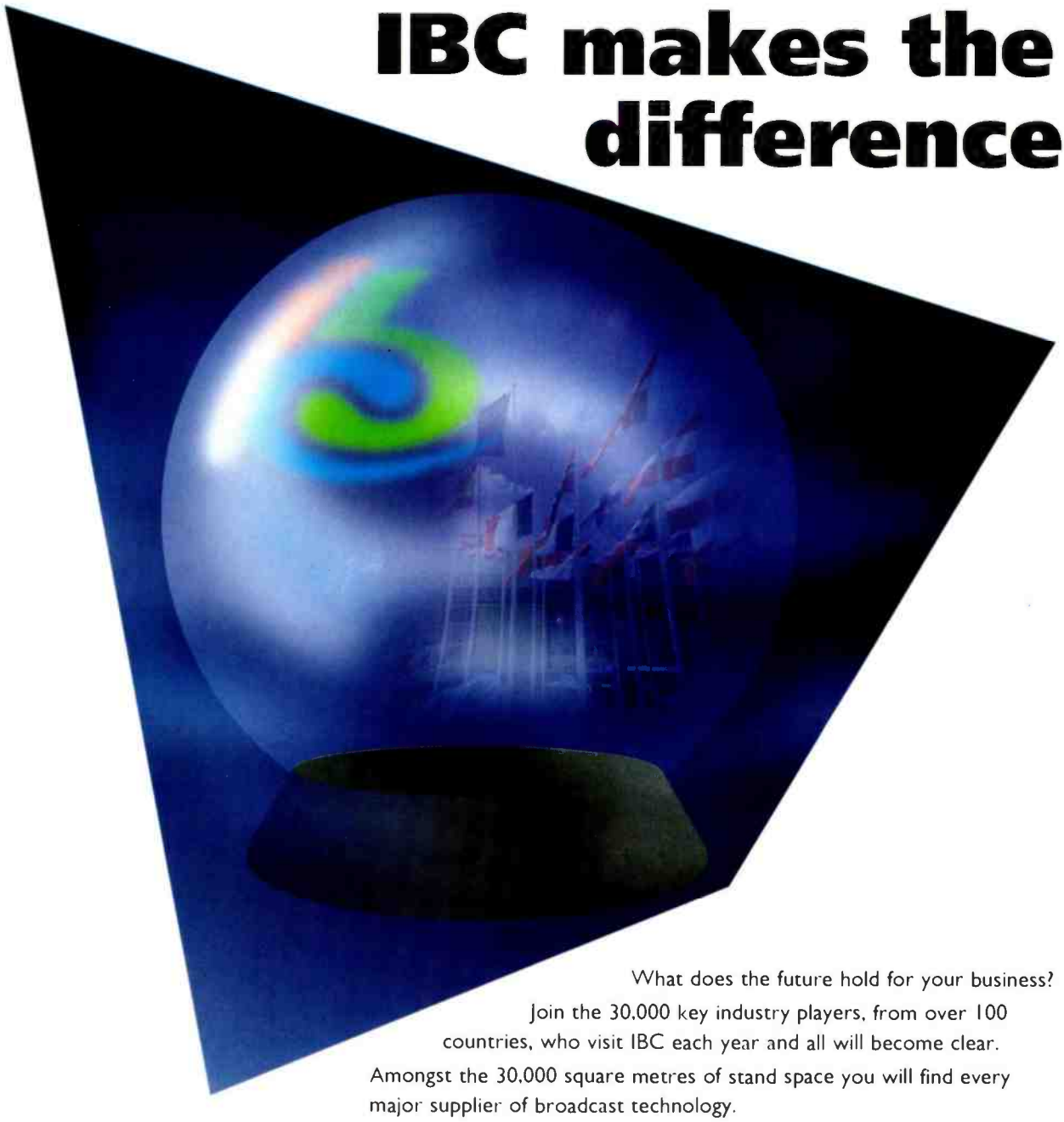

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Uncertainly yours...

Following the debate recently conducted in *Studio Sound* over the merits of 96kHz sampling, **Peter Bracke** of Barco's R&D department, discusses uncertainty and imaging

WE ALL READ with great interest John Watkinson's article ('Spotlight on 24-96', *Studio Sound*, February 1998) and the subsequent response. It is a useful debate because it forces the industry to search out why the 24-bit/96kHz demonstrations are superior to 24-bit/48kHz, and, as a consequence, what sound-impairing processes take place at 48kHz sampling, and what can be done to avoid it, and at what cost. As was mentioned, it is true that Heisenberg's Uncertainty Principle 'applies equally to frequency-time in electronic waveforms'. It is proven that for linear processing such as filters and transforms (Fourier, wavelet), the bandwidth-time product must be greater than one. This is not, however, the case for nonlinear processing such as the auditory system where there are parallel processed nonlinear filter banks in the cochlea and basilar membrane combination (with output to multiple input feedback). Human hearing achieves a better bandwidth-time product than is possible with linear processing, which is not 'a compromise' as quoted from 'Acoustics and Psychoacoustics'. This better bandwidth-time product poses a challenge when designing hardware or software perception models based on linear processing, and is one of the reasons why perceptual-coding algorithms, working with linear processing in the frequency domain, are not sufficiently transparent. Kates describes the inner ear as a filter bank with large bandwidths at the onset of sound for good time resolution, with two feedback mechanisms with different time constants that sharpen the filtering (enhance the Q) to the known steady-state frequency curves.[1] Heneghan uses 1-octave-wide filters in the filter bank to correctly model the masking in time, whereas 1-bark-wide (~0.25 octave) filters are used to model masking in the frequency domain.[2] Professor J Wouters from the Department of Brain and Behaviour Research (Catholic University of Leuven) encountered the same problem when designing the signal processing for a cochlear implant. Direct electrical stimulation of the auditory nerve supplants the function of the outer ear, middle ear and inner ear up to the cochlear nerve. It was Wouters who made me aware of this difference between linear processing and the auditory system, and stated that the bandwidth-time product for human hearing is 0.2 to 0.3. The uncertainty principle still applies, but it has a different value for human hearing than for linear transformation or processing, whereas the Heisenberg Uncertainty is a fixed law in quantum mechanics. So referring to the frequency-time trade-off as Heisenberg's Uncertainty Principle is not quite correct, if not confusing. It seems, therefore, that drawing conclusions from linear transforms and transform theory when it concerns psychoacoustics and human hearing, is a mistake, and that nonlinear mathematical theory should be applied. It is the dream of many researchers to create a psychoacoustic analyser that can evaluate sound processing and equipment without the need for listening tests with 'just noticeable differences for an average listener' as output. Since the mathematics behind nonlinear processing is not fully developed, and the required digital processing power would exclude real-time applications, some researchers are trying the known nonlinear analogue circuits with moderate success.

> It is proven that for linear processing such as filters and transforms, the bandwidth-time product must be greater than one. This is not, however, the case for nonlinear processing such as the auditory system where there are parallel processed nonlinear filter banks in the cochlea and basilar membrane combination

Getting the time resolution from the frequency bandwidth through linear transform theory, can underestimate the time resolution of the auditory system by a factor of 3 to 5. Does this mean that we need a 160kHz sampling rate for a 16kHz auditive bandwidth? Probably not, because the bandwidth-time product of 0.2 just applies the human speech range (to 3kHz). For higher frequencies the value rises, but still a lot of research is needed before we know these values and have a good nonlinear model for human hearing. Thus it is quite possible that energy above 20kHz is useful in human hearing to detect arrival time with greater precision; although it adds no useful pitch or level information to the perception of sound. (This violates linear transform theory, which does not apply to human hearing). The stated 16kHz bandwidth of human hearing applies for steady state signals.

What sampling rate is needed is not yet known from psychoacoustic research, and probably will not be known for a long time, due to a lack of funding since the availability of high capacity storage reduced the industrial interest for compression a few years ago. A 30kHz equivalent rectangular bandwidth (65kHz sampling rate) seems for the moment to be a fair number. It is certainly not necessary to have the same dynamic range for high frequencies as in the 1–4kHz band, so dither plus 4th order noise-shaping with a 5kHz corner frequency can be used to limit the amount of data, but it can only be applied once. This excludes its use in recording and the studio, and limits itself to mastering for a specific format.

The most frequently stated differences between 24-bit 96kHz and 24-bit/48kHz are a better image depth and an 'ease' in reproduction that becomes well noticed when going from 96kHz to 48kHz, and less so the other way around. Being able to do correct left-right positioning has no relation to image depth. I have never achieved an image depth with CD better than a third of the image depth with LP. From my many experiments with op-amps in audio amplifiers where similar differences can be found, I never could relate this to a standard specification such as the bandwidth (but this was always more than 20kHz). The differences between 24-48 and 24-96 could come from analogue circuits in the convertor such as the integrator in sigma-delta designs. Concerning the difference in 'ease' in reproduction, Pacific Microsonics' HDCD process also demonstrates this in comparison to standard CD.

Fig. 4 in 'The 96kHz debate: 2' (*Studio Sound*, June 1998) shows that there is ringing in the auditory system due to the parallel band-pass filter behaviour. This does not mean that the combined response of the basilar membrane has this (or any) amount of ringing, auditory research has not yet characterised this. It at least gives an indication that there is almost no pre-ringing in the auditory system compared to post-ringing. Digital reconstruction filters used with oversampling convertors are often based on sin(x) x schemes and have an equal >

< amount of pre- and post-ringing. From Fig.1d in 'The 96kHz debate: 2' you can observe that a 24kHz signal is generated before the impulse in an 48kHz system. The inner ear also generates a 12kHz subharmonic of this signal.

Is this the reason for the auditory differences? A listening test done in an audio workshop by the German magazine *Stereo* showed that every participant found the filter settings with the shortest symmetrical impulse response and with an analogue-filter-type impulse response without pre-ringing, superior, and the most natural sounding in the Sony CDP-XA50ES CD-player and T&A PreDa 3000 D-A convertor, both devices with multiple filter choices. Watkinson is right in stating that 'a rigorous implementation at conventional sampling rates may be all that is needed'; although the original sampling frequencies should have preferably been a bit higher than 44.1kHz.

Digital phase-linear decimation filters with a short symmetrical impulse response can be designed with FIR filters. The short impulse response can be achieved with window functions similar to the Flat Top window. The trade-off for the better impulse response is a not so flat frequency response at conventional sampling rates: -3dB at 20kHz.

Digital phase-linear decimation filters without pre-ringing can be designed with IIR filters. The narrow transition band from -1dB at 20kHz to -110dB at 22kHz makes the order of the filter extremely large. To implement this filter an ultra-high-performance DSP is needed, and the necessary high precision of the coefficients, and the calculation demand floating-point DSP.

Sin(x)/x based FIR filters or FIR filters designed in the frequency domain with the Remez exchange algorithm do not have the wanted time domain response, but are very easy to design (ideal for digital designers), well-documented and efficient to implement. For a decimation filter less hardware is needed when doing the decimation in stages. The IIR filter is too expensive to be used in a multitrack studio, but it is viable for a 2-channel mastering processor, so recording at 24-bit/96kHz with traditional FIR filters and mastering to 24-bit/48kHz with the IIR filters is an option. The pre-ringing will be half as long as in a 48kHz system with the same FIR filters, but this may not be sufficient.

The falling cost of digital electronics should make the IIR filter a viable proposition within 5 to 10 years. There is a well-known and often-used method that makes the IIR filter easier and cheaper to implement —oversampling. At 96kHz sampling the filter transition band is from 20kHz to 48kHz, an important relaxation of the specification

Looking to the future, the falling cost of digital electronics should make the IIR filter a viable proposition within 5 to 10 years. There is a well-known and often-used method that makes the IIR filter easier and cheaper to implement—oversampling. Oversampling is using a higher sampling frequency than twice the bandwidth of the signal. At a 96 kHz sampling frequency the filter transition band is from 20kHz to 48kHz, which is an important relaxation of the specification compared to a 20kHz–22kHz transition band at a 44.1kHz sampling frequency. The resultant filter could still be more expensive than the traditional filters, and using 2x oversampling would make the window-type FIR filter a better option. Ultimately, though, it is the economics of realisation of unnoticeable pre-ringing, rather than the quest for a higher bandwidth itself that practically favours the use of a higher sampling rate. ■

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2.'Investigating the Nonlinear Dynamics of Cellular Motion in the Inner Ear Using the Short-Time Fourier and Continuous Wavelet Transforms'. *IEEE Trans. Sig. Proc.*, Vol. 42, No. 12, December 1994

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Leader of the pack

Set your purist principles aside for a moment and consider a world in which audio data compression is a given fact. **Catherine Oates**, director of Nicral, has already

ALLOW ME TO ASK you a question: hands up who would work with digital audio data compression out of choice?

Since the printed word is not a visual medium as such, I can only guess that not many hands are in sight. I should, of course, qualify my question by saying: 'On quality criteria only, who would use digital audio data compression out of choice?' And if the whole world was one level playing field (without any accountants on either team) we would surely all be dealing in linear audio every time. But...

Over the past few years, the benefits of digital audio data-reduction techniques, notably MPEG ISO Layer 2, as well as other proprietary solutions such as apt-X100, have been great. This is especially so when combined with ISDN technology which has completely changed the way that high quality audio is zapped around the planet. I have to confess to having been one of its greatest exponents, having started Nicral seven years ago with the express intention of bringing ISDN-based audio technology to the broadcast industry. Since then, the aims of the technology have been vindicated. Without doubt it has been a great contributor to the progress of modern audio. I cannot think of a broadcaster in UK, Europe or America who has not made a significant operational commitment to ISDN-based audio technology.

ISDN-based audio applications are now numerous. These include real-time audio contributions from outside broadcast venues, cost-effective station-to-station links, and a total revolution in the commercial production industry with voice-overs able to reach the world without moving the car out of the driveway. ISDN is also frequently used as a supplementary circuit for STL (Studio to Transmitter Links) as it is an excellent candidate for infrequently used, but high-quality reserve lines in case the main leased-line should fail. But ISDN is not the first and last word in digital audio communication.

Using codecs for audio links in some parts of the broadcasting chain is highly appropriate and beneficial, but I have never advocated that it is all things to all women —or men, for that matter—which is why, for example, we have never pushed the technology into the recording industry as a tool for audio transfer premastering. Demo mixes are one thing, but who knows what will show up on the master if you have squashed and lost some of the signal on the way? Codecs designed to work on ISDN lines have a maximum data rate of 384kbit/s (3 x ISDN2), and even at this maximum data rate you are losing parts of the signal; although it would be impossible to determine by the human ear. And the same goes for any data-reduction algorithm you can think of —none of them can escape the fact that you are reducing data, therefore you do not receive the whole original signal.

> The telecommunications industry is, I believe, capable of far more than we are currently getting from it. Withdrawal of analogue services, such as BT is implementing as we speak, will, I think, help in the long term, to open up the market. This, I hope, will result in increased numbers of serious players providing digital services to business

The telecommunications industry is, I believe, capable of far more than we are currently getting from it. Withdrawal of analogue services, such as BT is implementing in the UK as we speak, will, I think, help in the long term to open up the market (even if it doesn't feel like it right now for many broadcasters). This, I hope, will result in increased numbers of serious players providing digital services to business plus expand their portfolio of services on offer particularly in the area of fixed digital lines which has previously been dominated by analogue music circuits. Nicral has codec technology, now, that operates from 64kbit/s up to 2Mbit/s which means from ISDN all the way to E1 (2Mbit/s) circuits.

In audio communications, as in many other areas at present, the consumer is king, so shop around because there is more than one place to get the kind of services that you want. Sooner rather than later there will be a really viable alternative nation-wide service provider who can deliver digital services to you in the format you want and cost-effectively.

In the meantime, I, for one, have turned my telescope around. Just for a minute I have stopped thinking about how few bits can be used to send an audio signal, and started seeing the not too distant day when the cost of digital services are so much cheaper that compression will become almost irrelevant in some applications; although I do stress 'some'. Even the bean counters may be happy for a 2Mbit service to be used for a single high-quality 20-bit 20kHz totally linear audio signal—ideal for studio to transmitter links, for example, or for the recording industry to at last be able to electronically transmit audio for mastering purposes without negative side-effects. There is still no such thing as a free lunch, but wait by the table because lunch may be about to become so much cheaper. ■

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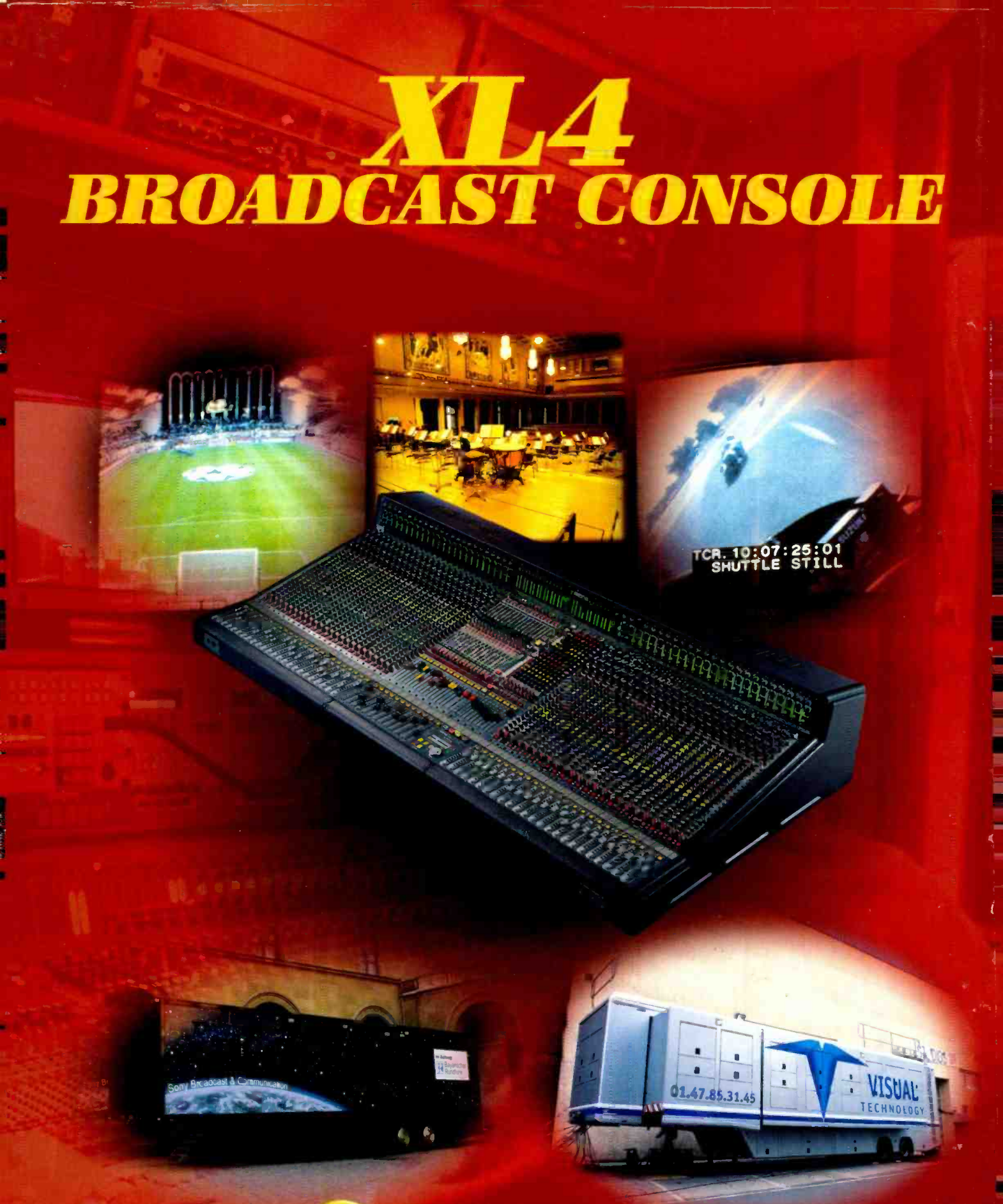

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