

July 2000 \$10.00 £5.00

Studio Sound

THE INTERNATIONAL PROFESSIONAL AUDIO MAGAZINE
FOR RECORDING, POSTPRODUCTION AND BROADCAST



EXCLUSIVES

Minipod
Amek Channel in a Box
Rosendahl Nanosyncs
Soundtracs DS-M
Yamaha H5000
Daking 52270
Daking 91579



SYSTEM 6000

**tc electronic's mainframe
multichannel processor**



The
**MARIO
CALDATO JR**
Interview

FIGHTING TO BE HEARD IN *GLADIATOR*
SURROUND SOUND IN ROOM DESIGN
BEST PRACTICE IN SOUND DESIGN
PLAYING THE FIELD IN EURO 2000



AMS Neve DFC



Buena Vista Sound, Hollywood



Skywalker Sound, Hollywood



Todd-AO, Hollywood



Berliner Synchron, Germany



Les Auditoriums de Boulogne



Warner Brothers, Burbank



International Recording, Italy

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Standard
in Film Mixing



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The Academy of Motion Picture Arts and Sciences has recognised the AMS Neve DFC as the first fully digital audio mixing console specifically designed for post-production film mixing.

The Scientific and Engineering Award for 1999 – presented to Mark Crabtree, Huw Gwilym and Karl Lynch of AMS Neve plc for the design of the DFC.

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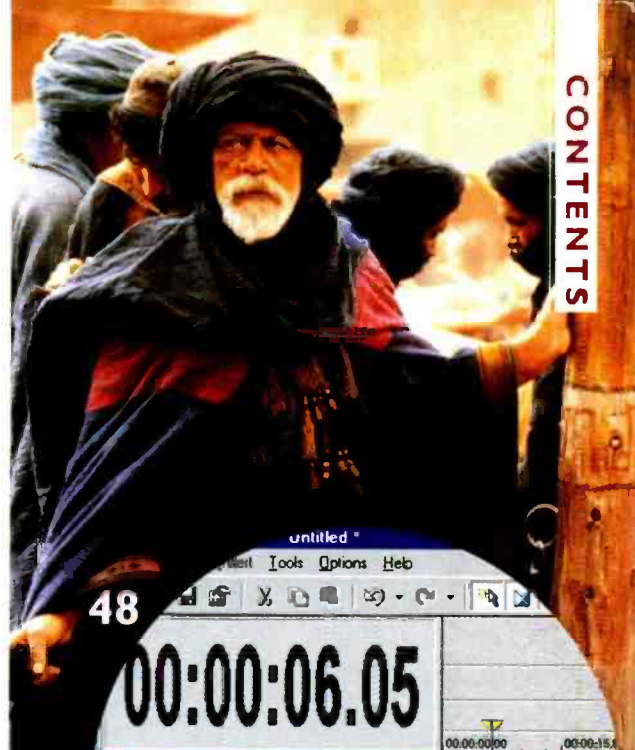
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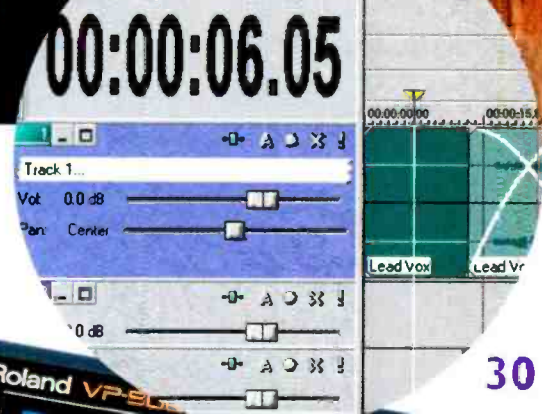
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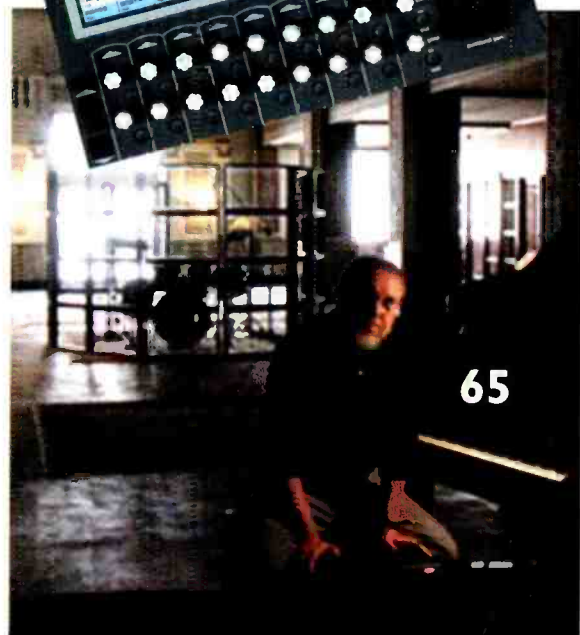
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Part of the machine

THE RAPID PROLIFERATION of the I love You virus served to remind us of the flaws associated with wide-scale standardisation on a product. It also served to highlight the potential danger of building in a little too much free flowing and easily accessible automation into a system that can be interfered with. The standardisation issue is an important one as it has analogies to the audio industry. A product becomes important as a direct result of changing needs, market forces or maybe just because it's a bit good. Users go for it big time and before long the particular sector that our product serves starts to look a tad dominated. The love virus' progress was all the more alarming because it radiated from one software product within a much larger whole, but still managed to cripple the total machine.

Standardisation on and domination by a product can provide enormous benefits for all parties involved, but it can be worrying because it takes power away from the buyer once the dependency is established. From this point in the manufacturer calls the shots not just by dictating the pace of development but also by deciding the point at which it stops and the replacement is ushered in. Too often there is a conflict of opinion on user and manufacturer interpretation of precisely when this point is reached.

Audio is littered with past standards that rose to prominence at a speed only bettered by the speed at which they fell from it. Support then becomes a major concern for users who make a stand on continuing with it and digital, in all its many and varied colours, has taught us the full scale of such a scenario. Contrast the continued support, appreciation and demand for Ampex' 2-track mastering machines (covered in our Masterclass in this issue) to that for the one-time people's favourite F1.

The lesson must be that when you buy into any digital systems orientated approach, no matter how affordable or how seemingly innocuous, buy with its inevitable, but not necessarily immediate obsolescence in mind and continually reappraise this impact upon your business. Those who don't, risk getting burnt.

Zenon Schoepe, executive editor

Hot source

NOW THEY'RE URBAN myths, once they were called memorates—both describe the integration of social anxiety with folk stories. Once there were witches then there were aliens, and now there are copyright bogeymen. All of them are memorates explaining the abduction of your children; in terms you can understand...

Conveniently, then, an American 'open source' programmer quoted recently in *The Wall Street Journal Europe* said: 'that in the future, copyright will be viewed the same way witch burning is today'. Article and comment followed the filing of a lawsuit by rock band Metallica naming in excess of 300,000 Napster users as copyright violators—apparently the only such suit currently filed by the authors of some of the pirated material.

Sadly, the crusade being pursued by Napster's acolytes (busy downloading unauthorised MP3 music files) threatens to compromise the real open-source movement as pioneered by the programmers behind the Linux operating system. For where Linus Torvalds sought to challenge Microsoft's stranglehold on the PC by publishing its source code and inviting all comers to contribute to its development, many in the Napster movement see copyright laws as an equally legitimate target. Yet where Torvalds' camp chose to waive the legal protection of the rights to their work, Napster offers simply to deprive musicians and songwriters of theirs. Perl language developer Larry Wall says: 'Open source should be about giving things away voluntarily', indicating growing concern that copyright piracy presents a threat to the more noble open-source cause. The controversy has already spread to the Debian version of Linux capable of showing DVD movies on a PC and being viewed by the film industry as a similar threat to Napster. The only significant dissenter appears to be Free Software Foundation Founder, Richard Stallman, whose argument reduces recorded music to the role of a lost leader for live events.

With this background, it is little wonder that the Big Five record companies are eager to implement an audio watermarking system with scant regard to its likely effect on the efforts of musicians and engineers. But their inability to distinguish between protecting the work of their artists and simple capitalism may yet prove a bigger threat to music recording.

Tim Goodyer, editor

Studio Sound

Incorporating Broadcast Engineering
July 2000. Vol 42. No 7. ISSN 0144 5944

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Roland Guillotel

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Studio Guillaume Tell, 20 avenue de la Belle Gabrielle,
92150 Suresnes, Paris Phone: +33 (0)1 42 04 05 05



SL 9000

Solid State Logic

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■ Cairo-based ERTU (Egyptian Radio and Television Union) has signed two contracts for the renewal of seven studios incorporating seven Studer D950 Digital Mixing Systems. One of these will be used for music recording, two for radio drama production and four for on-air studios. The move demonstrates ERTU's move to digital. The first contract is for Studio 46, a digital music recording studio centred on a 184-input, 48-fader D950S and Studer's D827 MkII 48-track DASH recorder. The second contract covers six studios including radio drama studios 37 and 38 and 24-fader D950B broadcast consoles for on-air studios.

Studer, Switzerland.

Tel: +41 1 870 75 11.

■ New York's NBC Broadcast and Network Operations has ordered a 56-channel Calrec Alpha 100 digital audio console for studio 3A at NBC Headquarters in Rockefeller Centre. This news control room is the first stage of the digitisation of Network News facilities on the site and the order forms part of a wider co-operation agreement between NBC and Calrec for the development of digital audio production consoles for live broadcast operations.

NBC, US. Tel: +1 212 664 5446.

Calrec Audio, UK.

Tel: +44 1422 842159.

■ London's Nomis recording studios has opened a hard-disk recording and editing room equipped with Digidesign Pro Tools, 24-bit Mix Plus system, running on an Apple Macintosh G4 450. The room also offers Logic Audio software, a Mackie console, HNB's CDR-850 CD recorder, an Otari MTR90II tape machine and ATC and Yamaha monitors. The studio has its own live room, plus upgraded soundproofing and air conditioning; the first session for the new studio saw producer-engineer Pete Smith working on Randy Crawford's new album.

Nomis Studios, UK.

Tel: +44 20 7602 6351.

HNB, UK. Tel: +44 20 8962 5000.

■ Fort Worth-based KTVT, a CBS owned and operated television station, has taken a 52-channel Euphonix System 5 console for its Live Audio Control Room where it will keep the company of a Grass Valley Series 7000 video switcher and Abacus DVE, linking to a VVG-Tektronix Profile Server. The new system is being used to handle audio mixing for daytime and evening including a half-hour newscast that airs at 10:00pm. All audio and video elements are replayed from a random-access Central Server. Seattle-based ABC-affiliate KOMO-TV has also taken a 88-channel System 5 for a new digital TV production and broadcasting facility. Owner Fisher Broadcasting operates 13 TV and 26 radio stations throughout the Northwest. California and Georgia. KOMO and KTVT-DT are among approximately 100 stations throughout North America broadcasting 5.1 channel Dolby Digital surround sound.

KTVT-HD/CBS-11, US.

Tel: +1 817 451 1111.

KOMO-TV Channel 4, Seattle, US.

Tel: +1 206 443 4199.

Euphonix, US. Tel: +1 650 855 0400.

■ Jakarta-based broadcaster Radio Sonora has installed four CEDAR

Series X and Series X+ processors in its efforts to restore its library of more than 7,000 LPs as well as cassettes and reel-to-reel tapes. The recordings suffer from clicks, crackle, and hiss problems and are being processed using CEDAR's declicker, decracker, dehisser and azimuth corrector. The entire catalogue will eventually be transferred to CD.

Radio Sonora, Java.

Tel: +62 21 633 7783.

CEDAR Audio, UK.

Tel: +44 1223 414117.

■ London's Molinaire post facility is to install two Soundtracs DPC-II digital consoles where they will work alongside AMS Neve AudioFiles in new rooms designed for DVD, Film, TV drama and documentary production.

Molinaire, UK.

Tel: +44 20 7478 7000.

Soundtracs, UK.

Tel: +44 20 8388 5000.

■ Recent rental acquisitions include GearBox Hire's Lexicon 960L and tc electronic M6000 surround processors, an Apogee AD8000 AD-DA converter, 2 Pro Tools v5 MIXPLUS systems and Pro Control, Genelec 1029A, 1030A, 1031A and 1032A monitors, Akai DD8plus digital dubbers, Universal Audio 1176LN & LA2A reissues, Teletronix LA2A, a rack of 24 original Neve 1073 mic-line parametric EQs and the last new Focusrite ISA215 in existence. Illinois-based dB Sound has purchased a 70-input ATI Paragon II console. The desk will provide monitoring on Savage Garden's world tour. The Australian-bred band joins Ricky Martin, Tina Turner, Sting and Bob in using the Paragon. Back in Britain, Lancashire's Northern Stage Services, has added a further six Sony Freedom wireless mic systems to its stock. The system includes a MB-806 6-way receiver and six WRT-805HR belt-pack transmitters incorporated into custom flightcases for rapid turnaround between jobs. Villa Audio and Ultrasound, meanwhile, have taken XTA's DP266, the latter for main control processing of its new L'Acoustics rig.

Net: www.gearbox.com/hire.html

■ Dublin's Totally Wired Recording has installed a Neve VR60 console with Flying Faders/Total Recall. The studio is based within The Factory media complex that incorporates independent record companies, Internet designers, PA-light hire and rehearsal facilities. The console is the second VR in Ireland replacing a Neve 8108. TWR has also opened a second studio-programming suite based around a Soundcraft DC console, 24-track Tascam DA-88, Logic Audio Platinum and Neumann and AKG microphones.

Totally Wired Recording, Ireland.

Email: twired@indigo.ie

■ US: New York City's MTV Networks has purchased an AMS Neve Libra Live Series II digital broadcast console for its Studio B control room. The console, the sixth sold to date, will be used for Total Request Live, sundry live music productions and the 2000 New Years Eve special.

MTV, US. Tel: +1 212 258 8300.

AMS Neve, New York.

Tel: +1 212 965 1400.

Northern Stage Services, UK.

Tel: +44 1706 849469.



▲ **Australia:** The Australian Broadcasting Corporation has ordered the first DSP Postation II for its facility in Perth. The DSP system will be complemented by JBL LSR Studio monitors, Crown Reference II power amplifiers and DK Audio Master Surround Displays. As Australia's largest and only national, non-commercial broadcaster, ABC-TV will use the Postation II predominantly for posting documentaries, promos and television series broadcast to 8.9m people in the five metropolitan cities, and 4.2m people in rural areas.

Technologies merge

Europe: The European DVD Association is to merge with the DVDA-Europe. Mark Windy of the EDA said, 'I believe the timing for this initiative is perfect and will build nicely on the work done by the EDA and Audiovisual Eureka.' Sue Cook Executive Director of the DVDA-Europe added, 'Collaboration and co-operation are vital to grow our industry effectively and our merger represents a stronger voice for the European industry.'

The two associations will bring important issues to key players and combine the AV side of the industry (including Cinema and TV) with the Corporate sector and web DVD. Most of the EDA Executive Committee will be joining the DVDA-Europe Board to continue building on the subtleties of the European market and ensure issues such as copyright and classification and asset sharing

remain a focus within the DVDA-Europe. Chapters of DVDA-Europe are currently being set up in the UK, Norway, Czech republic, Ireland, Italy and Belgium.

The European DVD Association was started in 1998 by Audiovisual Eureka to help to build a strong European DVD industry. An Executive Committee of DVD professionals assures the strategic decisions and carries out the day-to-day co-ordination of the association's key actions. Audiovisual Eureka is an intergovernmental organisation consisting of the 34 European Governments, that acts as a bridge between the EU and the outside countries in the media sector. DVDA-Europe was started in January 2000 as a wholly independent organisation linked to the DVDA-North America and DVDA-Asia. The DVDA-Europe has over 1,500 members world wide and is a non-profit, impartial organisation dedicated to supporting the growth of the DVD industry.



▲ **US:** Loud Recording has installed a Sony Oxford digital console, the second to arrive in Nashville following an installation at Ocean Way. The desk has seen action extending the studio's country music repertoire with Billy Gilman's new project being coproduced by David Malloy and Don Cook.



▲ Denmark: Sun Studio of Copenhagen has installed a 96-channel Avant digital film console in its new 60-seat viewing theatre. The room is fully THX approved and is fully equipped for Dolby SRD, DTS and Dolby EX, plus 8-channel SDDS. With facilities in Norway, Sweden and The Netherlands, Sun has a long history in feature film foreign language dubbing for Hollywood majors including Disney, Warner Brothers, DreamWorks, 20th Century Fox, and Columbia TriStar. A considerable amount of work is also provided from Scandinavian TV stations.

Media interchange format

US: The SMPTE has initiated a search for suitable software to support a Browsing Interchange Format (BIF) for video, audio and data. The Request For Technology (RFT) is an attempt to assist library and similar storage searches. Suitable schemes should 'permit rapid and efficient identification of material to be incorporated into productions or to be reviewed, reused and repurposed. To minimise the required

storage capacity and network bandwidth, the browsing signals must be highly compressed and scalable in bit rate'.

'The results of the RTF process will be new standards that allow integration of library systems and video, audio and data servers into television facilities ranging from newsrooms to feature film production operations,' said Juergen Heitmann, chairman of the RTF committee. The RTF document can be found at:

Net: www.smpete.org/engr/rftbif.pdf

An easy guide

World: *The NLE Buyers Guide* (ISBN 1-901950-04-2) is a new independent publication dedicated to nonlinear video editing systems, as well as disk recorders-servers aimed at editing applications. It is designed to help the potential purchaser to establish what is on the market so that they can begin the process of selecting the most suitable product(s) for their particular needs. With over 300 entries, this first edition aims to include not only every existing system but those soon to be launched, and, for completeness, also shows those products which are no longer in production.

Full listings include target markets, hardware and software specifications, operational features, interfacing with external devices, file sharing, archiving and mastering, future development plans, training and customer support, typical configurations and costs, and suppliers details for North America, Europe and Asia-Pacific.

Those familiar with SYPHA



▲ Austria: The spectacular Vienna Festival drew on 10 Chameleon 3500 DP2 amplifiers, BSS Omidrive 355 crossovers and Adamson Y-Axis speakers to deliver 25,000W of sound to an audience some 40,000 strong. Highlighting the combined talents of the Vienna Philharmonic Orchestra and the Arnold Schoenberg Choir, the Festival was mixed with two Midas XL 3/40 consoles and a Cadac J-type console for the solo mics. Outboard EQ included an Apogee CRQ 12 parametric and Apogee Correq. The programme included works from Mozart, Mahler and Schoenberg conducted by Zubin Mehta.

Email: sypha@compuserve.com
NAB Services, US.

Tel: +1 800 368 5644.

Net: www.nab.org/nabstore.

Birthday by design

Germany: Having spanned projects as diverse as recording studios, multimedia museums and conference rooms for architect Frank Gehry, acoustic specialist ACM is celebrating its 10th anniversary. Founders Don Davis, Jochen Veith and Michel Schreiber set out to design high-quality studio environments in 1989 and have since grown to a 12-strong team and finished 200 projects. The spectrum of the Munich-based company spans from optimising existing locations, to the planning and realisation of interior designs, to consulting on entire projects. They can also provide complete interior decoration, air-conditioning installations, sound isolation and architectural acoustics. Satisfied clients include the SZM Broadcasting Centre in Munich, German television stations Pro 7, Kabel 1, DSF and N24, public TV and radio studios, Beta Technik (Kirch Group), Buena Vista International, the Bavaria Studios, UFA, MMC, Studio Babelsberg and the college of music in Detmold.

Email: get-in-touch@t-online.de



▲ UK: The third Studio Sound Audio Industry Recognition Awards (SSAIRAs) saw a select group of audio manufacturers gather in London recently to accept accolades and sip pomagne.

An informal gathering saw representatives of console manufacturers rubbing noses with outboard manufacturers and software houses to celebrate their collective popularity with the Studio Sound's readership, whose votes decided these awards. The awards were presented by Studio Sound publisher Steve Haysom; editor Tim Goodyer, associate publisher Joe Hosken; and executive editor Zenon Schoepe. Notable winners included Euphonix' System 5 digital console, TL Audio's haul of four outboard awards, CEDAR Audio's hat-trick and Focusrite's pair.

A complete list of winners was published in the May issue of Studio Sound. Our thanks and congratulations to all. A selection of photographs from the evening can be found on the Studio Sound web-site at: www.prostudio.com/studiosound



◀ Republic of Korea: Munhwa Broadcasting Corporation has chosen the Euphonix R-1 as a master multitrack recorder for live and live-to-disk programs produced at the network's Seoul broadcast studios. The choice reflects the network's heavy workload and the R1's analogue-operational interface. Pictured (L-R) are Shin Ki Ok (System Engineer, MBC), Kim Hyun Soo (Sound Director, MBC), Kim Young Kyu (Best Logic Sound) and Shon Mog Heon (Chief Engineer, MBC).

Hot 'n' cold rock 'n' roll

US: Rob Reiner's 1984 spoof documentary, *This is Spinal Tap*, has been remastered into 5.1 for forthcoming MGM DVD release. Starring actor-musicians Michael McKean, Christopher Guest and Harry Shearer. Reiner's 'Rockumentary' of the Tap's exploits have enjoyed some 16 years as a staple in midnight movie houses before being presented to Burbank-based Chace Productions to bring 5.1 to the loudest band in rock 'n' roll.

For the project, MGM delivered a 16-track, 2-inch stereo master that contained LCRS M&E and a mono dialogue stem which was remixed using proprietary technology in the Rick Chace Theatre.

McKean, Guest and Shearer are musicians and are aware of the mastering process and of technical aspects of mixing. McKean played the singer of Spinal Tap and much of the comedy in the film is embedded in his lyrics so it was critical that the lyrics still be intelligible. McKean and Shearer, along with guitarist Christopher Guest, wrote most of the material and were closely involved in the remix.

Chace is a full-service audio postproduction facility, restoration and remastering of film soundtracks including audio compression, NoNoise audio repair, music and effects construction and sync conversion transfer services.

In the can

UK: Canford Audio has assumed the rights to design and manufacture the EMO range of products. Based in County Durham, EMO's highly-regarded range of studio and performance products (including mic splitters, DI boxes, racklights, and mains distribution products) has found favour in the highest of circles. Chas Kennedy, Canford's Group Manufacturing Director explained, 'We have been distributors of EMO product for many years and know the range very well. It is manufactured to the same high standards as our own. It was obvious EMO would be a perfect fit, complementary to our own products.'

Canford is keen to stress that there are no plans to change the branding of EMO products. They will continue to be marketed as a brand in their own right using Canford distribution expertise.

AK4394 24 bit
192KHz stereo DAC
120dB dynamic range

A-5394 24 bit
192KHz stereo ADC
123dB dynamic range

Even if you think you are not converted, you probably are. Most music is either played through or made using AKM multi-bit technology. For over 15 years, AKM have been a leading force in the world of Delta Sigma ADC/DAC and Codecs. AKM's advanced mixed signal CMOS technology provides solutions for up to 192KHz for the latest digital audio systems. From mini disk to MP3, live performance to CD, TV centres to set-top boxes, karaoke to DVD, AKM are here and now.

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World Wide Audio, without leaving your studio.

Rocket Network takes audio production beyond the boundaries of studio walls, making connections that let you work with anyone, anywhere, anytime. It's like a global multi-track.

On-line Flexibility.

Rocket Network uses the Internet to allow professionals to work together on audio productions without having to be in the same physical space. Instead of shipping tapes from place to place or renting high-capacity phone lines, you log into your Internet Recording Studio, where Rocket Network handles the details of passing your parts to others and vice versa. That leaves you free to concentrate on capturing the perfect take, using your own local system to record and edit. Whenever you're ready for others to hear your audio or MIDI parts, you simply post your work to the Internet Recording Studio, automatically updating everyone else's session.



Full Audio Fidelity.

With Rocket Network, there's no compromise in audio quality—the system handles files in a vast range of formats and compression levels, all the way up to uncompressed 24 bit/96kHz. And you don't need access to a super-fast connection; DSL or T1 is great, but you can also work productively over a humble 28.8 dial-up. The system supports multiple user-defined presets for posting and receiving, and handles all conversions, letting everyone participate in their own preferred format. That means you can conduct a session in a speedy, low bit-rate "draft" mode, then move on while the final parts are posted in the background at full-fidelity.

Escape the boundaries of your studio walls.

Register at www.rocketnetwork.com

source code: RN22



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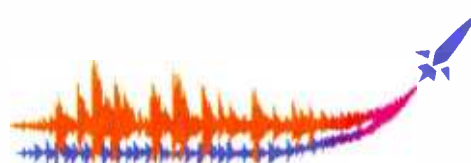
Professional Tools.

Through partnerships with leading audio developers, Rocket Network is bringing RocketPower™ to the professional tools you already use, starting with Steinberg Cubase VST and Emagic Logic Audio. A multi-level permission system lets you control access to your Internet Recording Studio. And our RocketControl™ client offers built-in chat capabilities, so everyone in the session can chime in with feedback as the project takes shape. The Rocket Network Web site offers additional resources and services for audio collaboration.



A Powerful Connection.

Rocket Network adds a new level of freedom to creative collaboration, allowing you to choose your team — singers, musicians, voice-talent, composers, engineers, producers — based on who's right for the project, wherever they happen to be. With full fidelity, plus anytime, anywhere productivity, Rocket Network is a powerful new connection to the world of audio production.



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Internet Recording Studios

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3-5

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& Conference Centre,
Glasgow, Scotland, UK.
Contact: Cate Gibberd,
The Radio Academy.
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6-8

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tc electronic System 6000

Endorsing the arrival of multichannel working, tc electronic's System 6000 combines everything from reverb to mastering and pitch shifting **Dave Foister climbs** on board

SURROUND IS FINALLY ON A ROLL. At last, after 30 years, we're past the stage of wondering whether the consumer will go for it—there's enough hardware out there now to allow us to move on to the next stage, working out what we're going to do with it. And the growing enthusiasm for surround music recording has highlighted what the film industry and the specialist surround fraternity have been saying for some time: there's a real need for multichannel signal processing that few are able to meet. For years people have been lashing together stereo processors to try to control four, five, six or more channels of audio, and only now are we seeing the first real tools to allow them to do their job properly.

So it was that I went to visit tc electronic's headquarters in Aarhus, Denmark, to find out about the background to the company's, much-vaunted flagship processor, the System 6000. Seeing a company at home is always fascinating, and the picture that emerged was of a young vibrant team, bursting with pride at what tc electronic does and how. Attention to detail is fundamental to the tc electronic culture, along with total involvement from all concerned.

tc electronic pioneered the idea of the open-architecture DSP platform with the M5000, and the expandability and flexibility the concept provided has meant that the product continues to grow. Indeed it would not be going too far to say that the System 6000 could perhaps have been introduced as a further development of the 5000, and in fact there is an upgrade path from the 5000 to full 6000 functionality, although it is by no means a simple job. The new system as supplied does all its work in a plain black box with nothing more on the front than a couple of switches, a PCMCIA card slot and a diskette drive, indicating again the open-ended nature of the design.

What goes on inside, in terms of the all-

important reverberation algorithms, is the ongoing result of major new investment in research and development. Virtual Space Simulation (VSS) was introduced on the M3000, and the 6000 sees its logical extension into full surround, VSS5.1. It could be said that tc electronic's approach is fundamentally based on physical modelling, but the designers are quick to dissociate themselves from the idea that creating real-sounding rooms ends with mimicking them in software. The initial steps in creating a surround room involve the use of Odeon, a powerful architectural acoustics tool that can predict reflection paths from a given position within a given space. Odeon can in fact generate 'auralisations'—simulated binaural impressions of what a space will sound like—and several of these are impressively demonstrated on the Odeon web-site. tc electronic uses the system to model early reflection patterns, assigning level, diffusion, colour and direction to each, and, of course, there can now be very large numbers of these thanks to the ever-increasing power of DSP. The system is so modular that the direction rendering is carried out in a separate module, allowing for various techniques to be used as required. The reverberant tails are handled separately, with the emphasis on total decorrelation, and smoothness where appropriate. The reflections and reverb are combined in a module that maps the directional output channels on to the required loudspeaker layout, downmixing where necessary, yet capable of delivering a full 7.1 set of discrete feeds.

All of this can be run in real time on tc electronic's pair of Silicon Graphics computers, networked around the building and allowing several teams to tweak things simultaneously. The main work is done in a listening room that can be set up for 5.1 music or cinema-style surround, and here the thousands of parameters that make up each preset

are carefully tuned to achieve the desired result. The perfect demonstration of tc electronic's approach is that alongside the SGI powerhouses, extensive use is made of an experienced opera singer on the team, who possesses the aural equivalent of a photographic memory. His job is to decide whether an intended simulation is working convincingly or not and help tweak it until it does. Once everybody is happy, the resulting preset, with its parameters carefully combined into a manageable user-set, is mapped into the 6000 and checked again.

So far the available selection is small but incredibly impressive. Listening to the early reflections alone on a dry source makes the loudspeakers and room walls disappear, placing you immediately in the intended space. Forget the old idea of a handful of reflections vaguely defining a character—there are so many here that they completely identify the space by themselves. Adding in the reverb tails completes the image, and the result is an enveloping sound stage with a sweet spot that covers most of the room.

Once it reaches the user this power is controlled by a very striking remote control unit called the Icon. Each Icon has its own control CPU in a 1U-high box, and connection to the mainframe is by standard Ethernet LAN technology. The obvious consequence of this is that mainframes can be sited in a central machine room and accessed from multiple control rooms within a facility, connected via existing network cabling. Since each 6000 mainframe contains four independent processing engines, this networking allows a single unit to be shared between rooms.

The Icon is striking partly because of its complete lack of buttons. It has two panels, a flat one with six motorised touch-sensitive faders, and a sloping one with a very high resolution touch screen. Touch-screens have





been a worry in the past, with unpredictable responses and a need for great care when selecting things on them, but this one is so good that you soon forget they're not real buttons. All the functions are on generous finger-sized buttons, with round ones for selecting menus and rectangular ones for selecting parameters, and the text and graphics are impressively clear and helpful. There's even a choice of colour schemes, including a very classy gold and platinum look, although the blue boot screen reveals the Windows NT roots of the system. If you need a bigger screen, for example so that several people can see what's going on, there's a monitor output on the Icon's CPU.

Getting around it is almost immediately intuitive, with a clever avoidance of deep layers of menus, and the only real complexity comes in the wealth of adjustable parameters in some of the algorithms. And there are a large number of algorithms: a good selection of M5000 presets is included, as well as new reverbs for music and post, and most importantly the multichannel algorithms. Even here there is a surprising range: although the thrust may appear to be the surround reverberation, never forget that it was t.c. electronic that brought us the Finalizer, and that all kinds of processing have new demands placed on them by the need to manipulate multiple channels simultaneously. Thus the System 6000 offers mastering dynamics, including 5.1 channel multi-band dynamic processing, making it perhaps the most flexible and powerful multichannel signal processor we've yet seen.

This flexibility extends to its interfacing capabilities. The basic system has eight channels of I-O built into its DSP card, presented as four AES-EBU pairs. Three further slots allow up to ten additional ins and outs, giving no less than 18 channels through the box. The system I had for review had three 2-channel analogue cards fitted, giving me 14 channels, and the patching flexibility this brings has huge potential. All the I-O is at 96kHz and 24 bits, although the internal processing is often at twice or even three times this sample rate, t.c. electronic claims that the jitter rejection and reclocking is so good that the system will improve most digital signals even without performing any processing on them.

Handling all this connectivity is done on a simple routing screen, showing all the physical inputs and outputs and the connections to all four processing engines, with the faders used to select sources and link them to destinations. Inputs can be connected to multiple engine inputs, and engine outputs mixed to physical outputs, allowing, for example, four aux sends to feed four independent stereo reverbs that all appear on one stereo return. At the other extreme, a surround-equipped console can feed multi-channel audio into a multichannel engine for full surround processing, and the 6000 can handle two of these setups simultaneously. Note that some of the full surround algorithms reduce the number of engines available—effectively using the DSP power of two engines—although their operation is controlled from one integrated set of controls.

Recalling presets into the engines from the library is straightforward, with factory presets arranged in several groups depending on application and association, such as music reverb, postproduction reverb, M5000 presets, dynamics, VSS5.1 and multichannel tools. Downloads from t.c. electronic's website can add to the complement (the same applies to software upgrades) and there's plenty of space for user-settings. Complete routing setups can be stored and recalled, and total system settings—known as Scenes—can also be saved, including all routing and the settings of all four engines.

If this sounds unmanageable, it isn't. The screen is so well laid out that even when running four simultaneous effects, it's easy to flip between them and tweak them, easier even than having four separate boxes to deal with. Adjusting the settings can be done in two ways, both using the motorised faders that are always in the right place to be grabbed and moved. One method assigns sets of parameters to the faders, and stepping through the sets gives quick access to any adjustment; the other allows any parameter on the screen to be touched and instantly adjusted by the sixth fader. A further refinement is a user-defined group of fader functions, allowing you to put your six most-used parameters on one strip for ready access. In fact, to my surprise, I found the system to be more encouraging of experiment and dabbling than most simple entry-level devices, so clear and friendly is the user-interface.

Beyond the reverbs are the processes >



< more familiar from the Finalizer, many available in multichannel versions. Full multichannel parametric EQ was not yet implemented on the review model; 5.1 multiband mastering dynamic processing is. Here a full chain of expansion, compression and brick-wall limiting is available, with a set of adjustments that makes you realise just how much thought had to go into making this kind of treatment work flexibly on surround material. The limiters work full-band across all the outputs, but the rest work on three definable frequency bands, just like the Finalizer, and can work completely independently on the six outputs—five mains and the LFE channel. The tricks come in how the five are interlinked, and here the magnitude of the step from straight stereo linking becomes apparent.

In stereo there's seldom any need to do anything other than make sure the two channels always track each other's gain adjustments, otherwise the image wanders from side to side as the gains get out of whack. In surround it's not quite so straightforward. True, front left and right will probably always need to operate as a pair, but will a true discrete centre necessarily want to go up and down with them? Will front information need to be turned down when an audience applauds loudly at the rear? The more you consider it, the more scenarios come to mind, and the more apparent it is that great flexibility is required. The 6000 achieves this by having not one common side-chain arrangement but three, so that two pairs of channels can be linked while the other remains independent. It goes much further still, allowing individual channels to have more or less influence on the others; allowing an

external input to control various channels; and introducing a mode where one channel will only control a second when its level exceeds that in the second. The power is enormous, and to simplify things there is a selection of standard operating modes, of which the one marked Normal controls all channels, with front left and right linked, surround left and right linked, and centre independent. The LFE channel is always controlled separately if processing is required at all, and has its own side-chain and adjustable parameters.

For this type of setup there is a standard way of configuring the inputs and outputs, and a monitor matrix in the 5.1 toolbox helps keep track of what's where. This takes up a whole processing engine, but during initial familiarisation it's well worth it. Here individual inputs and outputs can be both soloed and muted, making it very easy to check that the system is doing what you think it should be. If you don't use this, the available DSP allows the simultaneous and independent use of both VSS5.1 reverb and the 5-channel dynamics package, even on independent sets of input signals given enough I-O cards. The absence for the present of the EQ is a disappointment as it's such a central part of what the Finalizer can do, but still for multichannel dynamics the 6000 can do things I don't believe you could do any other way.

There is, of course, more: there's VSS SR, an algorithm that generates surround reverb in a format ready to be decoded by Dolby surround decoders, with independent control of front and rear decay times and delays but a unified surround effect; there's MD-3, a 2-channel multiband dynamics package that >

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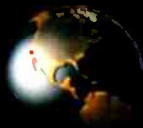
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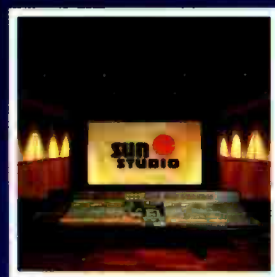
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< combines the MD-2 stuff from the M5000 with the same unit's Toolbox elements, with increased resolution, 96kHz capability and extra features; chorus, phase, flange and delay algorithms; Parametric EQ, with eight linkable channels of 4-band EQ; and VP-5.1, a true 6-channel pitch-shifter capable of shifting an entire stem in total sync. Some of these are still in the pipeline, and some require the purchase of additional licenses, although the asking price for these is surprisingly reasonable; as a guideline, the current UK price for a full-blown system including the MD5.1 mastering dynamics (which in turn includes MD-3) is not much more than £8,000.

It's clearly no coincidence that the world's leading players in the field of digital effects have produced major new multichannel processors within such a short space of time. In so many senses the time is right: the industry's need for such devices has grown from a specialist enthusiasm to

a real worldwide market, and the technology to do this kind of thing properly is now available at real-world prices. Everything that has put tc electronic where it is today is present in the System 6000; the quality and persuasiveness of its reverb algorithms have transferred spectacularly into the surround domain, and the thoroughness and power of its acclaimed mastering dynamics have been thoughtfully transposed into the realms of multichannel processing. The only frustration is knowing that there's so much more to come and not being able to play with it yet. ■

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Soundtracs DS-M

The reinvention of the Soundtracs brand continues apace with the launch of its DAW digital mixer postproduction combo. **Rob James** buys in

INTEGRATION IS THE NAME of the game. Anyone can play, but not everyone has the in-house expertise, time or money to start from scratch. In manufacturing, as in so many other things, it often pays to play to your strengths and as Soundtracs has already made inroads in sound-for-picture with the Virtua and DPC-II digital consoles it's no great surprise that the DS-M, is aimed directly at advertising and television postproduction.

The DS-M is based on the DS-3 console. The control surface is offered in a single 64-channel 32-bus configuration with the options of up to 112 physical inputs, 24 or 32 tracks of hard disk and optional nonlinear video. There is also DSP expansion to up to 96 channels and a 96kHz version.

The processing is housed in the surface



and console leg. I/O racks and the editor hardware can be remote sited. Audio connections from rack to the console are via MADI so suitably equipped machines or a router may be connected direct.

Console processing is courtesy of SHARCs in an essentially fixed architecture which is claimed to simplify life for the user and to promote greater reliability. The underlying operating system is Windows 98, although the operator will not be conscious of this during normal operation. Control surface model is a mixture of layered and assignable with three touch-screens and a slightly larger display-only screen mainly used for the editor. Opinion tends to be sharply divided on the merits of touch-screens. DAR, great champion of the technology, acknowledge this by offering the Storm DAW with a choice of controllers. Soundtracs has striven to please everybody by arranging things so that it is possible to drive the DS-M with a minimum of touch screen input or a good deal, whichever is preferred. The touch sensing is mechanical which is arguably more positive

than capacitive or similar types, but requires more than a gentle stroke. After years of dire warnings about shortening the life of LCD screens by touching them it takes the uninitiated a while to get used to giving the screen a healthy prod.

Two identical input banks of eight control strips occupy the left-hand side of the control surface. The centre LCD screen displays audio stream tracksheet and other edit screens. Dedicated editing controls fall close to hand above the inset QWERTY keyboard. A clearly delineated area, white, raised and profiled contains trackball, buttons and transport controls. Above are locator, global automation control and monitoring control keys.

The right-hand bank of eight faders are normally masters. Alternatively, they can control inputs if simultaneous 'hands on' control of more than 16 channels is required. Dynamic automation is displayed as 'tracks' on the touch-screen.

Each input bank has its own set of equaliser controls with joystick panner beneath. There is direct key access to any of the six layers and 'visible' channels can be freely swapped with invisible ones on other layers. Although it is easy to keep track of what is happening on the surface layer there is currently no display of information from hidden layers. This obliges operators to have

a retentive memory or work logically. The encoder at the top of each strip deals with channel input gain, the pair at the bottom are for pans, aux sends and dynamics.

A channel strip controls a single physical input or a pair for stereo. Stereo strips accept AB or M+S signals with width and balance controls. The common occurrence of left right or M+S reversal is dealt with via a left, right swap button.

Channel strips are selected to assignable controls in a number of ways. Touching the on-screen strip anywhere in the EQ or dynamics areas to select it is always active. The console can also be set up such that touching a fader and or pressing a Solo key selects the strip. In default view, key parameters of each strip section are displayed. When a strip is selected the grey background changes to orange. Blocks are viewed in greater detail by touching. On the Input module, adjusting the input gain will temporarily pop up more detail. The INPUT ROUTING button expands all the input displays together.

Touching the EQ graph expands the view

to include all the settings. A second touch restores. All four bands are full range 20Hz-20kHz. The middle pair are peaking, $\pm 18\text{dB}$ cut and boost with Q from 0.1 to 20. The top and bottom bands default to High and Low shelf respectively. Fixed 12dB octave high-pass/low-pass or bell options are selected by the CURVE switch.

A horizontal yellow band indicates which aux sends or pans the bottom two rows of encoders will affect. Two scroll keys move the band up and down and scroll this section of screen to reveal otherwise hidden controls. Up to 20 aux buses may be defined. Stereo auxes have pan and send level control. All auxes may be independently switched on or off and pre or post. A single switch does double duty here. To switch pre or post the send gain must be at zero.

The dynamics block expands across the bottom of the screen. Screen controls correspond to the lower rows of encoders and keys. Gating and compression are available on all input channels. Default layout is simple compressor-gate, but four other arrangements give alternative processing order and keying. When self keyed 12dB octave HP and LP filters are available for de-essing and other frequency conscious effects.

Adjacent odd and even channel pairs of dynamics can be linked for stereo.

Panning to the Main and Group buses is stereo or surround depending on the console mode. Surround panning can be set with two rotaries or, more intuitively, with the joystick. Two further controls set divergence and sub send level where appropriate. Divergence control is a little unusual. Full anticlockwise allows normal left right panning, the centre position gives hard centre, but full clockwise gives left right panning with no signal sent to the centre bus. Apparently this feature was requested by some music balancers.

The Routing block displays the full routing module for the strip. Inserts, direct outs and bus routings are all set from here. The M.I. button is used here to display all the routing modules for the bank of eight strips. It is also used to speed up multiple assignments of inputs and to gang together any or all of the channels within an input module. The ganging allows multichannel stems to be controlled from a single strip with similar, but not necessarily identical processing. Ganging also affects EQ and so on. Any relative offsets are retained, permanently for faders, or until something hits the end stops for other functions. Libraries enable complete channel presets, EQ settings and dynamics settings to be stored and recalled.

Configuring the console is undertaken using the master screen and associated command buttons. The easiest way to completely reconfigure the console is to recall a Session file. These contain (among other things) a complete console setup. Thirty- ➤

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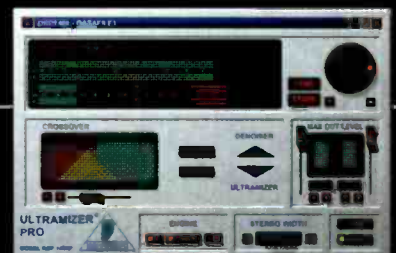


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< two buses are user-assignable and there is an extra, PFL bus.

If the Main bus mode is LCRS or 5.1 other buses may be in surround formats or mono or stereo, but if the Main bus is stereo the rest are restricted to stereo or mono. Each bus has Solo, Mute and a limiter facilities and no linking is possible. Buses also appear as input channel sources, allowing sub-grouping.

Control Groups are also available which behave like VCA groups.

Automation is well sorted. Enough complexity to keep the power hungry amused with initial simplicity to get you started with minimum pain. Channels may be dropped in and out of automation record by touch or by pressing local or global record keys. The snapshot (99 memories) and dynamic automation may be combined to increase versatility and to gain automation control of more parameters—with all Channels controls and Snap-

shots in Play mode any dynamically automated control is locked out when a snapshot is fired, but controls which cannot be dynamically automated (EQ curve, compressor attack...) change to reflect the snapshot. Dynamic automation has Isolate, Read, Write, Update and Trim modes with useful variations and manual or auto nulling. There is comprehensive offline editing using drag and drop, cut and paste or numeric parameters entry. MIDI program change and controller messages can be incorporated to automate external effects units. Monitoring facilities include a 32 x 6 matrix enabling complex monitoring configurations for formats up to 5.1. On-screen meters are supplemented by 30-segment LED bar-graph metering in the bridge, assignable between various input and output points. Apart from the internal synchronism between editor and mixer, machine control extends to time code and MTC generation and RS422 control.



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The Soundtracs implementation of the SADiE interface is proof positive of the minimalist mantra 'less is more'. The first thing to go was the SADiE mixer. This is rendered pretty much redundant by the console and



There is a perceptible time lag after PLAY is pressed before anything happens and a monitoring lag when punching into record. Neither of these delays is excessive.

The recorder-editor part of the DSM is just that. It is not intended to perform like a digital dubber. The removable disk-drive caddies are fitted to facilitate fast project changes. SADiE's Portia, which supplies the video, uses a separate drive. Video data can be transferred to an audio or PC drive to keep projects in one location, but will need to be moved back to a Portia drive for payout.

With complex projects involving a large number of audio streams the distribution of audio among drives can become an issue. As loading for audio playback increases, this is shown by a yellow indicator above the playlist. At points where this turns red, the system is likely to be unable to replay all the tracks. Making the list playable then requires user-intervention—moving clips to other drives.

Where material is prepared on another SADiE prior to moving to the DS-M, only edit,

fade and level information is currently used by the DS-M. SADiE EQ and other mixer based data is not used. By exploiting the auto-conform and file exchange capabilities it is perfectly possible to complete entire projects within the DS-M.

Integration is the DS-M's main *raison d'être*. This goes further than simply packaging a console, a workstation and a NLV in the same box. Doing this might superficially improve convenience, but would also completely miss the point. DS-M synchronisation between the editor, console and NLV is completely transparent, eliminating line-up, sample-rate, bit depth, clocking problems and machine control setup.

There is no 'rubber band' effect when jogging the system. This degree of lock is achieved by the use of several strategies and is tighter than most 'external' machines. Equally impressive is the ability to write automation data at any speed from stop to play or higher, forwards or backwards. I really missed reverse sync play. An odd omission given the same effect can be achieved with dextrous use of the jog wheel. Instead of requiring two keyboards and mice the single QWERTY and trackball move seamlessly between the two screens and activities. The Scrolling, horizontal zoom and vertical zoom of the audio stream and automation track displays are all aligned so you are always looking at approximately the same sources and channels over the same time period. Zooming and scrolling can be initiated from either screen and is instantly reflected in the other. Editor streams are associated with particular console channels, Tracks and faders are automatically given the same names so users don't have to remember if Track 5 is coming in AES 3 Left and so on. The console Solo function can be used to select clips to edit, the REC button can track-arm streams. Crucially, a single edit operation can affect and conform audio, video and automation data. Edits performed on the audio are instantly reflected in the automation. This allows clip-based EQ, ripple and slip in automation, transfer of elements from one channel to another, and quick copying of repeating elements. This is not restricted to EQ: all functions may be conformed by the editing. Audio, video, edit, automation and MIDI data may be wrapped up under a single project title. This can be stored, moved and archived via sneakernet or across a network.

Recently there have been signs that the revolutions in production processes brought about by workstations, fully automated consoles, nonlinear video and networking are beginning to coalesce into a cohesive whole.

DS-M represents a kick to get the ball rolling towards the ultimate goal. Other manufacturers have bits of the jigsaw.

While the DS-M doesn't complete the puzzle it moves the goal-posts and may well make the picture clearer. Meanwhile, it offers an opportunity to develop new and interesting ways of working. The downside is, once users grasp what might be possible, their wish lists are likely to grow ever longer... ■

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Spirit 324 Live

Taking a further step in the affordable digital mixing console stakes, the new Spirit swaps buses for functionality and targets the live fraternity. **Rob James** tries it for size



THE SPIRIT 328 offers a real alternative in the small, affordable digital console arena previously dominated by Yamaha's 0-series and their imitators. As such, the 324 Live brings the digital alternative to a new audience. Although there are very many obvious similarities between the two designs there are some significant and mostly useful differences.

This is a 32:4:2 and-or 32:4:1 design with additional matrix and floating analogue outputs which significantly increase its flexibility. The control model uses layering with 16 control strips 4 banks deep and assignable control over EQ and so on. Wordclock is internal only (with an output) which might seem like a limitation but when presented in this way, represents one less thing to get wrong. The SPDIF and AES-EBU inputs have sample-rate converters and the TDIF ports will be used for supplementary I-O or a digital multitrack which will happily sync to the

324. All the converters are 24-bit, 128x oversampling types. The quoted performance figures are adequate and sensible, more importantly the 324, like the 328, sounds good subjectively.

At the top of each of the 16 strips are analogue inputs and controls for bank A. An XLR mic input accepts balanced or unbalanced signals while the high-impedance line input is 1/4-inch jack for balanced or unbalanced sources. In familiar Spirit practice the insert jack is unbalanced, tip send, ring return. If the tip and ring are shorted this can also be used as an unbalanced send. The TRIM control ranges from +6dBu to -60dBu. A 100Hz 18dB octave high-pass filter will help deal with rumble. None of these controls' settings are stored with snapshots.

Below the input section are meters, 10-segment LED bar graphs, bank switchable like the faders, between Inputs 1-16. Inputs 17 to 32 or Master which shows Group, Aux and Effects master levels. A global DYNAMICS ON METERS key switches all the meters to display gain reduction. Just beneath each meter is an ALT PEAK LED that toggles with the bank switching. Usefully, this allows the operator to tell at a glance if any of the inputs hidden from view are going over the top.

Four keys above the 100mm motorised faders switch the EQ in or out.

The E Strip, first seen on the 328, runs horizontally across all 16 strips. The lower 'row' has 16 rotary encoders each with a concentric arc of indicator LEDs. The upper contains three keys to select what the surface faders, mutes and solos control. Bank A, Bank B or Masters. Pressing the currently illuminated key accesses the fourth bank for direct fader control of MIDI controller parameters. The next three keys bank switch the channel bar graphs. Two LEDs indicate whether the

encoders will affect Bank A or Bank B when no individual channel is selected. Eight keys switch between Level, Aux 1-4, FX 1, FX 2 and Pan. Encoder bank switching works with fader bank switching. When Level is selected, encoders control the level of the hidden input bank. With the remaining options, the controls affect the bank on the surface. This gives immediate access to, for example, all the pans.

Hitting a channel select key turns the E Strip into a single set of channel controls. The strip is then used with the select panel keys to adjust all parameters for the channel. From left to right the encoders control the three EQ bands, each with cut-boost, frequency and shape (Q). Frequency bands are restricted—the LF band covers 40Hz-800Hz, Mid, 200Hz-8kHz and HF 1kHz-20kHz. 15dB of cut or boost are available and the LF and HF bands can be set to peak or shelf types. The Q ranges from 0.3-2.8 and the remaining encoders set levels for Aux sends 1-4, the two FX sends and pan.

The Select panel works with the channel strips in an imaginative and intuitive manner. You can interrogate and change assignments in two ways. Looking forward from a single channel or looking backwards from the parameter to see the state of all channels. This is far easier to do than describe and feels completely natural.

Channels are routable to groups, and channels and groups to the main Mix and or Mono buses. Bank A has direct outs pre or post fader. So the first 16 inputs can be routed to the TDIF outputs, primarily for recording. Alternatively, the TDIF outputs take the four group outputs four times.

Additional to fader controlled channel inputs are two independent stereo inputs. STE-1 has an analogue input section. Input jacks are on the console surface with an >

Mic-line Interface

This 2U-high 19-inch rackmounting box contains an extra eight channels of mic-line analogue inputs and eight line level analogue outputs. Digital input is a single 25-pin sub-TDIF connection which is also used to for the wordclock source. Each of the input channels uses an XLR for mic level input and TRS 1/4-inch jacks for balanced line in and unbalanced insert. The outputs are jacks. On the front panel each of the eight channels has a GAIN pot with a range of -60dB to +6dB, switches and indicator LEDs for 48V phantom powering, Insert on-off and 100Hz 18dB/8ve HPF in-out. Two green and amber, and a red LED light at -30dB, -20dB, -10dB and -2dB respectively. Convertors are 24-bit 128x oversampling.

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< on-off switch and gain pot before the A-D converters. Alternatively STE-1, like the Digital Stereo input, can take its input from either the AES-EBU or SPDIF connectors.

Two rotary encoders underneath the output meters allow the level of both stereo inputs to be automated. An obvious use is external effect returns. The internal effects returns are similar. Each of the aux and effects sends is globally switched, pre or post fader.

The 324 comes with two TDIF ports which can be used for tape sends and returns or to connect one or more of Spirit's additional interfaces. I used an 8-way Mic-Line box to add a further 8 analogue inputs and outputs

Connectivity

Rear panel: Mains IEC, fuse and switch, binding post ground connection. 9-pin Sub-D RS-422 for linking to a PC or Mac for software updates. DINs for MIDI THRU, Out and In. A single BNC for wordclock out. Phono's cater for SPDIF in and out and XLRs for AES-EBU in and out. Two 25-pin Sub-D sockets allow access to Bank B inputs 17-24 and 25-32 and also either Group 1-4 outputs (repeated four times) or channel 1-16 direct outs. The surface: Each of the 16 channel strips have an XLR for mic level input a balanced 1/4-inch jack for line in and an unbalanced insert jack. AUX 1-4 out, Stereo Input 1, stereo monitor out and headphones are all jacks. Matrix 1-4 out, stereo mix out, Floating outputs 1&2 are XLR. A further XLR provides for the addition of a talkback mic.

(see sidebar).

There are no dedicated inserts on any of the digital inputs. The workaround is to use an aux out and return via a channel.

Four summing Matrix outputs can take any or all of the four groups, the Mix and/or Mono buses out. The level of the contribution from each source is variable. You can have a different mix on each matrix output for feeding different rooms or speaker clusters. Two further 'floating' outs can take the Mono bus, FX1 or FX2 or groups 1-4.

The two 16-segment meters display C/RM (Control Room) output level or dynamics gain reduction if the adjacent key is pressed. Solos can function as AFL, PFL, Solo and SIP (Solo In Place). Level controls are provided for the C/RM and Phones outputs. The talkback mic socket can be routed via a level pot to stereo pairs of auxes matrix outs the floating outs or mix output.

To deal with setup parameters and the multitude of effects possibilities Spirit has resorted to a small LCD screen with cursor, YES, NO and UNDO, REDO keys supplemented by eight dedicated keys for FX1 and FX2 and Dynamics Controllers 1 and 2, MIDI, Menu, Presets and Snapshots.

A lot of work has gone into making the snapshot automation as simple and transparent as possible. An essential feature for live work is the Isolate function which allows any channel to be isolated from snapshot recalls but remain active if new snapshots are written. There are 100 internal snapshot memories but these can be loaded or saved singly or as blocks to an external MIDI file.

The dedicated Cue Control Panel aids clarity. A red LED alphanumeric display indicates cue numbers, whether a cue has been edited since it was recalled, if there is MIDI data associated with it and if there is MIDI being input. Each cue can store 10 MIDI events for triggering a sampler or program changes on external effects units. The CUE SCROLL key switches the parameter encoder to scroll through the list.

Below the LED display the NEXT key recalls the next cue, LAST recalls the previous cue. Sensibly, LAST MUST be pressed and held before it acts. RECALL loads a selected cue out of sequence. HOME returns the display to the current cue number. STORE stores the current state to the currently displayed cue number, either the current cue or any cue selected with the parameter knob. A LOCK key allows selected functions to be locked out. There is no means of crossfading between snapshots.

F1 and F2 keys enable any of the LCD pages to be loaded by holding down the key and recalled instantly with a quick press. The 27 User Setups are similar to snapshots except they also store menu items (such as the sample rate). There are libraries of factory and user presets for EQ, dynamics and effects. Dynamic automation, if required, can be achieved by using an external MIDI sequencer.

Effects are from the Lexicon stable and part of the automation system—112 presets being augmented by 128 user-memories. Editing is comprehensive and reasonably straightforward via the LCD and cursor keys.

Dynamics are somewhat unconventional. Two sets of dynamics parameters are applied





to selected multiple channels in the console, but the controlling signal may only be derived from a single mono or stereo channel, an effects return or the main Mix.

There is a temptation to view this console simply as a 'cut-down' 328, but this would do it an injustice. The 324 appeals to me rather more than its sibling—as, instead of trying to do everything and just missing the mark, the 324 concentrates on doing what it does do,

well. The well thought snapshot automation should appeal to small theatres and conference venues as well as live music. Spirit have put a great deal of time and effort into making a layered, assignable console as accessible and safe as possible for live work.

There are a few irritations. Switching the console on or off produces a thump on the monitor outputs whether the gain is down or not, for example, and phantom powering is

globally on or off for all 16 inputs and the manual gives dire warnings about switching the 48V with the faders open. Given the intended applications, I would also have liked to see monitor and main output cut and dim keys. And 55s may not sound a long boot-time, but with an audience waiting it could seem like eternity—the cute Pacman chewing its way across the LCD fails to distract. Spirit wisely advocate the use of an inexpensive UPS. On the plus side the clever E Strip and bright, internally illuminated keys enable quick and intuitive control essential in live work.

Like the 328, the EQ and dynamics have an 'analogue' feel to them. I think the character of these will suit live work very well. Much the same applies to the effects. If you like the Lexicon sound and way of doing things the 324 will not disappoint.

With both the character of the sound and the control surface, Spirit have successfully differentiated themselves from other manufacturers. If you need 32 channels of live console in a compact package the 324 makes a lot of sense. ■

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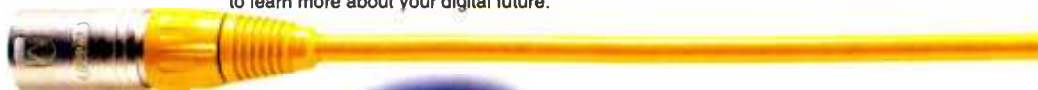
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Roland VP9000

If the 'VP' mark is now a familiar line in voice processors, the latest technology-laden model redefines Roland's role in postproduction circles. **Jim Betteridge** sings its praise

WERE ALL FAMILIAR with the idea of time-stretching and time-compression—the commercial's voice-over is a perfect performance, but a couple of seconds long; the loop is great, but 20bpm too fast... Theoretically, each one of these problems can be sorted using the Time function on any one of many DAWs or samplers. The problem is that, beyond the tiniest correction, especially with timestretch, strange artefacts creep in and the results become unusable. Similarly, changing the pitch of a performance is possible with many systems via their onboard DSP, but you quite quickly get the Donald Duck

change. The VP9000 is different.

With other systems that I'm aware of, either the time and pitch processes sound quite ropey or they need to be rendered—you ask it to pitch the vocal up a tone and wait several times its real-time length while the machine rerecords a new pitch-shifted version. With the VP9000, once you've encoded your sample, it's a real-time effect, making things much easier all round. I used an expensive sampler and a mid-priced DAW against which I compared the audio quality of its performance, and I can say that it was at least as good, and often very significantly better—especially when pitching up a solo

ing, especially where the line is quite rhythmic and well-defined. I'm sure it's also a matter of practise. Of course, what you don't get is the different emotional and tonal nuances at each pitch, so don't worry if you're a singer, this is certainly no replacement. Again, good for backing vocals if you're in a jam.

For constructing a piece from a number of disparate sources, the VP9000's ability to automatically match things is excellent. Let's say you have a number of musical elements taken from sample CDs that you'd like to combine into an arrangement, but they're all at different tempo's and in different keys,



or Darth Vader effect coming into play. If all this is a source of significant frustration to you, you may welcome news of Roland's wonder-box, the VP9000. It takes the whole process to another level.

The machine comes as a 2U-high rack-mount box with SPDIF and optical digital I/O, stereo balanced analogue in and six balanced analogue outs—everything being on TRS jacks. There's also a handy balanced mono line-level input on the front panel. This is not a guitar input and you wonder at the logic of making it mono. Perhaps it was simply available physical space.

The device works like a sampler. You record some sound in, top and tail it, loop it if desired, encode it (it took about 40s to process a 30s stereo file) and then twiddle one knob to slow or quicken it, another to pitch shift it, and another to groove it (swing it).

Lest there be any confusion, we are not talking about playing a sample up and down a MIDI keyboard. This results in a kind of varispeed effect where as the pitch changes so too does the tempo or speed of the sample playback. The processes we're discussing allow either pitch or time to be altered without it effecting the other. Neither should this process be compared with the likes of Steinberg's Re-Cycle which, ingenious though it is, works by chopping up a rhythmical sample into its smallest significant rhythmical components and then replaying them closer or further apart from each other to effect a tempo

voice while maintaining a natural sound. Perhaps the most significant difference, though, is the real-time control.

All playback parameters can be controlled via MIDI. One obvious application of this is a vocal to which you want to add harmonies. As with most other devices, you choose a key on the keyboard as a reference, say middle C, and pressing this will replay the sample at normal pitch. Adding the E above it will give a major 3rd, the G a major 5th, and so on. There are various triggering modes so that each time you add a note to your chord the additional pitch is either played back from the start of the sample or is triggered in synch with the originally pressed key. With the Human effect switched in to add minute pitch and timing differences to each of the notes added, the results when used with a vocal line (possibly the most demanding application) were better than anything else I've heard. Not the same as having a really good singer adding real harmonies, but very usable for backing vocals—which is more than can be said for most real-time harmony boxes.

Going one step further, Robot mode allows all the pitch information to be stripped from a sample. So in the case of the vocal line it is sung on a robot-like monotone. The idea is that you can then replay the phrase with a brand-new melody as played by you on the keyboard. Although somewhat variable, this can sound surprisingly convinc-

Robot mode allows all the pitch information to be stripped from a sample. So in the case of the vocal line it is sung on a robot-like monotone. The idea is that you can then replay the phrase with a brand-new melody as played by you on the keyboard

As you record each element in, the system works out the tempo and gives you a click against which you can check whether or not it's got it right. You can then tell it what tempo you'd like to work at and it will automatically stretch or compress each part to comply. The pitch of each part can then be adjusted either from a MIDI keyboard in semitone intervals or by ear from a knob on the front panel.

If you're working with a MIDI sequencer, you can lock the tempo of your loops to the MIDI clock so that longer samples will stay in sync with each other plus track your tempo map. It is also possible to map any MIDI controller to any of the main parameters so that, in the relevant window on your computer sequencer, you can draw with some >

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Because you're able to adjust the speed of delivery in real-time, it's possible to speed up certain words or phrases while leaving others in their natural states. If you have a MIDI sequencer, your front panel 'performance' can be recorded and replayed, so you only have to get it right once

< precision the variations in speed and pitch against time. In this way you can correct poor intonation and add nuances to a performance, elongating a vowel here, adding portamento there. Equally with a voice-over, it's amazing how the emphasis or mood of a phrase can be changed through subtle speed-pitch variations. The folks from the agency will love it. You'll be charging by the hour, of course. Because you're able to adjust the speed of delivery in real-time, it's possible to selectively speed up certain words or phrases while leaving others in their natural states. If you have a MIDI sequencer, your front panel 'performance' can be recorded and replayed, so you only have to get it right once.

When it comes to working with a full mix of, perhaps a song with a pumping bass, sizzling percussion and legato vocals, results are variable; not always superior in quality to my other systems, but, again, adjustments are made in real-time. This means it's worth trying even if time is limited, whereas if you have to wait for each new attempt to render, clients can get bored and the tendency might be to look at the clock and try another solution. Similarly, the swing function was a bit variable depending on the programme, but an interesting tool to have up your sleeve.

Although it can be used as a stand-alone device, the VP9000 is designed for use alongside a normal sampler and/or DAW. It's not intended to replace either of these devices, and, indeed, in its own it has some limitations. It's 6-note polyphonic and 6-part multitimbral—each part having its own balanced analogue output (also configurable as stereo pairs). It comes with 8Mb of RAM

and can be expanded to 136Mb. However, the longest a single sample can be is 8Mb—about 50s at 16-bit /44.1kHz stereo. So for longer pieces you'd need to work in chunks, editing the processed pieces back together in your DAW. For narration work this is obviously very limiting. Ideally, you would be able to work directly from disk, as per the GigaSampler. How about a plug-in version to work directly on your Pro Tools or VST files, off disk?

You can add compression and expansion into the record chain if desired, and there are also three effects buses offering chorus, reverb and a range of 40 Roland and Boss multi-effects including some guitar effects, analogue-style modulation, phono-graph, radio tuning and a bit-rate converter if you're looking for that old 8-bit grunge.

The machine is fitted with a 250Mb Jaz drive and comes with a few demo phrases and loops. The rear panel also offers both 50-pin and 25-pin SCSI connectors for use with external drives. Samples and performances (a set of up to six samples plus their processor settings) can be saved to disk. Samples can be loaded directly from discs in the Roland VP-CD, L-CD and L-CDP series plus the Akai S1000 series.

As you'd expect, the build quality is very good. The front panel has a reasonable size display with six soft keys plus the usual array of mode buttons, nudge buttons and a rotary encoder. Apparently the system layout is all quite standard stuff if you're familiar with Roland equipment, which I am not, particularly. I found it occasionally confusing, but generally very workable. The ability to plug-in an SVGA monitor, as with their other samplers, would be welcome, but would perhaps jack up the price beyond the average musician's range. Equally, the facility to add a Mac or PC front end would be useful.

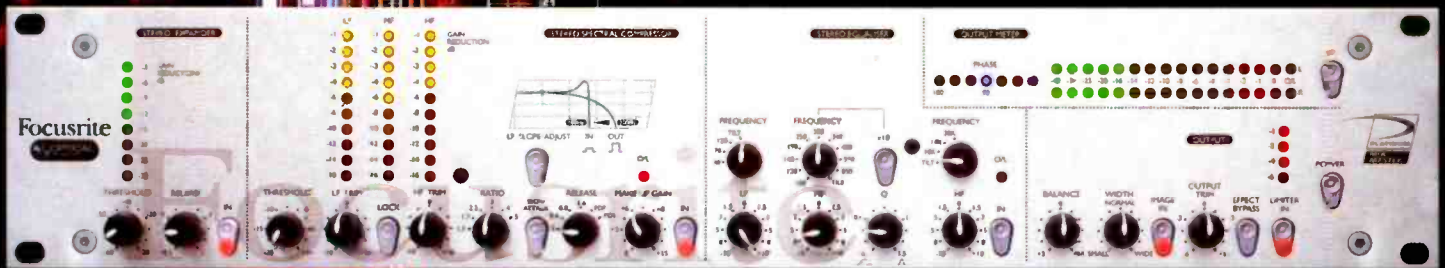
The VP9000 is not cheap compared with other Roland effects units, but it's also far from expensive compared with other big name processors. The ability to take any bit of sound, quickly tune it up, adjust its speed and even lock it to a MIDI clock is extremely appealing. For the musician who wants to deal quickly with loops and phrases, for the sound designer who wants to adjust sounds to fit the picture or for the commercials producer struggling to make the different spot lengths work, it is invaluable. It seemed like a bit of luxury when it arrived. Now I'm wondering if I can do without it. ■

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Sonic Foundry Vegas Pro

Escaping the uniformity imposed by a computer platform, Sonic Foundry's Media Editing System extends the workstation principle. **Rob James** visits Vegas

SONIC FOUNDRY DESCRIBES Vegas Pro as a Multitrack Media Editing System. An interesting sign of the times—the DAW is dead, long live the MES.

Sonic Foundry is no stranger to Windows-hosted audio. Sound Forge, the company's stereo wave editor, is now at v4.5 and the first multitrack product, the loop manipulation package, Acid, has been around for a year or so. Perhaps it is somewhat surprising the multitrack editor, Vegas Pro, has taken quite so long to appear. It has been worth the wait because Sonic Foundry has come up with a programme that really is different both in what it does and aspects of the user interface.

Installation is routine. The only real decision is whether to install the tutorial and demo files to a hard drive—I suggest you do. The tutorial does help to get a fast track grasp on what Vegas is all about. Initial installation requires the use of the serial number in the manual. This will activate the software for a 7-day trial period. To unlock it permanently a code is generated specific to the hardware configuration of the machine. This code must be e-mailed to Sonic Foundry who will return a new activation code. Alternatively, you can phone them. This form of protection is pretty much as irritating as a dongle. In some ways more so, since you will have to convince Sonic Foundry to give you a new code if you upgrade hardware or change machines.

Once past all this nonsense Vegas Pro opens with an optional 'tip of the day' fol-

lowed by a neat main screen arrangement. This consists of three basic 'frames'. Top left is Track List with VOLUME and PAN sliders, TRACK EFFECT, BUS, RECORD ARM, MUTE and SOLO buttons. Top right is a conventional track display and at the bottom of the screen is a frame for docking up to three windows. This works in much the same manner as docking tool-bars and provides an easy way of keeping things tidy. Candidates for docking are: the Mixer window which displays as many faders as there are buses plus a preview fader, the Video preview window, the Explorer and Trimmer windows.

This is a deceptively simple package. Anyone with prior experience of this type of software will find it is perfectly possible to open existing sound files and do some editing without recourse to the manual, on-line manual or help files. However, when the novelty has worn off and you begin to explore there is a problem. The printed manual is a slim volume of less than a hundred pages. The .pdf format on-line manual contains the same and more, but there is still a dearth of information on the assignable plug-in functions. This can only be found in the help files.

The total number of stereo output buses used by a project is set using the Project Properties dialogue. This number can be changed at any time, it does not have to be defined when starting a new project. Each of these buses appears as a fader strip in the mixer window with left and right sliders and

bar-graph meters.

Default setup is a single stereo group routed through the Windows Soundmapper. However, Vegas can cope with multiple soundcards directly and a maximum of 26 stereo buses are available. A to Z. The key to this can be found under the Routing tab of the Preferences menu. Here you can select between using the Soundmapper or Custom Routing. All installed soundcards are available as destinations. The buses defined by the Project Properties dialogue may be routed to any physical card output. This router works as a summing matrix. Routing multiple sources to a single destination is allowed, but, counter-intuitively, single source to multiple destinations is not.

Any bus may be used as a main or aux. It took me some time to work out the auxes since it is possible to inadvertently route a track to the same bus from both the main and aux paths. If you do this the resulting effects are undesirable.

Assignable effects buses do not eat into the 26 main and aux total. I added 12 before boredom set in, although the maximum is 32, each with up to 32 chained effects.

You can use multiple iterations of the same effect on individual tracks. However, using assignable or bus insert effects wherever possible is good practice since it reduces the processor load.

Most multitrack software packages use some form of graphic representation of a conventional mixing console. The constituent parts may be fixed or customisable from a palette of building blocks but the visual representation is strips of familiar faders, buttons and knobs. Sonic Foundry has elected not to do things this way. Only the bus master faders are 'conventional'.

I found the logic easier to follow by thinking about signal flow. The output from each track goes through an optional normalising stage to a preset maximum level of anywhere between 0dBFS and -12dBFS. Attack, Sustain and Release enveloping is then applied followed by track plug-in effects. At this point the signal may be routed to any available aux or effect bus pre or post fader. The path continues via individual TRACK LEVEL and PAN sliders and MUTE switch before assignment to any available main bus.

Effects buses have input and output faders and meters in the Mixer window.

In Vegas the EQ and dynamics mixer building blocks are DirectX plug-ins. However, this does not mean you can use any old third-party plug-in as a Track effect. These require a special 'wrapper'. Sonic Foundry supplies suitably modified EQ, compression and dithering plug-ins with Vegas. Normal DirectX plug-ins may be used as assignable and bus insert effects.

Inserting an assignable effect is a good illustration of the amount of thought which has gone into Vegas. Selecting ASSIGNABLE >



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< FX on the Insert menu brings up the Plug-In Chooser window. You can select any effect or chain of effects. When the selection is made, the control window for the effect, or the first effect in a chain, pops up. Very quick, very intuitive.

Once you become accustomed to the Vegas way of doing things the underlying power of the package becomes apparent. The DirectX plug-ins are economical with processor power and even on a relatively modest 450MHz PIII machine 24 tracks with EQ in each is not a problem. I particularly liked the EQ algorithms and the displays. Parameters may be entered numerically, incrementally or by clicking and dragging points in the graphical display.

Transport controls are conventional with scrub control handled by a horizontal slider, stop to 2x speed in both directions. The fixed forward playback speed is adjusted from 0.06 to 2.00 by positioning a triangular yellow marker. Scrub is not perfect, especially when changing from forwards to backwards at low speeds, but the dedicated controller is a good idea and it is perfectly usable. The only concession to MIDI is MTC and MIDI clock, okay for some purposes, but proper SMPTE would be appreciated. You can work in a variety of time bases including 35mm feet and there are comprehensive grid and snap options.

The Explorer window is similar to Windows Explorer and allows previewing of audio and video files prior to placement. The Trimmer window is a useful place to throw a raw file. The main track display uses a moving cursor while the Trimmer scrolls the track, a curious anomaly. Scrolling tracks uses processor power, but it would have been nice to have the option.

From the Trimmer, the bits you want can be grabbed and positioned in the main track window ridiculously quickly. If more sophisticated detail editing or processing is required you can launch an external editor from here. The file will appear automatically if the editor allows. My usual editor, Samplitude, does not. The obvious candidate is Sound Forge. The snag is Vegas supports 24 bits and Forge doesn't—yet. In any case, I found almost anything I would usually want to do can be accomplished using the tools inside Vegas. Crossfades are generated automatically by simply overlapping two audio events. A right click allows a choice of fade shapes. Trimming, fades and adjusting the snap 'hot spot' are all done by click and drag. A loop function enables an event to be extended beyond its original length. Events can be locked to prevent accidents, but I would have liked to be able to lock events in time while still allowing other tweaks.

Since Vegas is billed as a Media Editor you could be forgiven for expecting at least limited video editing. In fact the video track can be trimmed, but that's the lot, no editing and only one AVI file may be placed on the video track at a time.

In order to overcome a couple of bugs relating to crossfades, and so on. I was obliged to download the 7.5MB update to v1.0b. Hopefully future updates will fulfil Sonic Foundry's intentions to overcome another problem; sync of between sound and picture. This afflicts all the supported formats other

than Video for Windows AVI. DirectShow AVI support has been removed from this version and Quicktime or MPEG both exhibit sync problems. The work around is to save off the video stream as an AVI and work with that.

Many things impress. I like the simplicity of the interface and the file interchange possibilities are a real step in the right direction. Vegas Pro performs sample rate, bit depth and format conversion 'on the fly'. This really is a welcome development. I loved being able to throw almost any old WAV, AIFF (or even MP3 using the supplied plug-in) file into a track and play the lot without any problems. Even better, you can save the final product into a variety of formats and render into still more including complete AVI files with audio and video. Vegas incorporates timeline meta-data; markers which can be used as *aide memoire* or for Internet streaming file authoring. Scott Studios data is also supported.

Although Vegas uses standard Windows drivers I found with projects of reasonable size the playback buffering could be reduced to the point where latency of onscreen controls was more than acceptable. Adjusting levels and EQ is a much less frustrating experience than with many other packages. With the envelope automation of level pan and effects sends, sophisticated work is possible—at speed. Surround panning would have been nice, but I have to keep reminding myself of the price.

This is an attractive package, at this price, almost a great one. If Sonic Foundry can sort out the video synchronisation and maybe add proper SMPTE support it deserves to become a huge success.

Sonic Foundry has recently announced Vegas Video, a comprehensive video and audio editor-processor. (Vegas Pro will become Vegas Audio). I can't wait. ■

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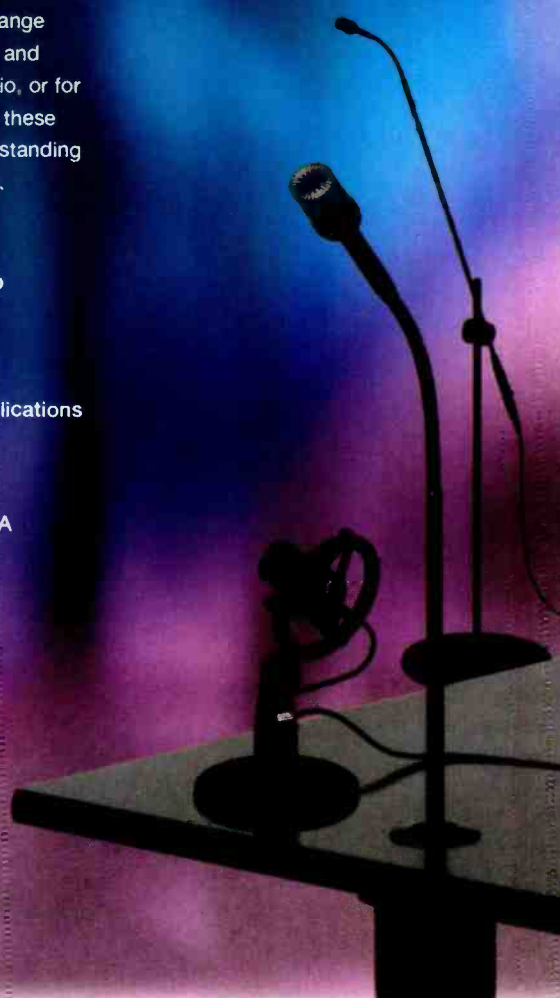
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Rosendahl Nanosyncs

Solving sync problems can be as costly as it is unglamorous.

Rob James finds a line of German units the answer to a prayer

IF YOU HAVE just paid the price of a decent car for a digital mixer, multitrack, outboard effects and converters, and/or a DAW and video, the last thing you want to hear at this point is clicks, grainy audio or somebody telling you to need to spend yet more money. But, as many have discovered, unless you are prepared to spend a little more you may well hear these artefacts and worse.

Happily for you, the Munich-based Rosendahl Studiotechnik makes a useful range of small boxes for a variety of time code and

tising errors, effectively reducing the bit rate. The optimum solution to sync problems is the use of a single source of wordclock, ideally locked to video via a common distribution amp. In any environment involving video, a 'black and burst' video sync generator will also prove useful if not essential. Obviously, if video is used as the master reference, its stability can affect wordclock as well. If MIDI, time code and/or RS422 synchronisers are involved these must be referenced to the same source as wordclock if timing nightmares are to be avoided.

The problem is that high-quality sync generators—let alone video sync generators—can cost as much as a small digital console, and distribution amps are not cheap. Consequently, it is extremely tempting to try and do without. In a simple setup with a single recorder and console this isn't usually a problem (assuming the internal generators are of decent quality); the trouble starts when more devices are added. It is really not a good idea to 'daisy chain' sync between several machines and operationally it is all too easy to end up with the tail wagging the dog.

The Nanosyncs is a shallow 1U-high, all metal box with just three switches and

wordclock applications, the latest of which is Nanosyncs. This unit combines generation and distribution of wordclock and video syncs using either a high-stability internal clock or a variety of external sources for reference. Two models are made to suit PAL and NTSC.

Much has been written on the subject of digital synchronism and jitter. Apart from the obvious and sometimes catastrophic effects of total lack of sync, other more subtle degradations arise with a low-stability timebase. By common consent, jitter is something to be minimised since in excess, it gives rise to quan-



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Millennia's Origin STT-1

Origin STT-1 'Straight-to-Tape' mono analogue recording channel with Twin Topology merges pure class-A tube and pure Class-A discrete solid-state circuit topologies. With over 130 different product combinations in a single 2U-high rack chassis, the STT-1 permits patching class-A tube equipment with discrete solid-state class-A equipment, along with euphonix transformer-coupling, transparent non-transformer coupling, and various input and output configurations. An array of front panel switches select between tube or solid state mic preamp, tube or solid state opto-compressor-limiter, tube or solid-state 4-band parametric EQ, tube or solid-state line input gain tube or solid-state de-essers, along with a tube DI followed by a tube or



solid-state gain stage. The Origin also includes a switchable audio path transformer specially designed for artistic audio 'colouration' at high levels.

Millennia Media, US.Tel: +1 530 647 0750.

25 LEDs on the front panel. Rear-panel connections consist of 14 BNC sockets, a phono and an XLR plus two pole mains. These provide four video sync (black and burst) outputs, six wordclock outputs and AES-EBU-SPDIF outputs. External reference inputs are provided for composite video, wordclock, and LTC and there is an LTC thru output. Each of the wordclock outputs are individually configurable. The first three can take either the basic sampling rate (44.1kHz/48kHz) or times two (88.2kHz/96kHz). The other three offer basic sampling rate or times 256 (Digidesign Super Clock). The common pull-up and pull-down factors 0.1% and 4% are supported for audio, video and film transfers.

Word clock is generated by a technique known as DDS (Direct Digital Synthesis) using a RISC controller with a claimed resolution of 73ps (pico seconds). The internal video reference is claimed to be accurate to 1ppm between 15 C and 30 C. Alternatively, the DDS can be resolved to any of the external references. In internal mode the video outputs are connected to the internal black and burst generator. In all other modes they distribute the video input signal. Lock range to external wordclock is from 32kHz to 51kHz and absolute clock jitter is claimed to be better than 800 pico seconds in all modes.

In the LTC modes word clock is resolved to external time code, forward running with a

$\pm 10\%$ window. The unit calibrates to the source over 30s or so. Once calibrated looping, ramping and low-speed or high-speed code should not affect the clock. The brightness of the blue LOCKED LED changes to indicate the level of jitter and deviation between the incoming LTC and the stable clock outputs.

I was unable to scientifically assess the stability claims, although I have no reason to doubt them. To retain maximum benefit from this degree of precision, wordclock cables should be of identical length and under 10m.

Operation is ridiculously simple. Pressing the MENU key enters edit mode and the current reference source LED blinks. Pressing the SELECT key cycles through the options. Subsequent presses on the MENU key move through the Basic Sample Rate and pull-up/pull-down sections and then each wordclock output to select the frequency multiplier. Once the desired parameters are set, pressing and holding the MENU key until the LEDs stop blinking stores the setup in non-volatile memory. For a temporary setup you simply stay in Edit mode. Repowering the unit will reset the previously stored parameters. At all times the front panel clearly displays what is happening via bright LEDs. The various options allow all the commonly encountered permutations to be achieved.

For many applications Nanosyncs will be the answer to a prayer, a simple and affordable way of solving video and digital audio

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Amek Channel in a Box

This box is the centrepiece of a range of unusual units designed by Rupert Neve. **George Shilling** seeks pure pleasure

CHANNEL IN A BOX differs from most outboard input channels in that there are two completely separate paths always available, mic and line, both with line-level output. Each has its own XLR inputs and outputs on the rear panel, and unusually, XLR connectors for external faders, if you don't want to use the (necessarily) small knobs on the front. Here there are input and output controls for the two channels, along with filters, EQ and compressor, which can be assigned to either path. However, these are not simple, cut-down bargain processors—each section is fully featured.

Connecting power at the rear results in two surprises. The first is fan noise—it is not loud, but a little whiney, and with no front panel power switch could be irritating (although when rackmounted it is probably less audi-



ble, with the vents at the sides). The second surprise is that the badges on the rack ears illuminate—groovy!

For a 1U-high design, there is an astonishing amount packed in here. All knobs are well-damped and buttons lit in bright colours. A MIC GAIN rotary switch is accompanied by a FINE TRIM, and there is a LINE TRIM pot. Switches for PHANTOM, PHASE (separate for each path), and LED-METER SOURCE are provided, along with overload LEDs. Next are the high-pass and low-pass filters, selectable for either mic or line and the side chain. The EQ section is cleverly divided—its 4 bands are arranged LF, LMF, HMF, HF reading from left to right, however the two mid bands are grouped together, as are the extreme bands, as regards switching and routing. These, like the filters can be applied to mic, line or side chain. Extra LEDs on the LF band helpfully indicate status (as this is dually controlled by switches for and alongside the HF band). Next comes the compressor, which can be switched to the mic or line paths. Pressing KEY picks up the control signal from the opposite path to that which is selected. Finally, there is an output fader knob for each path, with zero centre detente. On the rear are XLR sockets to connect external faders, along with selector switches. A METER SOURCE button switches the Input meter to show Output Level, and this can be set pre-fader or post-fader, depending on internal jumper settings. All push-buttons glow colourfully when depressed, giving good visual feedback as to status, essential on such a crowded front.

So what does it sound like? The mic amp is described, as a TLA (not the company, but Transformer-Like-Amplifier). Firstly, the mic amp holds up well alongside any competition.

This uses a circuit that supposedly behaves like a transformer, with 5kΩ impedance. At unity gain it can handle more than +20dBu and can therefore be used happily as a second Line input. Unsurprisingly, it retains the System 9098i and Focusrite characteristic sound, with very detailed extreme high frequencies, and extended and smooth low frequencies, all masterfully under control. There is plenty of gain, and performance is excellent. Mr Neve's views on extended frequency response are well documented, and this unit performs well above and below the frequencies where most of the competition has rolled away.

The filters are powerful 18dB/octave with wide-ranging continuous sweeps. The EQ is, not surprisingly, quite similar to previous AMEK-Neve designs. The low frequency band features PEAK and GLOW buttons, the high fea-

tures PEAK and SHEEN buttons. For glow and sheen, read shelf when boosted, broad-band bell-curve when cut. Both mids feature x5 frequency shift buttons, for a huge range—these cover higher and lower frequencies than the high and low bands respectively. Each band has massive ±18dB gain and a separate Q control. And it sounds just as marvellous as other recent Neve designs, with plenty of clean power available.

The compressor features pots for THRESHOLD, RATIO, ATTACK, RELEASE and GAIN, and a LINK button for multiple units. There is also a mysterious &MM button, which stands for 'And Much More'. In fact, this switches the unit from hard-knee to soft-knee operation, which gives rise to much more audible compression characteristics, especially at reasonably fast settings. The ranges of the Attack and Release knobs are very wide, covering all possible requirements. With the amount of features present, it would be churlish to question the lack of any automatic settings. At the top is an LED-meter showing gain reduction to a fairly fine resolution.

With extensive side-chain and key routing, it is a very simple matter to set up a de-esser, or other frequency-conscious compression.

The Filters, Mids and Extreme EQ sections are all separately switchable between mic, line and side-chain paths, so many possibilities are available to the ingenious engineer.

This is an astonishing feat of shoe-horning, and all the expected Rupert performance is in evidence, along with wonderfully solid Amek build quality. My only slight reservation would be the fan noise—this unit is excellent value, and comes from an unusual design approach. ■

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Tel: +44 161 868 2400.
Net: www.amek.com
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Tel: +1 818 508 9788.

NEW TECHNOLOGIES

Quad and dual dynamics

The Altair NG404 offers four identical noise gates in 1U. Dedicated filters are provided for triggering with test monitoring of the signal. Fully variable threshold and release are complemented by switchable range and



attack settings. Channels can be linked for stereo operation. The CN202 is a dual-stereo compressor noise-gate again with dedicated filters for the side-chain trigger and similar gate-parameter controls. Band compression is available with adjustable frequency selection, threshold and ratio.

Altair, Spain. Tel: +34 918 04 32 65.

Digital EQ and dynamics

The TwinEQ2001 is a digital 4-channel 30-band EQ with two of the bands available for use as variable low-high shelving or low-high cut filters. Each channel also has a 32-bit delay for up to 2.5s and a peak limiter. Control is via PC software or EQR2001 remote for 32 or 16 channels respectively. The TwinCOM2000 is a digital 4-channel dynamic processor with 32-bit floating point processing and 24-bit converters. Settings can be stored in 100 memories. The box offers compressor, limiter, expander and gate duties with hard and variable knee characteristics.

Kronauer, Germany. Tel: +49 78 21 98 32 41.

Flash recorder

Denon's DN-F20R flash-memory recorder is targeted at radio and television acquisition and is powered by six AA batteries, an external battery pack or a mains adaptor. The machine weighs 1kg and comes with balanced XLR mic inputs, bass cut, a lim-



iter, input attenuators, recording level control and headphones socket with level control. As well as handling linear PCM audio it also records in MPEG Layer 1 and Layer 2 and a twin flash card port when filled with 192Mb gives 17 minutes of stereo linear audio at 48kHz or 408 minutes at MPEG Layer 2. WAV files are supported.

Denon, UK. Tel: +44 1753 880109.

Swissonic

Forthcoming digital-audio solutions from Swissonic include the new WD8 wordclock unit and two additional USB products for working with audio on a PC or Macintosh computer. Designed to solve digital audio clocking problems, the WD8 wordclock generator-distributor has two wordclock inputs and eight wordclock outputs. In



William Wittman is a multi-platinum Producer/Engineer, former Staff Producer/A&R Vice President (RCA/BMC Records and Columbia/Sony Records), Musician and Songwriter. His career truly covers all the bases.

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LSR. Profiles

The world’s most noted recording professionals discuss the world’s most advanced monitoring systems.



Joel Jaffe is an award winning Engineer/Producer/Composer and co-owner of Studio D Recording, Inc., home to a long list of platinum and Grammy Award winning albums and artists. Currently, Joel is working on DVD surround mixes for some of the industry’s top touring acts. LSR surround systems are his choice for stereo and 5.1 channel multimedia projects.

“The THX Approved 5.1 JBL LSR28P with the JBL LSR12P subwoofer provide an extremely linear response, great transients and full-frequency monitoring in a near-field set up. In addition, the LSR speakers allow us to be able to go between stereo mixing and multi-speaker formats, which is absolutely necessary today in a state-of-the-art studio.”

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The Only Workstation Monitor Good Enough To Be Called LSR.

Daking 52270 & 91579

Geoffrey Daking's compressor-limiter designs revive aspects of assorted classic designs. **George Shilling** spills his secrets

I WOULD BE SURPRISED to learn that many readers outside the US have heard of Geoffrey Daking. I certainly hadn't, but I can now tell you that the man himself is a studio owner and 'audio tinkerer' whose units have been available in the US for a couple of years and have quickly attracted an impressive list of famous users.

If you read Daking's promotional literature, the names Trident, API and Neve make several appearances and this upper echelon of the market is where he aims to compete. He criticises the cost-cutting aspect of modern console design philosophy and takes a no-compromise approach. Discrete class-A cir-



cuitry is the order of the day here (although the physical heat associated with such designs is, thankfully, not particularly apparent). Daking's US website claims that these models undercut API 500 series prices considerably, although after crossing the Atlantic there is not a huge difference when power supplies are taken into account.

These well-built units are somewhat larger than API modules. Each takes up 1U of rack-space. However, a glance around the back reveals fairly shallow, narrower cases, approximately the size of Neve 1073 modules, with metal braces attaching them to the full width of a standard rack front. Power comes from a separate box via 25-pin D-connectors. The power supply can accommodate up to four modules. Daking also supplies a custom 4U-high rack case, which is padded and includes a top compartment for storing power supply and leads, with a neat Daking logo on the top—very natty.

All controls except the EQ GAIN/CUT and MIC PRE FADER knobs are switchable. Much is made of the fact that the big metal knobs are custom made. I like their size, but they have a rough, clunky feel. Legending is unexceptional but a vertical front panel arrangement is an option.

The mic pre's coarse switchable gain is labelled in a slightly confusing way with nominal input level rather than gain, so it looks backwards at first. Positions to the right are for line input. In the centre of this is a FADER pot, that negates any need to use a desk channel. Switches for PHASE, PAD, MUTE and BYPASS all light and use relays with gold contacts. The Mic Pre is wonderfully upfront and exciting, making instruments and vocals sound beautifully enhanced, even with lowly dynamic microphones. Everything sounds punchy and crystal clear, with aston-

ishing presence, yet, paradoxically, more natural than the perceived ultra-high frequency detail of Focusrite designs. This suits most acoustic instruments and vocals well, although very occasionally (such as with a miked bass cab) something less bold seemed more appropriate. I loved the Mic Pre and would have thought a separate mic amp might be a popular unit at a lower price. It goes without saying that the noise and distortion figures are up with the best. Signal connections are XLR, mic input phantom is always on. The website PDF manual (no printed one was supplied) mentions an insert socket to enable pre-EQ compression but this was not present on the review model. The EQ is a 4-band design (two shelving and two peaking bands) with no bandwidth controls and just four switched frequencies per band (and an Off position). Daking makes the point that sweepable pots are cheap components, whereas switched knobs cost considerably more. These frequencies are based on a rare seventies Trident design, and are extremely well chosen, with 17dB boost on the mid peaks, 15dB boost on the shelves, and 15dB cut across the board. This is very powerful and really enjoyable to use when making gentle or bold changes to voices and instruments, as if one is sculpting the sound. Perhaps deliberately, there is no legending or calibration of the EQ GAIN/CUT pots. There are also switched high-pass and low-pass 12dB/octave filters, each having only three settings.

The compressor-limiter features Line Input and Stereo Link sockets of the dual-jack/XLR type, while output is on XLR. The vu meter looks fairly cheap, but seems to work well, although illumination may have been useful. It can be toggle-switched to show Input, Output and Gain Reduction. A second toggle-switch enables Stereo-Link and Bypass modes. All rotary controls are switched, which with 2dB steps means any really fine threshold adjustment must be made externally with the level before signal arrives. Ratios can be chosen at 1.5:1, 2:1, 3:1, 5:1, 10:1 and 20:1. Of course, much depends on the attack and release settings. Attack is calibrated from 250µs to 64ms with nine settings. Release is labelled A to G (seven settings) and is quite novel. 'A', 'B' and 'C' represent fixed settings of 0.5s, 1.0s and 1.5s respectively. Position 'D' replicates the dual-time-constant auto setting of the Neve 33609. 'E' is the dual time constant of the Audio Design Complex auto setting. 'F' and 'G' are copies of the triple-time-constant settings of the Fairchild 670 auto positions 5 and 6 respectively. These are superb recreations, and sound warm and smooth. The faster settings are great on drums and sometimes vocals.

Overall these are terrific units and surely qualify for a 'best kept secret' award. Highly recommended. ■

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NEW TECHNOLOGIES

addition, it has a Superclock output for Digidesign systems and an AES-EBU digital audio I-O. The system can lock to either one of the workclock inputs or to the digital input. It recovers the clock and sends it to all the outputs, including the digital output, which is a silent audio signal. The digital input allows a digital source, such as a nonsynchronisable DAT, or a digital recorder without wordclock output, to be part of the system. Similarly, the silent audio signal can be used to synchronise units with a digital input, but without a wordclock input. This product is not expected until autumn 2000. USB Studio and USB Studio D both provide inputs for almost any audio source. The only difference between the two USB units is the additional optical-coaxial I-O on the D version. Both units offer two balanced mic inputs with insert, phantom power and phase-inversion switch. There are also two high impedance inputs for acoustic instrument pickups, four stereo line inputs for electronic instruments and one stereo input with switchable phono preamp for record decks. There are separate outputs for master, monitor and headphones. Also common to both USB devices is the useful ability to switch between input and playback monitoring—either manually or under the control of the recording software—in order to circumvent latency issues. **Swissonic, Switzerland. Tel: +41 55 28 58 610.**

Cheaper Rode

Rode has released the low-cost condenser NT3 with battery or phantom powering. The hypercardioid pattern, wide frequency response and high immunity to RF combine with a heavy-duty cast satin nickel body, high strength mesh head and internal capsule shock mounting. **HHB, UK. Tel: +44 8962 5000.**



Nady amps

Nady has introduced the SRA Series power amps which claim to offer studio quality audio for project recording and live applications. The SRA 2150 is powered at 2 x 150W into 4Ω (250W mono bridged in to 8Ω) while the SRA 2250 develops 2 x 250W and 350W respectively. Both amps are 1U high, and offer calibrated 41 detent volume controls, headphone output with speaker-disable switch for private monitoring, switchable low-cut filters for eliminating speaker rumble and resonance, switchable 2Ω/4Ω/8Ω operation, and stereo/mono-bridged switch. Both include TRS and XLR balanced and RCA unbalanced inputs, and binding post (banana jack) speaker outputs. **Nady, US. Tel: +1 510 652 2411.**

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Yamaha H5000

For methodology see *Studio Sound*, June 1999, page 27.

See it on the web-site:

www.prostudio.com/studiosound/index.html

Studio Sound's bench test amplifier reviews continue with the H5000. **Paul Miller** reports



OVER THE LAST FEW months, *Studio Sound* has bench-tested a number of powerful amplifiers at or around the £1,500 mark, but none come close to matching the prodigious capability of Yamaha's H5000. Rated at 550W/8Ω, this compact, 2U-high rack-mount design succeeds in deliv-

ering at full 2x600W/8Ω and 2x900W/4Ω. Fig.1 demonstrates that with a limit of 0.25% THD, the amplifier exceeds its specification from 20Hz through to 20kHz, the 20kHz rating improving to 590W/8Ω with a relaxed 0.5% tolerance. This compares very favourably with the 375W output of the Hafler P9505, ATC's

200W SPA2-150 and the 300W of the Briston 4B. Neither is the H5000 especially bulky or, importantly, run to within an inch of thermal meltdown.

In practice, forced-air cooling is provided by two variable-speed fans mounted at the front of the case, in line with the 'heat-pipe' sinking that runs the depth

of the enclosure. Proficient cooling is half the story, but Yamaha remains somewhat coy about the electrical efficiency of the design that hinges on a proprietary 'EEEngine' (Energy Efficient Engine) technology. While claiming that the H5000 and its partners, the H3000 and H7000, outperform competing schemes

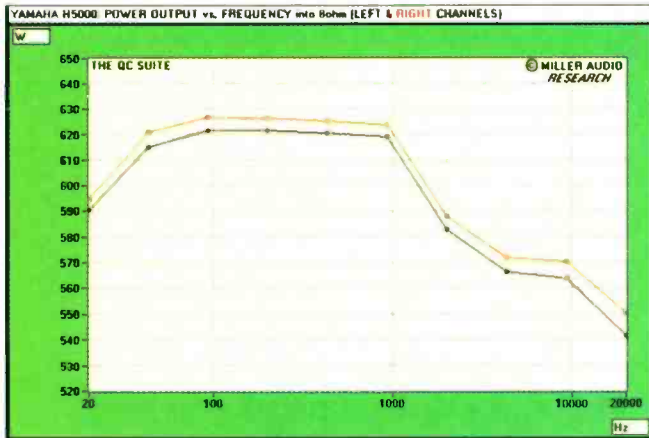


Fig.1

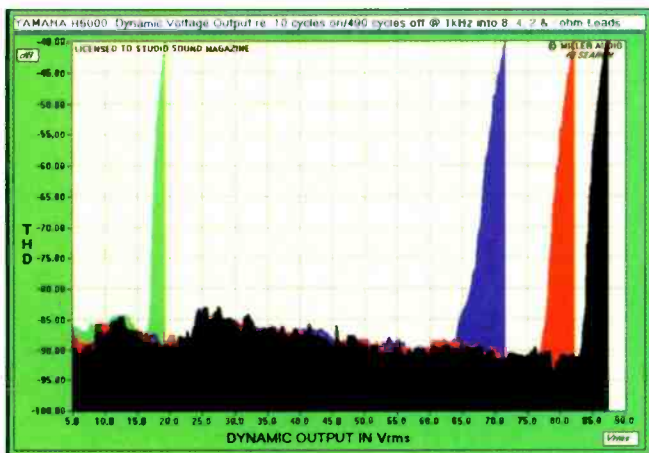


Fig.2

Power Amplifier: Yamaha H5000

(Rated Spec. in brackets where given):

	20Hz	1kHz	20kHz
Max Continuous Power Output:			
1% THD into 8Ω (one channel)	590W	650W	590W
1% THD into 8Ω (two channels)	570W	620W (550W)	570W
1% THD into 4Ω (two channels)	775W	900W (700W)	740W
Frequency Response @ 0dBW	-0.1dB	0.0dB	-0.1dB
Dynamic Headroom (IHF)		+1.7dB (955W)	
Maximum Current (10msec, 1% THD)		35.8A	
Output Impedance	0.014Ω (<0.04Ω)		
Damping Factor	560 (>200)		
Unbalanced Input			
Stereo Separation (1kHz)	90dB (>85dB)		
	(20kHz) 66dB (>65dB)		
Channel Balance, 1kHz (0dBW)	0.07dB		
Total Harmonic Distortion			
(0dBW, 1kHz/20kHz)	-95dB / -76dB (<-66dB)		
(2/3 power, 1kHz/20kHz)	-95dB / -75dB (<-66dB)		
CCIR Intermodulation Distortion			
(0dBW)	-85dB (<-66dB)		
	(2/3 power) -84dB		
Noise (A wtd. re. 0dBW)			
(re. 2/3 power)	82.5dB	108.0dB	
Residual noise (unwtd)	-67.3dBV		
Input Sensitivity (for 0dBW)	72.7mV		
	(for full output) 1780mV (1550mV)		
Input loading	15kΩ		
DC offset, left/right	+1mV / +4.5mV		
Serial Number	PY01004		
Retail Price	£1,499 (inc-VAT)		

including rail-switching, PWM supplies and switching series-regulation, Yamaha still fails to elaborate on the precise details of its own EEEngine topology.

Judging by the mixed pairs of Sanken devices (eight A1492 /C3856 and five A1294/C3263 per channel), it seems reasonable to assume that banks of the devices themselves are switched according to signal conditions. The use of multiple devices not only spreads power dissipation, but also helps reduce the amplifier's output impedance to just 0.014Ω. This suggests the amplifier will be less sensitive to long cable runs and, assuming the cable is of low resistance, ensure the overall system response is unaffected by HF impedance swings on the part of the speakers.

A low source impedance also coincides with the high instantaneous current delivery of the H5000, which achieves 955W (10.9A—black trace), 1690W (20.5A—red trace) and a whopping 2560W (35.8A—blue trace) into 8Ω, 4Ω and 2Ω loads, respectively. The lack of any invasive VI-limiting is evident from Fig.2 which, otherwise, would show a progressive, typically linear increase in THD up to a notional 1% clip limit (in practice, the H5000 is 'in clip' above 0.1%). Difficult speaker loads are clearly no problem, but power-limiting is invoked into loads at or below 1Ω, restricting the output to 'just' 370W (19.2A—green trace).

Power on-off muting is provided along with over-temperature (>90°C) and DC offset protection, the latter indicated by LED's on the H5000's fascia. Two finely calibrated attenuators sit below and, once set, may be locked into position with a set of matching covers. Full gain, or 0dB attenuation, is +31.8dB (x38.9). A single set of Channel A-B 4mm speaker outlets are also concealed by a protective cover on the rear of the amp, alongside male-female XLR and 1/4-inch jack input sockets. An earth lift facility is included to break hum loops while stereo, parallel and bridge-mode output configurations improve its flexibility still further. Should multiple amplifiers be stored in a rackmount cabinet (with a sealed rear section), Yamaha can provide a 1U-high venti-

lation panel to accommodate groups of four H5000's.

The sound of the H5000, too, reflects this cool and slick efficiency. Transients have real bite and gusto and bass drums a solid weight and momentum, for while it lacks the sweetness of the Hafler or the smooth treble of the ATC, the H5000 always seems to cut through to the essential detail of the mix. Distortion is especially low and far lower than indicated by Yamaha's own graphs, which have a limit of 0.05%. Fig.3 shows that THD is typically 15x lower than this at between 0.001%–0.003% over much of the upper bass and mid range, increasing to just 0.015% at 20kHz (re. 10W/8Ω). The FFT (lower plot, Fig.3) shows a declining and subjectively innocuous pattern of 2nd, 3rd and 4th harmonics.

But what of THD versus power output? This, too, is both remarkably low and quite astonishingly consistent. Plotted on log scales, Fig.4 indicates that the mean THD from 1W to 500W is just 0.0015% with a variation of just ±0.0007%. Once again, Yamaha's 0.05% specification represents the top of the Log% axis. Its frequency response is within +0.0/-0.1dB from 20Hz–20kHz (see Fig.5) which, in the 'real-world' is equally impressive, though the A-wtd S-N is a little below average at 82.5dB (re. 0dBW). Figures at or greater than 85dB are more typical these days, though there's plenty of anecdotal evidence to suggest that wider S-N ratios tend to encourage a harder, less forgiving sound as low-level distortion mechanisms (including RF-related artefacts) are 'exposed'.

The semantics of noise aside, Yamaha's H5000 provides a textbook performance and a bench-mark of power output versus compactness and efficiency. Where else will the power-crazed engineer achieve 600W+ at vanishingly low levels of distortion for just £1,500? Sure enough, the H5000 pulls no punches, but, if you want a rose-tinted view of events, then choose a valve amplifier. If, however, you need pin-sharp detail with power to spare, then there's little to touch the Yamaha below £2,000. ■

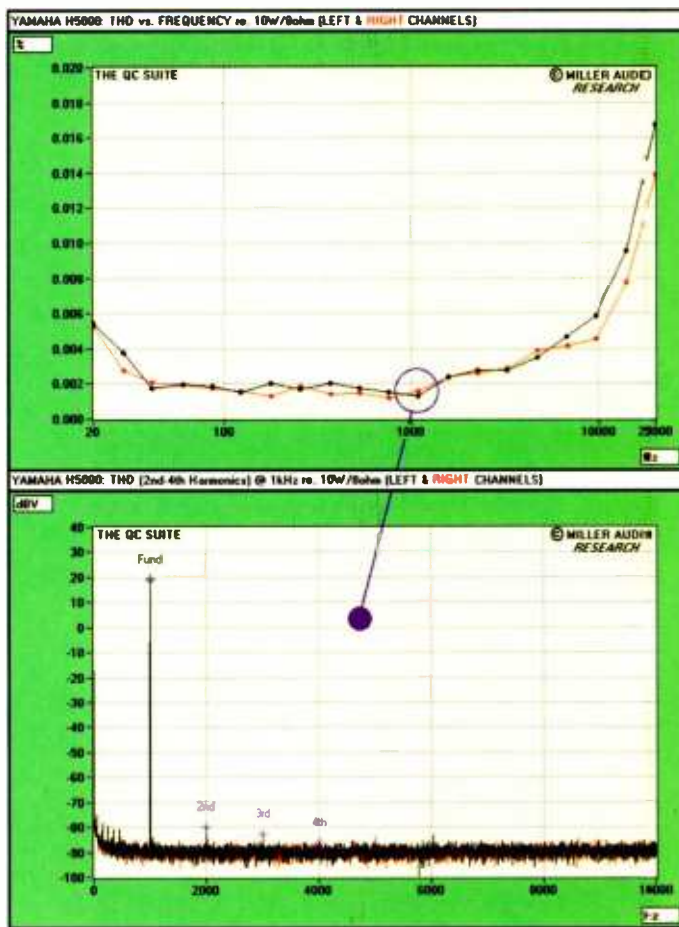


Fig.3

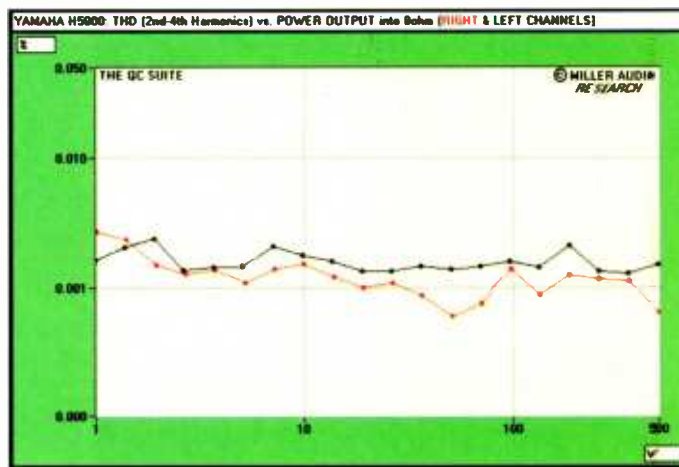


Fig.4

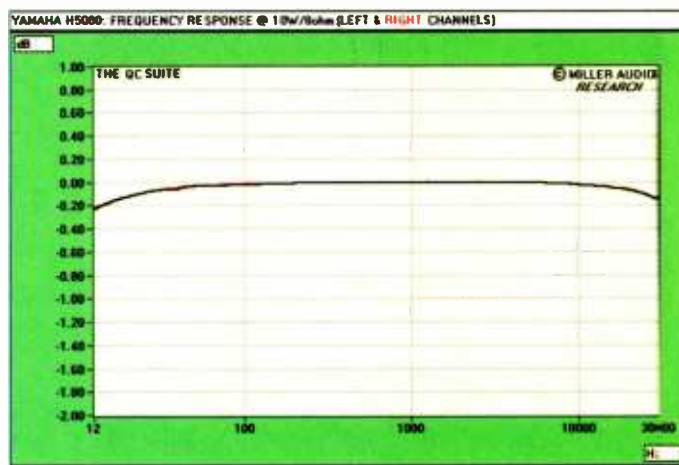


Fig.5

Minipod

For methodology see *Studio Sound*, April 1998, page 14.
 See it on the Internet net-site:
www.prostudio.com/studiosound/apr198/r_tannoy.html

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

THE MINIPOD is a most unusual loudspeaker. The cabinet is of moulded plastic and has a sort of organic or sculpted look, without corners or flat surfaces. It is a 2-way, passive design comprising a 150mm woofer with a Kevlar cone and a 25mm fabric-domed tweeter. The cross-



over frequency is specified as 3kHz with 2nd order low-pass and 3rd order high-pass slopes. The cabinet is equipped with a port below the woofer, has overall dimensions of 340mm high by 210mm wide by 220mm deep and the entire loudspeaker weighs just 2.37kg. The Minipod has a nominal impedance of 4Ω, and, although no maximum output figures were quoted, the manufacturers recommend the use of power amplifiers from 10 to 100W RMS output into 4Ω, which, when considering the nominal sensitivity of 90dB, equates to a theoretical maxi-

mum output level of about 110dB SPL at 1m per cabinet. A set of spikes are supplied, that screw into the bottom of the cabinet to act as short stands; these were not used for this review.

The on-axis frequency response and harmonic distortion for the Minipod are shown in Fig.1. The frequency response is seen to lie within ±3dB limits from 80Hz to 20kHz with a 4th order low-frequency roll-off giving -10dB at about 50Hz. Low frequency harmonic distortion performance is exceptional at better than -55dB (0.2%) except for a peak to -40dB (1%) at around 50Hz. There is some mid-range distortion present with the 3rd harmonic peaking to -45dB (0.6%) at 2kHz. Figs 5 and 6 show the horizontal and vertical off-axis frequency responses respectively. The horizontal directivity is very well controlled, with a coverage angle that narrows smoothly with increasing frequency; there is no evidence of mid-range narrowing. The vertical results clearly show the interference notch in the upward direction which is a characteristic of most non-coaxial driver designs. The step response (Fig.3) demonstrates excellent time-alignment with almost zero delay between the high and mid frequencies. The acoustic source position plot (Fig.2) is typical of a 4th order system, with the low frequency parts of transient signals starting some 2m behind the loudspeaker. Some evidence of reflections or echoes at 0.2ms, 0.4ms and 0.55ms can be seen in the power cepstrum (Fig.4); these are responsible for the irregularities in the on-axis frequency response seen in Fig.1. Finally, the waterfall plot (Fig.7) shows a rapid decay at low frequencies, and very little ringing in the mid frequency range. The Minipod is indeed an unusual loudspeaker. Whether one appreciates the styling or not, the measurements presented in this review

demonstrate that the design works extremely well. The harmonic distortion in particular is exceptional for a loudspeaker of this size, putting many larger and more powerful designs to shame. The frequency response is a little uneven, but it stays within ±3dB limits nevertheless, and the time domain performance is ex-

cellent with near perfect driver time-alignment and a rapid decay at all frequencies. At only 2.37kg each, the Minipods are by far the lightest loudspeaker tested to date, and are therefore ideal as a portable reference; indeed, a purpose-made rucksack-type carry case is available as an optional extra. ■

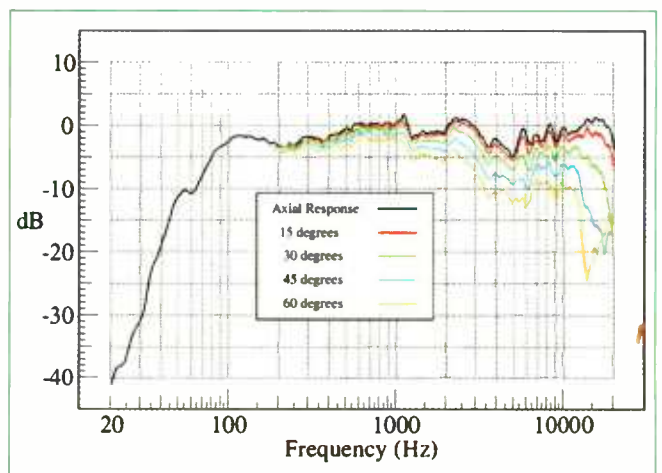


Fig.5: Horizontal Directivity

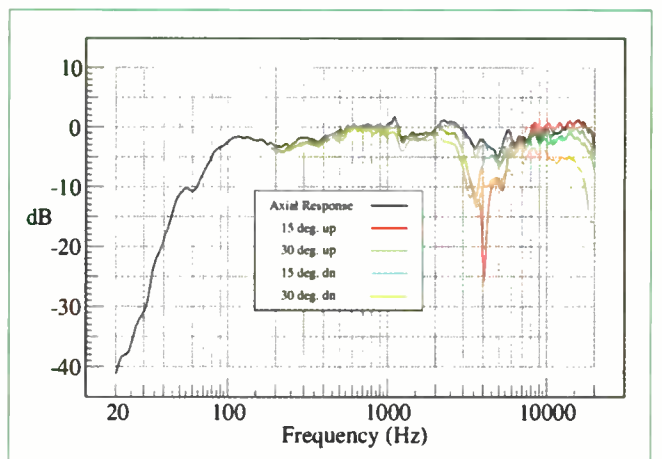


Fig.6: Vertical Directivity

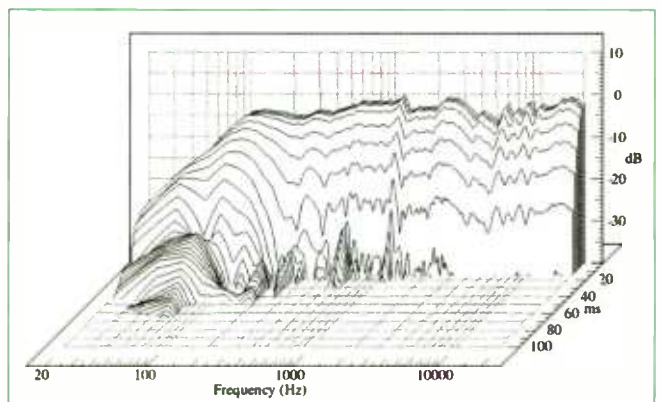


Fig.7: Waterfall

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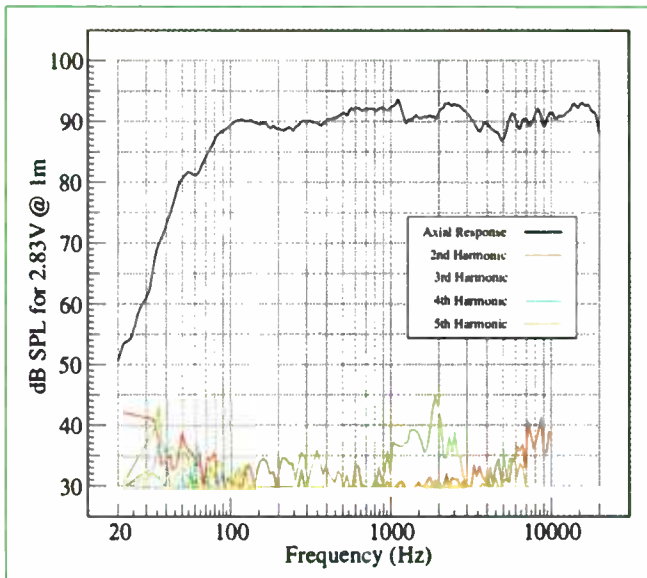


Fig 1: On-axis Frequency Response and Distortion

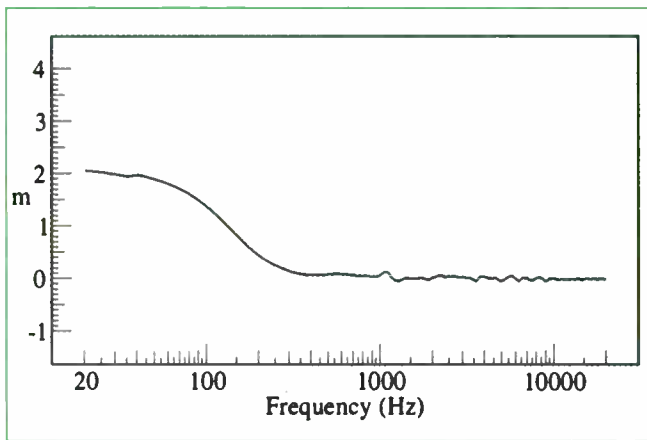


Fig.2: Acoustic Source

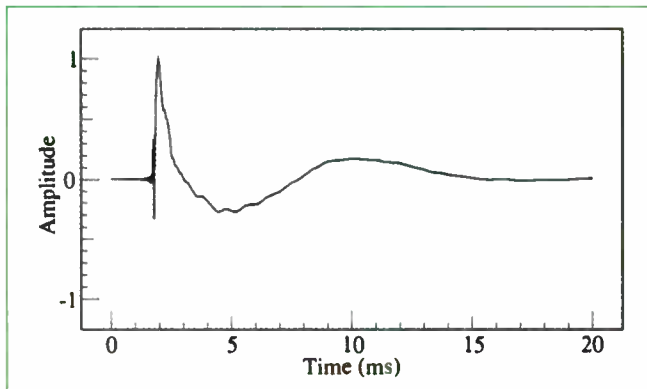


Fig.3: Step Response

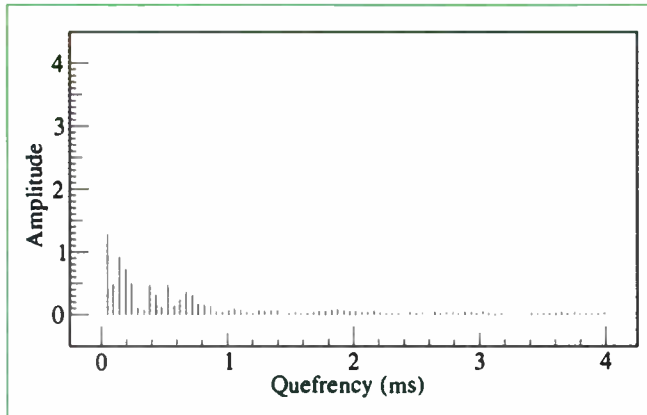
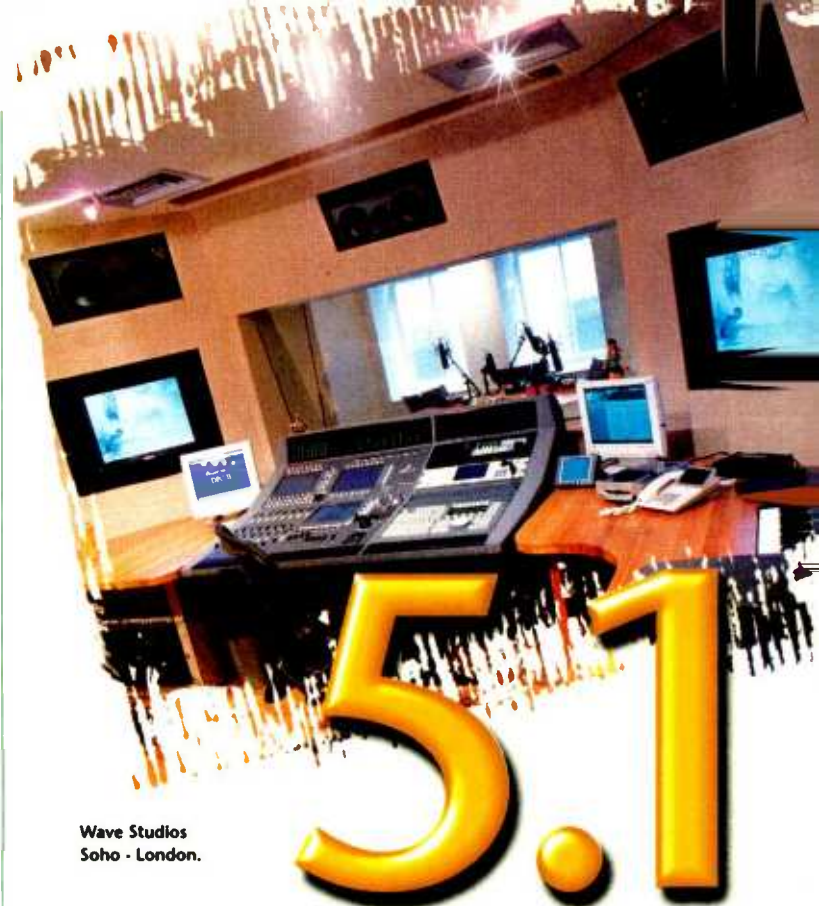


Fig.4: Power Cepstrum



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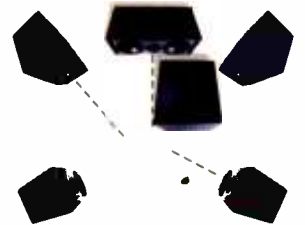
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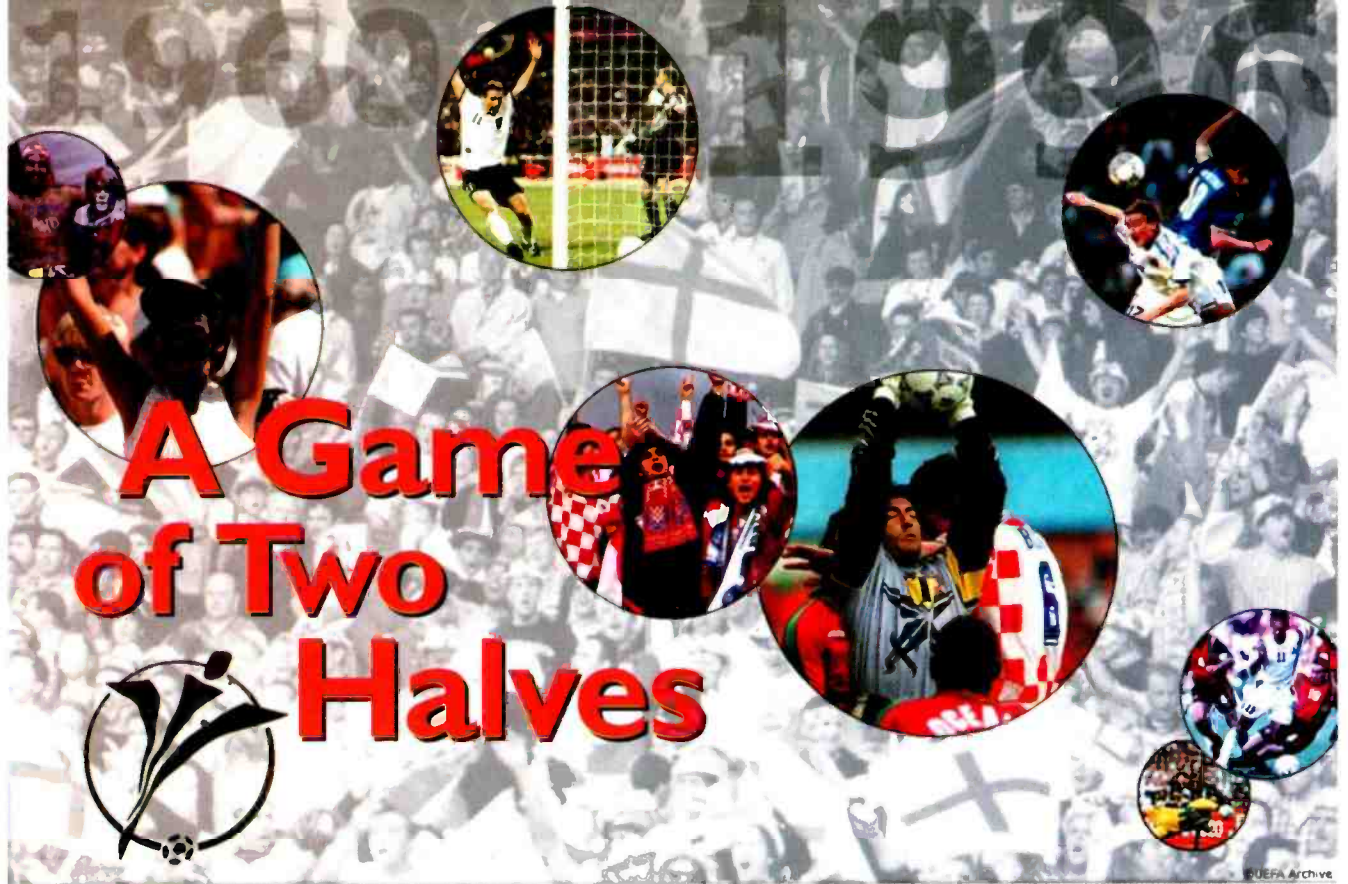
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A Game of Two Halves



The Euro 2000 football tournament is set to build on the success of earlier years' events and their broadcast arrangements. Euro 96 vet **Neil Hillman** listens in

WHEN THE EUROPEAN football championship finals, EURO 2000, kicked-off on 11th June, the first major sporting event to be broadcast in the 21st Century, ahead of the Sydney Olympics, had commenced. With 31 matches to be held in eight stadiums across two countries simultaneously, 80 visiting broadcasting companies, 2,000 crew members and a cumulative worldwide television audience of a projected 7bn viewers, the host broadcasters' placed themselves in the glare of the industry's spotlight and under the critical gaze of an evermore discerning viewing public.

In July 1995, the European Football Union (UEFA) selected Belgium and the Netherlands to host the European Football Championship 2000. In accordance with the agreement between the European Broadcasting Union (EBU) and UEFA on broadcasting rights for the championship, the Dutch and Belgian EBU-members Nederlandse Omroep Stichting (NOS), Radio, Télévision Belge pour la Communauté Française (RTBF) and Vlaamse Radio-en Televisieomroep (VRT) were commissioned to act as the Host Broadcaster. These three companies subsequently formed a joint venture, similar in principle to that engaged

by BBC and ITV companies for EURO 96 in England—Football Operations, Radio and Television Organisation; or more familiarly, FORTO 2000.

FORTO 2000's main tasks are twofold. Firstly, there is the production of multi-lateral video and audio signals—known as the International Feeds of the matches—and the transmission of these signals to all rights-holding Broadcasters in the world. Secondly, there is the provision and co-ordination of unilateral services and facilities required by rights-holding broadcasters wishing to personalise the coverage with their own programmes.

On one side are the host broadcasters, serving many and varied masters; on the other are the visiting Outside Broadcast Sound Supervisors at the behest of one.

Last year when FORTO assembled its team for EURO 2000, it selected Fernand Holthof as one of the venue managers.

'We have a large compound, not too far away from the stadium. Cable ducting into the ground already existed, but two new extra runs were built for Euro 2000. The effects microphones, 23 in total—four on camera and 19 static—are configured for stereo international sound, but not for surround. It is a rather common array for football, but we still had a few test matches to try different setups.'

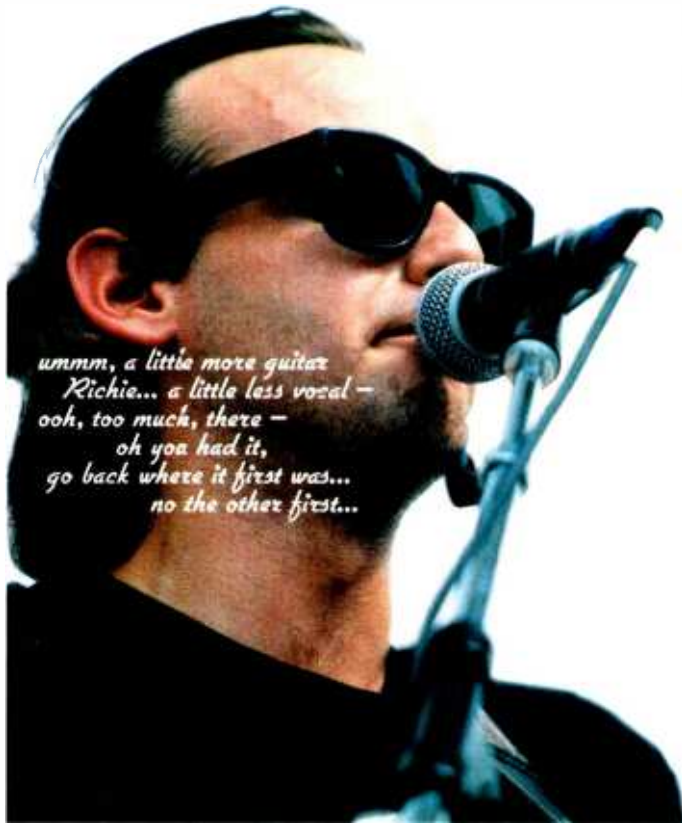
FORTO 2000 will equip 600 commentary positions for radio and television reporters in the eight stadiums, enabling all broadcasters the opportunity to add their own live commentary and match reports. In each stadium, two 'flash' interview positions are installed in the players' tunnel, enabling either prematch, half-time and full-time short-duration 'on-the spot' interviews with

participants to take place. Two television interview studios and one radio studio are created in the players' dressing-room area. Also, within each stadium will be a commentary gantry and at least five presentation studios, each with a view of the pitch to allow broadcasters to present their pre-match introductions, and the usual format of half-time and full-time interviews, chat and analysis with guests and experts.

The matches of EURO 2000 will be covered by 18 cameras situated in identical positions across all the stadiums and match-sound will be produced in stereo through a 14-microphone array. Two hundred kilometres of cabling has been prepared to connect the 144 cameras, 122 microphones, 32 slow-motion machines, and eight graphic workstations across the tournament. The matches in Charleroi and Liège will be covered by RTBF crews, in Bruges by VRT. The matches in Brussels will be shared between RTBF and VRT, while NOS will cover the matches in Amsterdam, Arnhem, Eindhoven and Rotterdam.

It has been a year in planning for Holthof since he was selected for the job as venue manager.

'Of course for an event of this size, at each stadium there is a TOC (Technical Operation Centre) where all the necessary cabling comes and goes, whether it is digital or analogue video, or analogue stereo for the international sound. A part of the TOC is the Commentary Control Room where all the commentary circuits can be monitored. From the TOC the sound will go embedded with video over different fibre cables on the ATM-network at 270Mb/s to the International Broadcasting Centre in Amsterdam. There it is uplinked to EBU >



*ummm, a little more guitar
Richie... a little less vocal -
ooh, too much, there -
oh you had it,
go back where it first was...
no the other first...*

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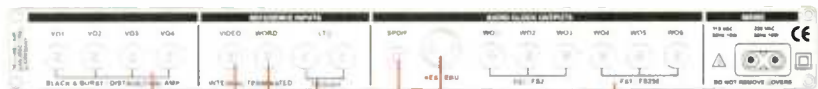
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Transmission circuits from the venues to the IBC are completely digital and terminate at 270Mb/s (CCIR) with embedded 2-channel audio (AES-EBU). The circuits will use MPEG2/4:2:2/24Mbps and will be transported via optical fibre and microwave links. During EURO 2000 there will be a total of 64 video and audio circuits available.

'It is hard to tell where we will have the most problems, but we feel we can help solve all the problems visiting broadcasters could have when they come here. Hopefully, all the visitors will need to worry about is the weather!'

Tim Butcher is a sound supervisor for Carlton-021, arguably the biggest and most successful OB company in the UK. He will be responsible for the ITV Sport audio from their various featured matches throughout the tournament.

'At that time I felt confident in many aspects of sound recording, but I had virtually no appreciation of the communications involved which actually represent at least 75% of a TV sound engineers workload—that came as quite a shock at the time'. Undaunted, his preference for this type of production has delivered him to the Low Countries.

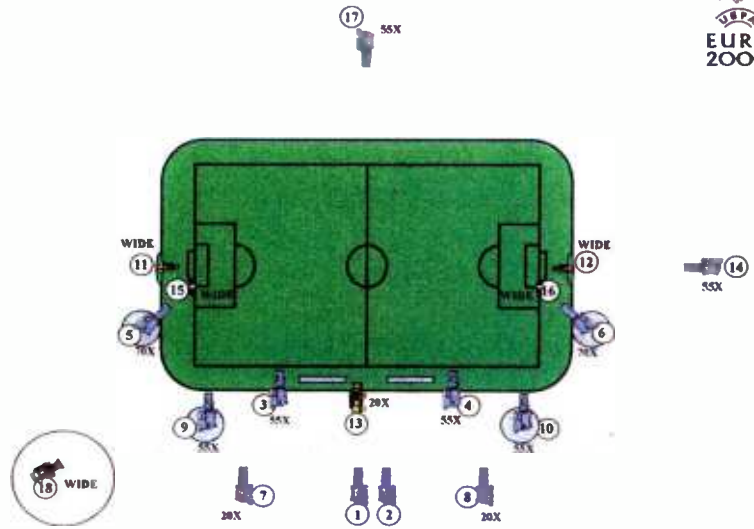
'Essentially—apart from the hype—this will not be any different as an OB to any other foreign "add on", as we call them. We are taking the host broadcasters coverage and making our own production by adding on-site studio involvement before and after the games, as well as half-time, adding commentary during the match and, hopefully, media rules permitting, getting pitchside interviews as well. I will not be putting out FX mics apart from a stereo pair of Sennheiser 416s as general ambience, which will always be safe to take since the world feed may be putting out VT Titles, interviews or even nothing at all when I need something to build the atmosphere.'

It's as well to be prepared, however, as not all host countries are as thorough as FORTO 2000 has been.

'In the past, the World Feed has varied from a mono crowd with distortion to very sophisticated pitch coverage involving manned-and-panned ball FX mics. I have no real idea what to expect this year except that generally for quality the further north in Europe you can be, the better.'

The terrestrial ITV programmes will be directed from a studio-MCR setup in the IBC in Amsterdam, but the OBs are not the only type of coverage planned: some games will be covered by commentary only via the host's commentary positions with no OB on site. Other more important games may have extra communications and links but still no OB.

'Only those games considered most important will qualify for full OB coverage and these are the games I will be



Camera positions in all stadiums

mixing—I will, of course, be taking my white gloves. For all games, however, the ITV coverage is directed from the Amsterdam Studio. The local director on site will control the OB's ITV cameras, floor managers and reporters, while on-screen anchor Des Lynham will be listening to the Amsterdam director with probably only a keyed talk from the on-site director.'

Communications therefore involves a production 4-wire between the two directors, an editorial 4-wire between the Amsterdam editor and the on-site editor (probably sat with the commentators), an engineering 4-wire for sound and a presenter's 4-wire circuit directly to Des Lynham. His mic will be permanently available in Amsterdam. In addition there will be a reverse clean-feed (mix-minus) circuit for foldback as the Amsterdam studio will take the stadium feed-in and add its own VT, which the remote OB will need to hear. All-in-all, then, quite straightforward? Butcher has his own views.

'The difficulty now is that everyone in production wants to hear these incoming circuits—except the engineering 4-wire—in case they miss something. The on-site director can now have a large number of different sources—some local, some remote—with information coming in. If the levels are not trimmed carefully, confusion and frustration will quickly become apparent. From experience, productions often forget to tell us in sound what they actually expect from the facility, although they will tell us what was wanted after it's too late or when we get it wrong. So it will be up to us to decide and implement the system we think best suited to achieve what they seem to want before they get there.'

For its fee, FORTO 2000 will provide for visiting broadcasters a commentary position with equipment, 4-wires and a music circuit in such a way that any commentator could turn up and

send back a live clean commentary match report. The British ITV will be using these circuits even when there is an OB on site as the commentary position and circuits are part of that package. This all adds to Butcher's patching-list.

'The OB will use the physical commentary position with our own equipment and then link up with the FORTO circuits, plus one or two extra of our own in order to get all the circuits we need. The music circuits will also be linked from site. Most of the communication circuits will be ISDN, so the delay problems should be minimised, but the linked music circuits will be the ones transmitted through the Amsterdam studio and there is considerable danger that the Amsterdam control room monitoring will spill over the talkback to our on-screen talent. My biggest problem may be ensuring he does not hear any delay of his own voice on the 4-wire to his earpiece.'

Obviously, the commentary positions will have cabling provided by FORTO which the OB will have to access. Again not all hosts can be relied on like FORTO 2000, as Butcher can testify.

'In similar cases in the past such cabling has been very basic—in particular unbalanced and lacking any earths—which has presented great problems with powering mics or powered talkback systems which require three separate wires per circuit. Solutions in the past have involved using three such unbalanced circuits rewired at each end to provide two balanced pairs—a process involving the rewiring of XLRs with strange looking Y-cords.'

OBs are sometimes a black art. You get the impression that Butcher probably even has a spell-book to hand though. I was a FORTO engineer for Euro 96 in England. Somehow I'm going to enjoy this game with two halves a little more for not being involved. It might even turn into a tournament of a few pints. ■

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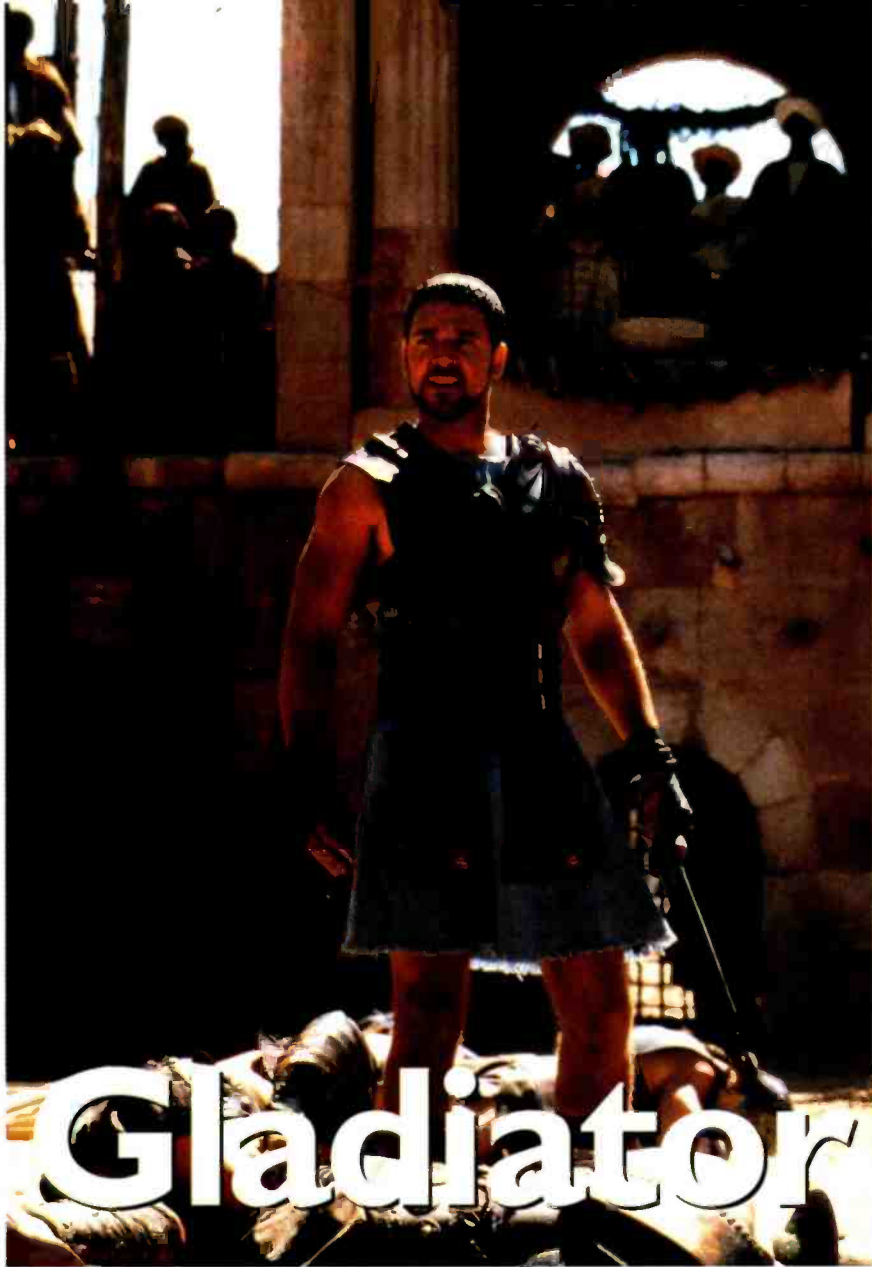
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Gladiator

With a sound brief that spanned the recreation of lost civilisations to covering the death of Oliver Reed, *Gladiator* presented many challenges. **Kevin Hilton** is in at the kill

CHARLTON HESTON once said that the epic was the easiest kind of movie to make badly—and he should know. So it is no surprise that when master craftsman Ridley Scott came to reinvent the sword, sandal and toga movie for the 21st Century, he lavished as much care and money on it as any of his other projects. *Gladiator* pays tribute to many of the epics made in the 1950s, but updates the genre, mixing heroism with modern cynicism.

Loosely based on historical fact, *Gladiator* sets out to recreate the glory of the Roman empire and the spectacle and brutality of the gladiatorial games. Attention has naturally focused on the film's visuals, which mix armies of extras, impressive models and extensive computer generated images. But the sound design is a vital component to the overall recreation of an extravagant past. When Proximo, the former gladiator turned slave trader who becomes mentor to Maximus, the story's

hero, describes the experience of walking into the Coliseum, he says the roar of the crowd rises up 'like a storm'. Such a build-up put pressure on the sound team, which had to balance very dense effects tracks with dialogue and an almost constant music score in a realistic manner.

Audio postproduction began in August 1999 at Soundelux Studios on Hollywood Boulevard after the first director's cut, with the target being a Christmas release. As Dreamworks, the main production company, in conjunction with the director's Scott Free Productions, had other movies slated for Christmas, release was delayed to the Spring. Supervising sound editor Per Hallberg comments, 'We could have made the first date, but putting it back gave us more time to work on everything.'

Hallberg's editors and the rerecording mixers were doubly blessed, firstly, by the extra time and, secondly, by Ridley Scott's demand for attention to

detail. 'Ridley wanted to make it feel as though the audience is there, especially during the Rome sequences,' observes Hallberg. 'The big issue for us was that Visual Effects did a great job and we had to match it. Rome at that time held over one million people of different languages and cultures: in some ways it was like New York in rush-hour, only 24 hours a day.'

While the capital becomes the focus, the action takes in many locations, from the horrifying opening battle in Germania to Southern Spain and Northern Africa. As each has a distinct look, they naturally have different audio characteristics.

Hallberg and his editors at Soundelux prepared the dialogue and Foley tracks on Cyberframe-Waveframe workstations, while many of the big sound effects were laid onto Digidesign ProTools. These were transferred on disk drives to Todd-AO West for the dub. The rerecording team of Scott Millan and Bob Beemer worked in two identical suites, Millan handling dialogue and music; Beemer the effects, Foley and backgrounds.

Work on temporary dubs started in late October 1999; predubs began in early November. The denseness that Hallberg speaks of can be gauged by the fact that Millan and Beemer were working with 18 8-track predubs, three of which were just of crowds, horses and chariots. As well as comprising footsteps and movement effects, the Foley tracks also included some elements of sword hits and impacts, which were considered additional to the main 'hard' effects.

All this was loaded onto Tascam MMR and MMP dubbers at 24-bit; the predubs were kept at 16-bit to allow easy translation back to the Waveframes. Most of the work was done on the MMRs, which ended up containing four stems: music, dialogue, effects one and effects two. The two effects stems enabled Beemer to change and rethink the effects without confusion: editors were on the dubbing stages during the whole process and any elements that were not predubs could be saved on the second dubber in case they were needed later.

This large number of tracks called for extra mixing capability. The two stages used at Todd-AO West, which was originally built by George Lucas as his Southern California facility and was acquired by Todd-AO around five years ago, are equipped with Otari Premier consoles. These were augmented by an Otari Status and a Harrison Series Twelve in both suites.

The task facing sound designers on historical films is creating something real despite having no real guide as to how these worlds sounded. 'I like history,' says Per Hallberg, 'and this project gave me the opportunity to pick up some good books and, by putting those together with the script, to try to get an idea of how Rome looked and felt. We

also met with the historians who consulted on the movie to get specifics like expressions people would have used.'

Overly familiar noises can be jarring in these circumstances, which usually means the use of sound-effect libraries is precluded. 'We created things from scratch,' Hallberg agrees. 'I've never worked on a movie that is as multilayered as this. The different places were created from single elements, which the rerecording mixers layered and put together. The process started with just human-voice elements, adding one voice at a time to create a city. Apart from one or two elements used in the dream-like sequences, everything is organic, natural sound.'

Hallberg says that the intention in some scenes was to create a tonality, conveying tension or foreboding through low-level or almost subliminal sound. 'It can affect the audience even though they don't realise that they're noticing it,' he says. 'Sometimes effects can be as affecting as music.'

The most obvious sound-effect sequences are the Games, which do live up to Proximo's description. 'Ridley wanted the audience in the Coliseum to be a character that participates in the action,' recalls Hallberg. The Coliseum held in the region of 50,000 people, all, according to most accounts, seized by bloodlust. Even today's football crowds would probably sound like a garden party by comparison. 'We tried using material from modern day arenas,' says Hallberg, 'but we couldn't have whistling on the soundtrack. It is common today, but in Ancient Rome they booed instead. So we had to create the size of the audience and all the specific reactions; just as a hockey audience cheers the movement of the puck, so the Coliseum crowd would cheer specific action.'

Again this was a multilayered process. A large amount of material was recorded on location in Malta, with 2000 extras in a stadium. 'That was used as a bed because it didn't have the definition or variety that we really needed,' explains Hallberg. A recording session was arranged in Los Angeles with 200 extras on an in-door sound stage to add specific reactions and chants. These were cut and worked into the main bed to give the impression of a huge crowd involved in every stroke and blow in the arena.

The problem with large numbers of people chanting or singing is that the timing goes and the result is an incoherent mess. Hallberg acknowledges this, saying that the editing process had to be precise to avoid this. ProTools was chosen for the effects because of its channel capability, which enabled three different levels of crowd noise to be created: the sound of 50,000 bloodthirsty Romans standing up, creating a roar; a medium-sized sound for variation; and closer-up reactions. 'We had these three

levels in the machine and had control all the time,' says Hallberg.

Another danger is that, with such a mass of sound as a bed, the dialogue and music will be fighting for room. Hallberg says the production sound, acquired on location by Ken Weston, was well recorded and so, relatively little ADR was necessary. The music, by Hans Zimmer and Lisa Gerrard, was mixed by Alan Meyerson and is reminiscent of *Gladiator's* epic predecessors, both in its sweeping themes and the fact that it plays through nearly the whole movie.

Scott Millan observes that the challenge was to create a sonic palate for all the scenes, so there was continuity despite the different locations and their very different natures. Bob Beemer adds that there are shifts in gear, moving between hard and soft noises. 'We were trying to make everything as realistic as possible,' says Beemer. 'This is not a hyped up Hollywood soundtrack. Ridley wanted to make sure that the battles and chariot sequences had impact but were not false. There's more of a textured approach.'

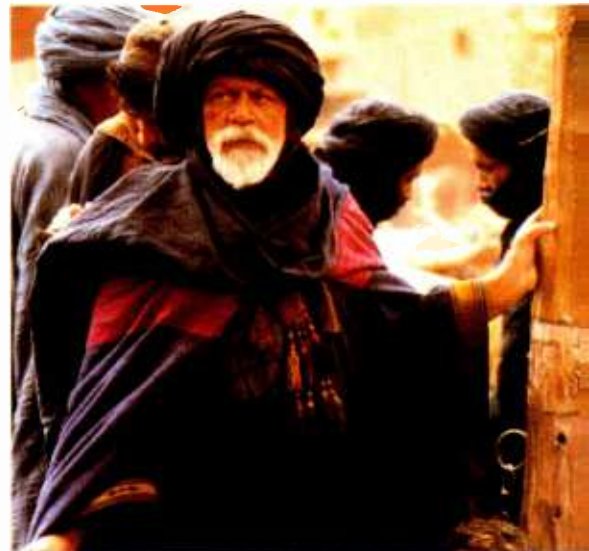
The movie was mastered in Dolby Digital (EX), DTS and SDDS. 'The rear channels were mainly used to give more separation and detail,' states Per Hallberg, 'not to make it louder. You don't need each loudspeaker to do as much work because of these discrete systems.' Bob Beemer adds that the rear channels mainly contain backgrounds, reverbs and effects, to give an impression of space. 'The surrounds are always playing on this movie, whether subtly or aggressively,' he says.

Gladiator had some unwelcome pre-publicity when Oliver Reed, who plays Proximo, died during production. The larger-than-life, hell-raising actor went as he probably would have wished, after a drinking session in an Irish bar in the Maltese capital of Valetta. While most of his principle scenes had already been shot, there were still some gaps. Much was made of CGI technology being used to put Reed's face on the body of a stand-in; little public attention was given to how any missing sound would be created.

Reed had already recorded most of his dialogue by the time he died; Hallberg says that some missing elements were 'stolen' from existing takes. 'What we really needed were reactions, like grunts and shouts,' he says. An LA voice artist who could react and sound like Reed was brought in

for this purpose. 'I think we got this guy to replace one line of dialogue,' comments Hallberg, 'but I'll let you work out which it is. There wasn't much to do, but what there was had to be very sensitively and carefully done to make it as flawless as possible. It was quite a puzzle, but it worked out well.'

Gladiator is fitting as Oliver Reed's last film. It is also a tribute to what can be achieved when the audio postproduction department is given enough time to do the job without working nightmarish overtime. As Per Hallberg says, it is the difference between being merely good and being great. ■





Mario Caldato, Jr.

From setting up the Delicious Vinyl label to refereeing the Beastie Boys arguments, Mario Caldato, Jr. has built a wealth of experience. **Richard Buskin** finds it matched by a wealth of equipment.

YOU LEARN as you go,' states Mario Caldato, Jr. 'You basically learn a lot of things by accident and you gain more experience, and as a result you learn about what to do and what to expect. I mean, collaborating is interesting. I did that a lot with The Beastie Boys. They had plenty of ideas, I had ideas, and we just worked them out.'

Whenever possible. Over the course of ten years, four albums, and numerous singles and EPs, Caldato gained plenty of kudos and experience working with the notoriously in-your-face trio of white middle-class metal-rap merchants. However, at the same time he also gained his fair share of grey hairs while supplementing his skills as a producer, engineer and remixer with those of a referee.

'That was my job,' he admits. 'I'd have to back up one guy on one idea and then back up another guy on another idea. Still, that's the work and that's what we do. We do enjoy it, but it can also be frustrating, especially when some songs are personal. One guy might want certain things, and if the other people didn't like them it would be very hard for somebody to state, "You know, we're not into that." It's a difficult thing.'

Sure is, especially when the material that does pass the litmus test still is dependent upon its creators getting their collective act together.

'I'd kind of have to push to get things done, otherwise people would just sit around,' recalls Caldato. 'You know, we were perfectly happy playing basketball at their studio... Which was a nice thing too, because we weren't pressured into playing. We only played when we'd want to. In other situations, if you don't have that time or that luxury, then it's "Alright, let's get this thing going. Get the party started". So, I'd have to make sure that we'd dive into stuff.'

And once they did dive in? Well, then it was actually plain sailing in terms of

the time, if not in terms of the conflicting opinions.

'With The Beasties we'd never spend more than a couple of days on a song,' Caldato says. 'We'd spend a couple of days and then go forward. If there was a problem we'd revisit it or remix it and then update it—like "That direction wasn't right on the vocal" and so we'd make a couple of changes and that'd be it—but there was none of this spending a week on a song or whatever. At the most it would be two days, maybe even two-and-a-half for one or two songs, but we always knew what we wanted by the end of the first day. There was a little bit of nitpicking on a few things, but we mainly knew what we were going for.'

And when they didn't, Caldato could at least divert his attention via projects with Tone Loc, Young MC, Beck, Bjork, Blur, The Cult, John Lee Hooker, Soulfly, Technotronic and Yoko Ono. Other production assignments have included the original soundtracks of movies such as *True Lies*, *Chueless*, *Bound by Honor* and *Marked for Death*.

Born in Sao Paulo, Brazil, in 1961, Mario Caldato, Jr. relocated to Los Angeles with his family a couple of years later and cut his musical teeth learning to play the organ at age four. Later on he graduated to lessons on the piano, but, shunning the classical music that he was forced to play, Caldato reverted to the organ when playing in a variety of school covers bands. One of these outfits, The Jungle Bugs, recorded the compositions of lead singer, Money Mark, at a 16-track college studio. Co-produced by Caldato and engineer Brian Foxworthy, these resulted in a self-distributed single, but not a lot else.

'Brian Foxworthy was the guy who actually taught me about drum miking and things like that,' says Caldato. 'After that, I bought some condenser mics and started recording stuff at home on my 4-track, so that gave me a little bit of

experience that I wouldn't get with the rap stuff and all of the drum loops.'

The Jungle Bugs' demise coincided with that of ska, and by the mid-eighties Caldato was hanging out in LA's nightclubs, renting out DJ equipment and taking in the sounds of go-go, rap, disco and whatever else took his fancy. It was at one such venue named Power Tools that he first encountered the pre-fame Beastie Boys, and after taking care of the sound setup there for the next 18 months, Caldato indirectly laid the foundations for their future relationship by way of a collaboration with the club's resident DJ, Matt Dike.

'Matt was getting tired of DJing and wanted to make his own records,' Caldato recounts. 'So, he got out of the nightclub with about \$10,000, and with \$5,000 of that we bought a Tascam 8-track and an [E-mul] SP12 sampling drum machine. As a DJ he had 25,000 records, so we started sampling all of them and putting stuff together. I set up a studio in his living room, and that was the beginning of Delicious Vinyl.'

'Matt's partner was Mike Ross, who was working at MCA at the time and would bring home all of these demos. Well, the first guy who they signed was Tone Loc. At first they went into a professional studio with Tone Loc and spent all of this money on a 24-track recording. Matt played it for me and I said, "Considering how much money you've spent, that sounds pretty bad. I could do better on the 4-track". As it happens, we went and bought that 8-track machine and I showed them that we could do it better on that. Matt was like, "Yeah, you're right, it does sound better. It definitely works". So, we continued recording that way and started to pile up a backlog of songs. They signed these other young artists such as Young MC and we just started developing them. We were working on all of them at the same time, using them for the vocals while Matt was in charge of

the music. That was the formula which worked, because Matt had good musical taste, and so we were making all of these songs and then finally we put out the first Tone Loc 12-inch singles.'

It was 1988, and in the wake of Tone Loc's first small successes, Caldato quit his long-held day job at an aerospace company and immersed himself full-time in PA rentals and engineering. With practically all of his \$10,000 severance pay he purchased a 16-track machine and Allen & Heath automated console, and, in his own words, 'really started recording to the next level'.

'I had my own studio in Matt's living room,' he says. 'We turned a closet into a vocal booth, and he did all of the music, I brought in records to sample and helped out a little bit with some of the co-writing, and we would sit and arrange and work on music all day, everyday, four or five different groups at a time. It was a little factory, and for me that was a great experience.'

As indeed it was for the increasingly successful Tone Loc and Young MC, as well as the production team of Mike Simpson and John King. Better known as The Dust Brothers, this likely duo contributed to the multimillion-selling 'Loc-ed After Dark' and Young MC's 'Stone Cold Rhyming', and they also had their demos recorded by Caldato. When a couple of these demos were sent to The Beastie Boys, they reacted immediately. The band members were still emerging from their well-publicised bust-up with *License to Ill* producer Rick Rubin, and The Dust Brothers' avant-garde hip-hop grooves represented a welcome change of direction. Their subsequent collaboration would result in the retro-funk-psychedelia of *Paul's Boutique*, an album that, at the start of the nineties, was largely ignored by critics and public alike, but whose cut-and-paste sampling techniques would eventually be hailed as visionary and earn a cult following.

'We spent about nine months to a year doing that record, but, for all the effort,

it still bombed,' Caldato laments. 'When they decided to make their next record, they said, "For the same money that we spent on *Paul's Boutique* we could have bought a studio." I said, "Yeah, that's a good idea" and so we went and found an old ballroom with high ceilings in Atwater Village [just north of LA] and converted it into a studio. Money Mark's carpenter cut walls and put in doors, and we put a 24-track Tascam machine and Neotek console in there, and that became G Spot... which was Son of the G Spot.'

'What with building their own studio and then starting on the new recordings, it took about two-and-a-half years to do *Check Your Head*. It was a casual, slow process making that record, but it ended up bringing them back, so in that respect it was very important. At that point I was just working with The Beasties, without The Dust Brothers. They'd had a falling-out over writers' splits and stuff on the record, and so they weren't really talking to each other. On top of that, even though the record was great, it obviously didn't do well, and so it was a weird time. Now they were starting over, and I was basically there to work with them on the new stuff, so I got more officially into the producer's role. Beforehand, I'd done some production, but now I was more in the seat with them.'

'Also, Money Mark helped coproduce *Check Your Head* quite a bit. He wrote a lot of the instrumental stuff, so he came up with a lot of ideas. We had been recording together a lot longer and he has a load of his own production techniques, so we worked on that record for over two years and it came out well and brought back the band. Then, from there I just continued working with them, and whenever I wasn't working with them I was involved in other projects, and I started to do more and more producing.'

Indeed, over the next few years, Mario Caldato, Jr would bring a number of outside projects into The Beastie Boys' >



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< studio, including those with Beck, Jakob Dylan and Jon Spencer. Yet, following The Beasties' *Ill Communication* album—which was recorded in both LA and New York in 1994—and initial sessions for its 1998 successor, *Hello Nasty*, the band members would never return to their own facility.

'Things kind of changed around and the guys were talking about moving back to New York,' Caldato recalls. 'One guy moved and then the other guy moved, and it was like, "Let's record in New York". So, I went to record with them a little in New York and then they ended up staying there.'

The band's own studio sits vacant. Nevertheless, regardless of the venue, Caldato's approach when working with The Beasties was always the same.

'We used to record everything, including the jamming,' he explains. 'I was always set up to record. There would be a bunch of mics up, and I'd be sitting at the console in the control room and always have a DAT rolling. Well, a lot of stuff that was recorded on DAT ended up being used on the records just because they captured the moment.'

The Home Setup

ASSEMBLED OVER 20 YEARS, Mario Caldato, Jr's setup is what could modestly be described as comprehensive, focusing on Pro Tools in its present incarnation.

'I have a Mix Plus with Apogee 8000, two Digi 888s, the 1622 auxiliary plug-in, a MIDI Timepiece, a SMPTE slave driver, a Pro Control, and three full racks of outboard gear, including a rack of APIs, graphics and compressors, and a rack of Telefunken V72s, Neve 1073s and a pair of Neve 1084s,' says Caldato without pausing to take a breath. 'Then, in terms of compressors, I've got Neve 2254s, 2264s, the 33609, a pair of 1176s, a pair of LA3s, a pair of DBX 160s, 162s, a pair of 165s, the Manley Variable MU and Manley LA tube, an Alan Smart compressor, an NTI Nightpro EQ...'

And this is only the gear that he can recall off the top of his head. 'There's a whole bunch of reverbs and other stuff,' he adds. Surprise, surprise...

'I've got an EMT plate, some old Farfisas,

While they were playing I would be randomly going to the echo or the reverb and kicking them on. They were messing with the songs and I was messing with effects and recording that, and as a result we captured some stuff that we could never recreate. We'd then sample those things or add onto them, and a good example was this one song 'Something's Got To Give', which was just a background track of them jamming with two PZM mics laid randomly on the ground. It captured this crazy vibe and we just added onto it, creating this whole song.

'Some of the hardcore songs were straight off the DAT mixes. Like 'Time For Living'; I recorded them rehearsing it and refined the mix, and by the time they learned the song—an hour and several takes later—I had the mix perfect. Then we just threw vocals on it. As it happens, the console had a couple of channels that were out of phase and I actually printed the instrumental tracks on one of those channels. I didn't really notice it until we went to mastering, but then the guy put it in mono and when the phase was reversed the

Furman spring reverbs, a PCM41, a PCM42, a Mutator, a Vocoder, Genelec 10 and Mackie HR824 speakers...'

And vintage mics?

'Yeah; a pair of U47s, an M49, a pair of U54s, the U87 with the tube kit, a regular U87, a C414, a Cole ribbon, an RCA 77, a pair of 451s, 460s, half a dozen 421s, 441s, the old square D12, the old D19, a bunch of SM57s...'

Still, the flavour of the moment for this avid collector has to be the Pro Tools.

'At the moment it's wonderful,' he asserts. 'I have an MCI 2-inch 24-track and 2-inch 16-track, and right now they're not set up because my room was just cluttered.'

None of which comes as that much of a shock when considering that all of this gear, racked up and ready to move, is housed within one 12ft x 12ft room.

'I have a 1/2-inch machine and a 1/4-inch,' Caldato says. 'I love analogue. My Mackie 32:8 console was genius and I mixed a lot of stuff on that, but I've just retired it because lately I've been doing everything on the Pro Tools.'

vocal disappeared. If you listen to that the song, the vocal disappears if you put it in mono. However, we just left it that way, because we couldn't change it.

'With a couple of exceptions, all of the songs on the later records were formulated in the studio. Everyone had drum machines and samplers, and they would go home, find a beat and come up with a song. They'd bring it in and say, "We just need to add vocals to it"



and I would say, "Fine, let's go for it." Boom. Then it'd be like "Oh, let's add this, let's add that"—everyone would add their two cents' worth—but the most creative guy is Ad-Rock [Adam Horovitz]. Musically, on *Hello Nasty* he was responsible for like 75% of the record. He did all of the music, and there are some songs where he actually played all of the instruments.'

Caldato is currently working on some Brazilian projects with bands such as Planet Hemp and Mundo Livre, while other assignments will simply amount to 'whatever inspires me.'

'I'm still into more of a retro sound,' he confirms. 'I don't like this modern dance stuff. I'm into the Latin, Brazilian, Afro, reggae kind of vibe. That's the sound that I love and will always love. The Latin stuff that is popular right now is lame. I'm into older cool and groovy stuff; boogaloo and that kind of thing. That's my direction, along with Brazilian samba rock. So, that's where I'm headed.' ■

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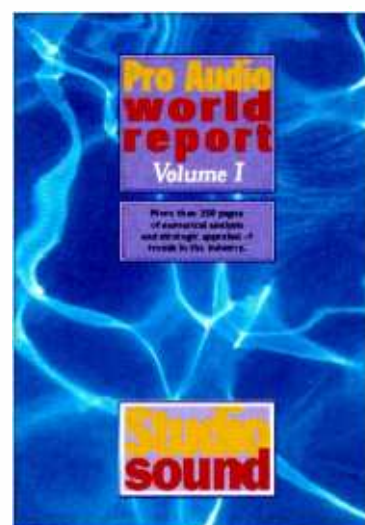
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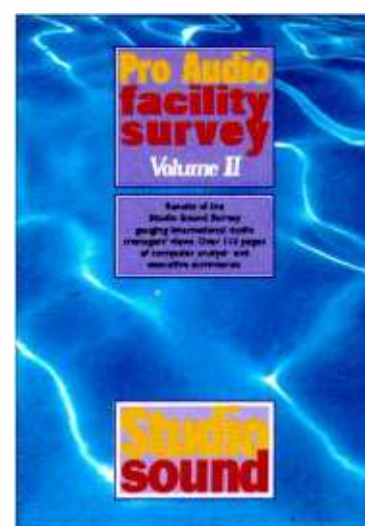


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Surround Sound Studio Design

Whether for recording, post or broadcast, the trend in studio design is for surround sound. White Mark's **David Bell** explores the issues facing anyone planning a journey beyond stereo

ALL OF THE ISSUES OF STEREO studio monitoring environments face a designer planning a room for surround use. On top of these, however, are many further considerations that need to be brought into the equation in the attempt to reconcile the conflicting requirements for acoustic treatment placement, loudspeaker mounting, monitoring position, live room and picture viewing angles, and so on. These factors are not always based within the remit of engineering design and involve such diverse influences as engineer-sound designer preference and the, sometimes unclear, targets set by some licensing bodies.

In solving these problems, a fine balance has to be drawn between the desired performance of the finished facility, the realities of the space chosen within which to build it and the requirements of the licensing bodies that the client's business demands are satisfied.

To deal with the targets for the engineering performance first, basic stereo requirements demand that monitoring environments be even in decay time with frequency and that there be no colouration of the sound by reflection-induced comb filtering or nodal problems. Imaging, too, is badly affected by early reflections and these considerations lead to the need for an area of the room in which monitoring can be deemed accurate that is free from early reflection and which is supplied with a diffuse and evenly decaying reverberant field. This is achieved by the placement of the monitors, broadband absorptive materials, diffuse wall surface elements and of the monitoring position itself. Careful consideration of these issues can result in a significant increase in the area within which good stereo imaging and tonal balance are available.

The net result of these influences is represented by the common control room configuration of stereo monitors, soffit mounted into a soft front wall that forms, together with the front part of the ceiling and side walls, a broadband absorber. The rear parts of the side walls and ceiling, together with the rear wall itself, offer the opportunity to return energy into the room in as diffuse a way

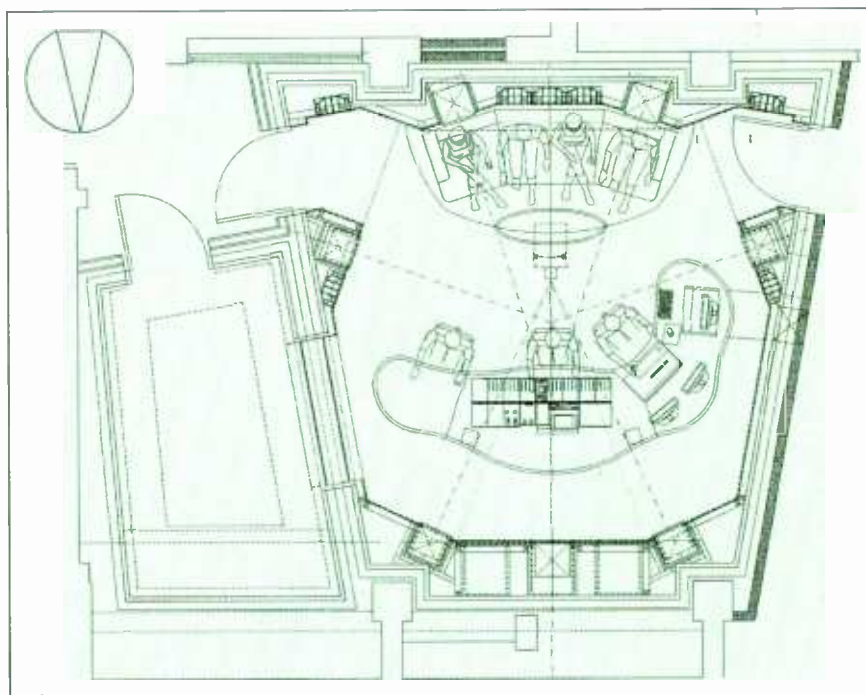
as possible in order to give the required decay time. This is often achieved with proprietary diffusing elements mounted in the rear wall and reflective side walls configured to place energy into that rear diffusing array. Rear wall bass build up can be countered by the inclusion of low frequency diffusive elements or absorption either separately or as part of the main rear wall diffusive array.

These basic requirements are complicated by the addition of the needs of surround monitoring. True 5.1 (or Dolby EX or even 7.1) monitoring needs the correct mounting of a number of monitors around the room to reproduce the three main front and a number of surround channels. Each of these channels should be given just as much consideration in respect of imaging and colouration as the main stereo pair discussed above. Additionally, the distances between the monitoring position and the monitor mounting have to be given detailed thought. The perceived tonal quality of a monitor is affected by the relative amounts of direct sound reaching the listener and of sound that originates from the reverberant field set

up in the room by the monitor. This effect is mitigated by good, neutral room design, but is still a significant consideration when laying out the monitors within a space. It is also worth considering that license related surround standards have firm requirements for the time alignment of the various channels and that these will need satisfying by a combination of the room design and the electronic preprocessing available to the engineers charged with commissioning the technical systems when the room is complete.

Further, license related, requirements dictate the level that each channel of the monitor system is capable of reproducing at the monitoring position. This, together with considerations of the size of the room, places restrictions on the choice of monitoring system and legislates towards larger monitors in order that the minimum level can be comfortably attained. Surround standards for all non-diffuse systems further suggest the use of consistent full range monitors throughout the room, or very similar matched units of only slightly reduced output capability, for the surround channels. The directivity of the units chosen is also of importance and is an issue that could easily fill an article by itself. It is, however, the engineer-sound designer, whose choice of monitor is so central to their way of working, that should be given highest priority in this critical area as it is their abilities on which the success of the studio is ultimately built.

Once the need for absorptive and diffuse treatments in the room related to the disposition of monitors is recognised, together with the need for correct soffit mounting of the monitors themselves, it seems almost unreason- >



Control room set out for Dolby EX surround mixing at Wild Tracks, Soho, London.

able that the room also has to have doors through which to enter and leave and windows through which to see recording areas and daylight. This, however, seems to be necessary!

The mounting of the sub-bass monitor, or 'L1' channel, is also a complex issue. The cut-off frequency for this recorded channel is set around 120Hz depending on license requirement or monitor system design considerations and, classically, this would give some freedom in the positioning of this source. This is due to the general understanding that human locational ability, at frequencies in this region of the spectrum and below, is poor and therefore the positioning of the monitor(s) producing them is not important. This is not our experience. In rooms that are using the sub-bass for surround mixing of music, it is imperative to have a centrally located unit, if there is but one driver, and a symmetrically placed pair if two are used. Any non-central low frequency image is readily located and is very distracting in use. This has led to the design of custom bass monitors to allow the central mounting of the drivers in spaces that can be integrated into the room shape without dominating the room layout. This was particularly true of Hit Factory Criteria in Miami where custom cabinets were designed by Matt Dobson for inclusion into the room structure without disrupting major sight lines into the studio live rooms in both Studio A and Studio E. These had to be particularly high performance units to match the characteristics of the main monitoring system and, as space was limited, the required specification could not be met with commercially available items.

Additional bass problems are encountered when trying to satisfy the requirements of surround postproduction control rooms. The need to accurately simulate the performance of large arrays of bass loudspeakers in a cinema system in a small monitoring environment leads to even greater problems of level and perception of source location. This

has been solved in a number of White Mark rooms by Matt Dobson who has combined custom bass monitor design with the distribution of sub bass information over a number of the full frequency monitors in order to produce the required combination of level and diffuse source characteristics for realistic mixing and quality assurance.

These requirements can best be achieved by using the full range of design skills of both the room designer and the designer of the monitoring system. It has resulted in a number of monitor systems being managed by serially controlled digital crossover units. The primary function of these elements is the management of both signal combination and time delays such that the physical placement of custom and commercially available monitor units within a room can be optimised to accurately produce the responses that both the client and licensing companies require. In a number of the Soho postproduction rooms, for which we have been jointly responsible with Matt for the monitor configuration, there is addition of the sub-bass information into the surround and main monitor channels to overcome the localisation problems that exist, particularly in smaller monitoring environments.

Similarly, delays and levels can be optimised for all channels in respect of the main monitor position and alternative setups can be offered where delays and levels are optimised for other monitor positions within the room. This facility is also operational in a number of our rooms in Soho.

The physical size of the monitoring environment has a number of further repercussions on the disposition of monitors for postproduction. The width of the screen for video projection is fixed by the lens configuration on available projectors and by the generally accepted maximum for screen size of 1.33x viewing distance. This is in contradiction to the angular separation needed for good stereo panning in a

room if the music and effects placement are going to be realistic and the loudspeakers are to be behind the outer edges of the picture. The issue of panning within the picture is difficult to address given many of the finite room sizes available and a realistic approach has to be taken when discussing this with the licensing bodies.

Further issues arise when allowing for the requirements for client seating, particularly in postproduction rooms. Engineer-sound designer preference for seating behind or in front of the mixing position (or both), coupled with the considerations of equal monitor distance discussed above, tend to leave the mixing position centrally situated in the room. This places fundamental importance on the design with respect to the modal response of the room. The mix engineer being centrally positioned longitudinally by these requirements, combined with the lateral centrality imposed by considerations of stereo symmetry and vertically by the limited headroom so often imposed by Soho buildings, places the control of room mode distribution high on the list of initial design issues to be addressed. The central positioning of the mixer also complicates the creation of the reflection free zone due to the greater distance between the engineer and the front monitors.

A final major consideration is the dual nature of many control rooms in the need to both record performances and subsequently mix them. The former usually takes place in stereo (certainly for voice-over work in postproduction this is overwhelmingly the case) and the latter either in stereo or in surround. A paramount requirement for good performance creation is the eye contact between the engineer and the performing artist, be this a musical or voice-over performance. This can also be expanded to include the producer and others, once again particularly in voice-over situations. Thus the positioning of the booth and the communication window must be such to allow this to >

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< happen for all seating areas involved. Principally, this concerns the engineer and those seated close to them but can also involve a number of other people. This reinforces the usual preference for a front window in the control room.

The requirements for surround mixing to picture, however, would legislate towards three monitors across the front of the room at comfortable listening height. Acoustic considerations would have this area soft and absorptive and licensing requirements usually decree that a screen of considerable size be located for central projection and correct sound placement of monitors within the picture.

These two requirements are fundamentally at odds with one another. One solution is to have a side positioned booth. This has the advantage of excellent sight lines due to the closeness of the engineering position and the performer(s) but it becomes difficult to include more people within this closeness. A large reflective piece of glass mounted in the side of the room risks lateral reflections and complicates the positioning of the hard, reflective and all too necessary access door, further than normal. Side windows can, however, be very successfully used, but need careful design to avoid reflective and modal problems. They are not to all sound designer's taste and therefore they are not suitable for all applications.

A second solution is to have a central front window and either mount the monitors above it, employing a motorised lowering screen or mount the stereo monitors at the comfortable level either side of the screen and raise, lower or carry in the centre monitor as appropriate. The former layout meets with difficulty due to the incorrect vertical placement of the acoustic image within the picture and some engineer's dislike of listening to high-mounted monitors, and the second falls foul of difficulties of correct and consistent mounting of the centre loudspeaker. White Mark has worked on a number of solutions which

involve motorised centre monitors. Solutions have also involved the moving of the critical mid and high frequency drive elements of the centre channel together with an integral acoustic baffle and separate fixed mounting of the low frequency element. This work was again carried out with Matt Dobson and again involved the design of custom monitors for the correct solution to be engineered to the client's satisfaction. The window, once again, is not optimally placed when trying to create the reflection-free zone necessary for good monitoring.

Licence compliance adds a further layer to all of the above considerations. While all of the above engineering considerations are necessary for good studio design practice, the business needs of the studio community also require the acceptance of the completed studio by a number of licensing bodies and their subsequent issue of approval certification.

It can be seen that the engineering difficulties that apply to surround monitoring environments are already many. Studio designers must strive to balance the aspirations of the studio design with the realities of the building in which the facility is to be housed. They must accommodate the equipment and layout preferences of the resident engineers and sound designers (often born of many years of experience in the field) and the need for accurate and true monitoring for a number of seating positions within the room. Above all the client must be given all the help that is possible in producing a studio that functions accurately and can be built to meet known performance criteria and within a controllable and predetermined budget and time scale.

Many of the criteria produced for studio design by the surround standardisation companies are simple and easy to design into a real studio proposal. Where these are expressed as performance specifications that can be aspired to and then measured and quantified,

real engineering effort can be applied to their achievement. It is in this respect that there is a responsibility on the part of those that seek to standardise the design of monitoring environments to make them simple, clear and realisable. White Mark has designed rooms based on commercially available monitors from Genelec, DynaudioAcoustics, Spondor, PMC, Augsburg, Boxer, Tannoy and others, but has also seen the need to employ custom designs where the specific needs of the project demanded it. We have also had good experience of liaison with Dolby engineers on specific project problems and their solution.

The freedom to adopt any engineering solution to a studio design problem must be maintained. The success of a studio design must be based on the quantifiable nature of openly published design parameters that have to be met and must be assessable against measurable performance criteria in the finished facility. Any limitation on innovations adopted in the design of rooms should be based on best design and performance practice only or the attempt to raise standards will result in a decrease in performance to an easily attainable common level.

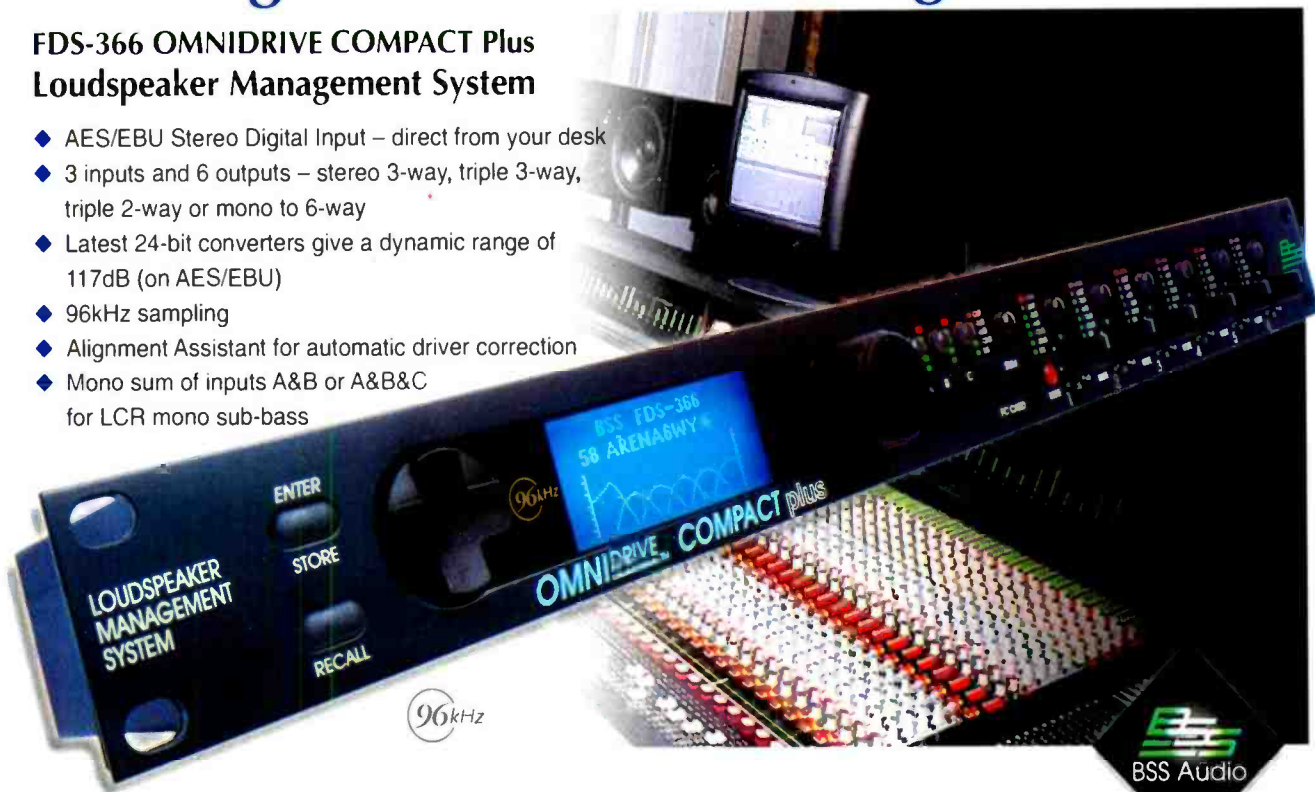
A number of music studios designed by White Mark have been brought to completion on both sides of the Atlantic. These have been created without the need to satisfy stringent standards imposed externally and have had the freedom to develop to a level judged by their clients only on the quality of the product that leaves on tape. The quality of surround sound output that is produced in such studios as Strongroom or The Hit Factory is truly spectacular. While it is acknowledged that the desire for a reproducible standard of room performance from the studio to the cinema is a commendable one, it should not come to pass that the standard is reduced for those that aspire to excellence in order to accommodate the many who don't. ■



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The Grand Design

Obscured by the ambiguity of its title, sound design offers endless opportunities for creativity and resourcefulness. **Rob James** hears pictures

SOUND DESIGN MIGHT BE simply defined as: the creation of a coherent and consistent overall sound style for a project and the recording, creation and/or transformation of sounds to this end. I am well aware there are many other definitions but this will do for the purposes of this article.

In film circles, the term came into more general use after *Apocalypse Now* although I suspect use in live theatre predates this. As Tom Holman points out in his essential book, *Sound for Film and Television*, the title 'Sound Designer' can be an emotive one. The term means very different things to different people. Certainly in the film world it is wise to check the local conventions before describing oneself thus. Depending on the company you are in you might well be taken for a self-aggrandising upstart or be recognised for your breadth of skills and experience. Today, many Universities and Film Schools offer courses, often post graduate, in 'Sound Design for the Moving Image' or similar which has helped the term gain currency. If the job descriptions Sound Supervisor, Supervising Sound Editor and Dubbing (Rerecording) Mixer are used in their strictest sense it is possible to find all three credits on the same production although it is now common for two or more of these roles to be combined. Strictly, a Sound Supervisor is responsible for the whole sound process for a movie including

location recording.

Another possible source of the title Sound Designer comes from the traditional way in which tracks or units are prepared for the dubbing stage. Working with mag film the Supervising Sound Editor may well draw on cue sheets the blocks of sounds they require in the positions they should be layed up. These skeleton cue sheets are handed to other dubbing editors in the team to do the practical work.

There has always been a degree of (mostly) creative tension between the Sound Editing department and the dubbing (rerecording) mixers. The mixers were always tempted to suspect the editors of making their lives almost impossible due to incompetence or even malice aforethought and editors always had a suspicion the mixers were not extracting maximum value from their offerings for similar reasons. Both also sometimes harboured the idea that, given the chance, they could do better than the other guy.

As a happy side effect of changing technology the demarcation lines between these roles is becoming increasingly blurred even in Hollywood. Editing now involves a good deal of mixing and processing and mixers use editing tools to deal with problems on the mixing stage.

Walter Murch (*Apocalypse Now*, *The English Patient*) and Gary Rydstrom (*Star Wars*, *Toy Story 2*) are high-profile exam-

ples of real crossover, but there are many others. In the UK Adrian Rhodes epitomises the breed. A successful sound designer with awards for *Wallace and Grommit*, lead mixer credits on *The Avengers* and *The Full Monty* he was also effects editor on *Tomorrow Never Dies*.

In television, advertising and the burgeoning computer games market this blend of skills is rapidly becoming the norm.

It is probably fair to make a distinction between projects which simply require a competent craft job and those which demand serious design input. The difference, if you like, between *A Bug's Life* or *The Matrix* and *As Good As It Gets*. All excellent tracks, but the design element is more clearly visible in the former pair. Films like these require the creation of many of the sound elements in addition to the overall mood and style. Between these extremes lie the majority of projects that require some creative design in an otherwise 'straight' piece.

From my own experience, the late seventies and early eighties were an exciting time to be involved in sound design—almost as exciting as now. Much of the excitement came from the discovery, almost daily, of new devices and techniques which could be pressed into service. Nowadays, the excitement is due to the plethora of creative possibilities available to the widest possible user base.

In the cold, hard, commercial world it is important to remember there is often a trade off between speed and cost. There are often many paths to the same end but some paths are shorter. Where time is money speed is often more critical than the cost of the kit. This is really good news for everybody. Expensive hardware and software tools have their place in sound design and so do surprisingly economical ones.

The great delight of sound design is that anything goes. This is not to imply a laissez faire attitude, it simply means anything and everything which makes a noise is fair game. The most successful sound effects often have their roots in real-world recordings of things which have nothing to do with what they are used to represent. This is especially true of the human voice. Suitably bent, voice can contribute to a vast range of effects from a gentle breeze to an exploding volcano. Location recording is a big part of sound design and it is not surprising that much of the interest and not a few of the developments in multichannel recording techniques come from this area. Contact mics like the C-Ducer range also have a part to play. Attached to a bit of old iron a small pile of scrap can become a menacing cacophony of destruction.

To understand the impact of technology on sound design over the last 25 years it is worth remembering some >

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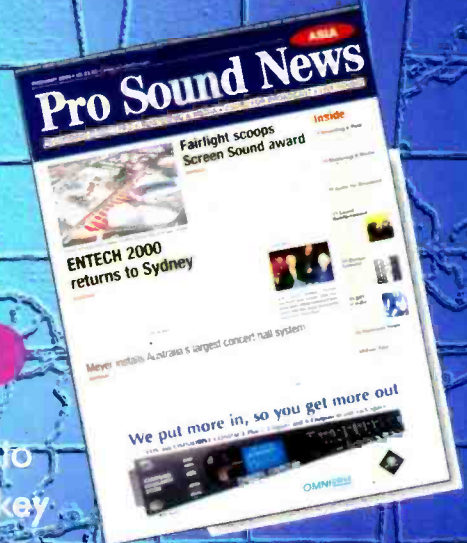
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< of the older techniques. For example, background atmospheres were frequently and laboriously constructed by editing suitable sections from a recording on film into a loop. Without recourse to using expensive dubbing theatre time there was no way of crossfading the edits, restricting the number of possible edit points. The loop was either played in at the mix or transferred to continuous film and tracklaid. Quality was frequently sub-standard since the edits, especially on 16mm film, gave rise to drop outs and the loops soon became worn. The whole process was time consuming and expensive in stock usage and manpower. The same applies to many other effects we take for granted now.

Pitchshifting could be achieved by speeding up or slowing down tapes. There was even a 1/4-inch timestretch machine, the EMI Eltro, that used a rotating head drum to maintain constant pitch while the linear speed and thus length of the recording was varied. Flutter echo was sometimes generated by using two, 3-head 1/4-inch machines with the tape running from the feed spool of one to the take up of the second, passing over both sets of heads. If the first machine records and some of the output from both replays is folded back to the input echo results. The timing could be varied to some extent by changing the tape speed and physically moving the machines. The effect resembled a sort of giant WEM Copycat. Then as now, designers relied heavily on recordings of real world sounds modified edited and mixed in unusual ways.

The range of freely available tools began to expand dramatically when the early samplers arrived in the late seventies. The Fairlight CMI and New England Digital Synclavier were stunningly expensive when they first appeared but they fired the imagination. The demonstrations of the CMI playing well known songs in dog barks and Max Hoskins performing feats of virtuosity with effects on the Synclavier stick in my mind. Even today many Synclaviers and CMIs are still in use in both music and sound design applications.

One of my particular favourites for rapidly creating effects was the AMS 15-80 digital delay. This was pressed into service as a rapid means of looping all manner of sounds and pitch changing or time stretching where required. The great virtue of this box is speed and simplicity of operation. A snatch of atmosphere could be grabbed retrospectively 'on the fly' and be ready to use moments later. The 15-80 could also be used to fire off spot effects. Those who took the trouble to learn to make it really dance discovered many other creative possibilities and quickly found themselves in demand.

The Eventide Harmonizers were also revolutionary. Apart from the more obvious uses, real-world sounds could

be quickly transformed into abstracts. The current Orville continues the tradition. A unique resource for the creation and manipulation of sounds. The only snag is, with all the new features, it is no longer quite as accessible as the early Harmonizers.

These two neatly illustrate a point. Other machines performed pretty much the same tricks as the 910 Harmonizer and 15-80s, but none were as quick and convenient to use. The same rule applies today. In an expensive environment such as a mixing stage, the successful machine will be the one that is easiest and quickest to use.

As smaller and cheaper samplers appeared enterprising editors and mixers were quick to experiment 'unofficially'. The early Casio samples found uses its makers had never envisaged looping background atmospheres and manipulating all manner of sound effects. The engineering hierarchy responsible for maintaining technical standards in the big studios were aghast at the paper specifications of such early devices, but creativity won the argument. Audio quality rapidly improved and as hard disk drives were added, the sampler became an even more useful tool to the designer and gave a foretaste of the workstations to come.

Akai's samplers, especially the S-1000 found their way into film-sound editing rooms much to the consternation of some of the more reactionary editors and mixers. E-mu's samplers are now particularly prized for design work although Akai, Yamaha and Roland are also well represented. All manner of synthesisers also found their place, often blurring the distinctions between music and sound design. In the early eighties I used an EDP Wasp and a Roland Juno 6 as a source of 'doom tones' and atmospheres not to mention more obviously electronic effects along with a collection of Boss pedals and a rhythm composer. I would never describe myself as a musician yet I have a music award for one sequence.

Nowadays, the choice is mind boggling. Apart from all the familiar hardware synths there are now batteries of soft synths to consider.

Some tools have appeared specifically aimed at the sound designer, however. Synchro Arts, best known for Jeff Bloom's WordFit, has a software plug-in which takes a small sample of atmosphere and resynthesises it. In theory at least, this enables atmospheres of any length to be created without hearing the repeating edit in a loop. One of the more esoteric systems promoted for sound design is Symbolic Sound's Kyma. This hardware and software system is hosted by a Mac. The massive DSP power is reflected by a premium price.

If an object-based digital processor set appeals there are more affordable options. The Windows-based Wave-

Warp software from Soundslogical allows designers to construct highly complex and unusual processors. WaveWarp V2 can also be used as a DirectX plug-in emphasising the increasing symbiosis between raw design and tracklaying. Both Kyma and WaveWarp really require at least a working knowledge of how digital processors work in order to get the best out of them. Both will repay the time spent in getting into them with access to sounds unobtainable by more conventional means.

For those without the time or inclination to get quite so far into the nuts and bolts, software-only packages such as Sonic Foundry's Sound Forge are very popular. On the Mac, Bias Peak, Deck and others offer alternatives.

If native processing is too limiting, Creamware's Pulsar and Scope systems and Soundscape's Mixtreme all have plenty to offer the sound designer as well as integrating DAW functions. By souping up the native processing power of the host PC or Mac with extra DSP horsepower serious real time synthesis and effects become less of a lottery.

As the samplers evolved into DAW's many of the more creative tools were left behind. Now development is going full circle and the integration of sampling, synthesis and sequencing with audio and video workstations will change the way we work once again.

One of the reasons for the popularity of Digidesign's Pro Tools in sound for picture work is the incorporation of many Sound Designer functions. This makes it comparatively easy to create, bend and manipulate sounds on the same platform used to tracklay the results. The vast number of plug-in tools available from various suppliers also greatly enhance the appeal. But Pro Tools does not have this area to itself, several others are snapping at its heels.

Although the sound design world is moving rapidly to adopt integrated solutions such as these, creative designers will continue to use their own favourite outboard devices. The Holy Grail is always something new.

At one end of the scale productions are becoming ever slicker and more complex. At the other, although budgets and time are inevitably squeezed, sound design is in demand. Manufacturers with an eye on this increasingly lucrative market would do well to consider the virtues of speed and simplicity and not simply concentrate on cramming in the greatest possible number of features. Some have already got the point. Lafont Audio Labs with the P-23 Telephone Simulator, BSS, notably their 901 dynamic equaliser and Dolby have all done well with products exhibiting these virtues. Others who succeed in combining innovation with ease of operation may be pleasantly surprised at the response. ■

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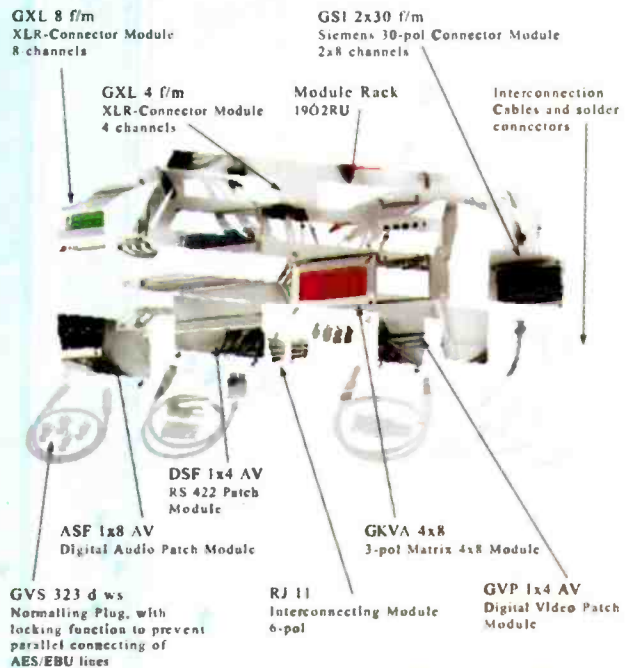
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Turning Memphis Music from a page in the history books into a brand name, and building a studio to do it, is what producer Norbert Putnam has in mind for the birthplace of rock and soul music, says **Dan Daley**

MEMPHIS ISN'T MUCH to look at. It's a hodgepodge of neighbourhoods, some fine, some funky, that follows the 'nice house, nice house-trailer, nice house-trailer, nice trailer, nice house' pattern that obtains throughout the American South. Situated at the far western end of the state, Memphis is Tennessee's wealthiest city, with an estimated 27,000 millionaires

in its population of one million. And it's one of the historical riverboat junctions along the Mississippi River, below St Louis and above Biloxi, a regular stop for Mark Twain during his early days as a riverboat pilot, before he was forced to turn to writing to make ends meet. It is one of those cities of which was said, until very recently, that it had seen better days.

On some of those better days, however, Memphis managed to change the course of musical history. It was here that an amazing and eclectic array of talent converged, over the period from the late forties through the early seventies, building on the foundation created by seminal artists like WC Handy and Bessie Smith in early part of the century, to not simply make music but to literally invent new genres—rock & roll, rockabilly, soul and R&B—with artists including Elvis Presley, Ike Turner, Otis

Redding, Johnny Cash, Jerry Lee Lewis, Roy Orbison, Carl Perkins, Rufus Thomas, Wilson Pickett, the Bar-Kays and Sam & Dave, to name a relative handful. For a combination of reasons, historical, geographical, social, racial, economic—too many factors to be anything more than serendipitously random—Memphis wound up being the birthplace of rock and soul music.

That's a sobriquet the city half-heartedly proclaimed from banners that flew above Beale Street, the thoroughfare which in the mythology of music is to R&B what Bourbon Street in New Orleans is to jazz, but which, in reality, had been until very recently a 2-blocks-long collection of rowdy clubs, gin mills and T-shirt shops. The soaring stock market has pulled everything in America upscale, whether it wanted to go there or not, and Beale Street has been propelled along towards

a semblance of propriety in the last few years, thanks to corporate sponsorships that have funded everything from street cleaning and more music festivals to better signs in the bars.

But things are changing even more, and the city is rediscovering the roots of its musical legacy, one that the city's elite burghers had looked down on for so long while the music moved from embryo to industrial-strength powerhouse in the last 50 years. There's nothing terribly altruistic about this sudden realisation of Memphis' place in cultural history. It is due, rather, to the realisation that 'there is gold in them thar hills'. Just last year, the new Memphis & Shelby County Music Commission was formed to help exploit the city's status as cradle to R&B, soul and rock music. And to run it, Memphis' city government brought back another native son, Jerry Schilling—who was Elvis Presley's >

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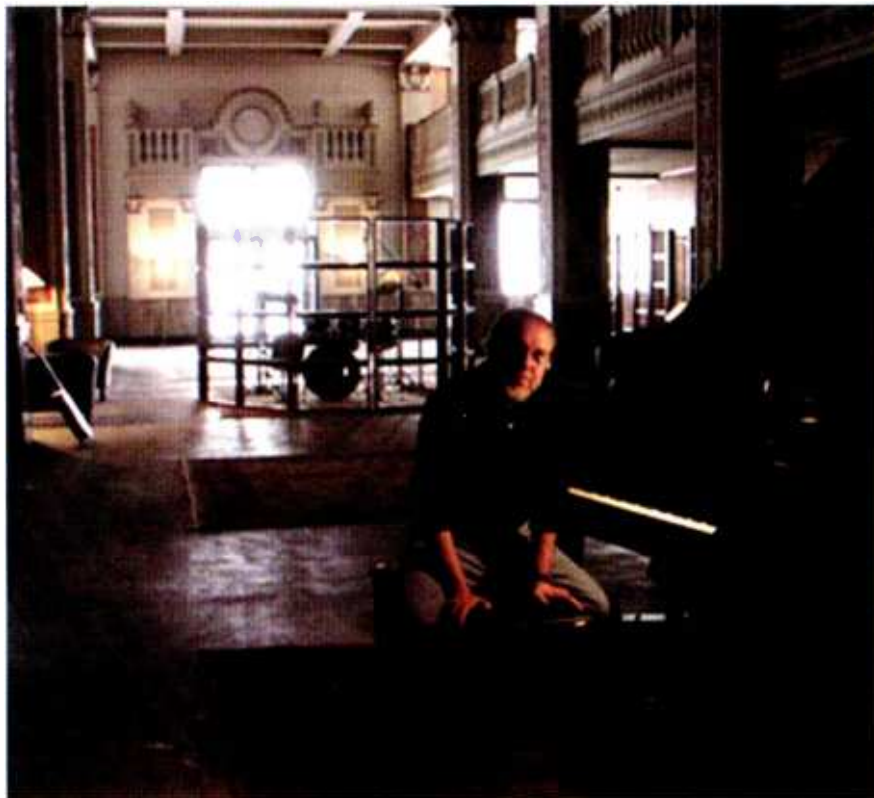
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tour manager, security director and personal confidante for much of the artist's career, and who also was the Beach Boys' personal manager for a time—as the commission's first director. (Members of the Commission's board include Knox Phillips, son of Sun Records founder Sam Phillips; and Pat Tigrett, stepmother to Hard Rock Cafe founder Isaac Tigrett.) Schilling, recalling how the Memphis establishment looked down on what it considered 'race music' decades ago, told this writer that the city now is making amends and making up for lost time.

'Elvis was not treated well by this town for years, and in return he gave us Graceland and put us on the cultural map,' says Schilling. 'It's time [Memphis] returned the favour.'

Into this milieu in which Memphis is rediscovering its roots comes Cadre

Entertainment, and its production division, Cadre Studios. Funded by a consortium of about two dozen Memphis businessmen and headed by CEO Norbert Putnam, the Muscle Shoals bassist-turned-producer who created an island of successful pop music in the middle of Nashville in the seventies, Cadre is on a mission to leverage the linkage between Memphis and that city's music. The company's Internet label venture, cdmemphis.com, intends to sign new and classic recording artists whose work has associations with the city that regards itself as the birthplace of the blues, R&B and soul. Its roster already has vintage artists Dobie Gray, Rufus Thomas and Jerry Butler, as well as Planet Swan, a new artist and daughter of country-rockabilly singer Billy Swan ('I Can Help').

Cadre Studios, part of Cadre Enter-

tainment, is located in Memphis' downtown area, a few blocks from the Mississippi riverfront and the city's new Triple-A baseball stadium, in a bank building built in 1928. The massive main lobby serves as the tracking room. Marble walls and brass-trimmed terrazzo floors provide a reverberation decay time approaching three seconds. The main area has been left open and furnished with a few plush chairs, end tables and table lamps, giving the space a comfortable sense of intimacy despite its cavernous proportions. What had been two rows of wood-paneled offices along the side walls are now nine isolation booths, and some of the office windows have been dismantled and reassembled as a drum booth and as movable gobos. Below the main floor is the bank's 60 x 20-foot vault, which is in the process of being converted into a live echo chamber.

The 20-foot x 32-foot control room sits at the top of a marble staircase and is made from two executive offices whose dividing wall has been removed, but whose Tiffany banker lamps remain. The front of the room is curved glass, with excellent visual connections to the entire tracking space and the marble mezzanine that surrounds it. As a result, monitors—KRK 8 speakers, in this case—are free-standing rather than soffited. The control room was initially fitted with a Mackie D8b digital console, which was scheduled to be replaced with a pre-owned 36-input Neve VR board. Two other Mackie digital consoles are used in two other studio rooms of the 32,000-square-foot edifice, for editing and for use by Cadre Academy, an audio engineering school based out of the studio and run by long-time Nashville engineer Willie Pevear, who engineered many records for Jimmy Bowen in the eighties. A third floor is currently being leased to another, non-audio company, but could be eventually converted to studio facilities.

The facility is Memphis' newest in several years, a market which for two

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Cadre's investors hope to tap into a deeply infused sense of connection between the city and its music, on one hand and the enduring enthusiasm with which that music has charmed the world for the last hundred years, on the other



decades has been anchored by two major studio facilities, Ardent Studios and House of Blues Studios. (House of Blues Studio owner Gary Belz is also a small investor in Cadre Entertainment.)

As huge as the studio facility is, however, it represents but a small component in a much larger business plan formulated largely by Putnam, who after playing bass on records for Elvis Presley and many other seminal Memphis-sound artists opened Quad Studios in Nashville in 1970. He worked there and in two other studios he built subsequently in Nashville, producing hit records for Joan Baez, Dan Fogelberg and Jimmy Buffett, including Buffett's signature song, 'Magaritaville'.

Nonetheless, the studio is an important piece of the plan, and the facility's layout and technical complement reflects Putnam's long-held view that an emphasis on design and high-end technology is not the key to a successful studio. 'I'm not anti-designer, but I've always said that studio designers don't produce records,' he observes. 'I'd rather have a studio that was put together

by someone who makes records.'

Putnam notes that it has never been his intention to achieve very high marks in areas such as isolation. New double-paned windows hold some of the downtown Memphis street noise at bay, but passing trucks still make their presence felt. 'But it's not a matter of unbelievable isolation,' he stresses. 'It's a matter of creating a space that works for music. And that's what every studio I've ever built has done.'

Microphone lines are not run behind walls; rather, they run on the floor in a pair of long snakes. A grand piano and two drum kits sit close by each other, baffled only by gobos. Putnam says that necessary levels of isolation can be achieved using good microphone placement techniques. 'The whole thing is about not putting barriers between people and music,' he stresses. 'When you go back and listen to the records the artists in Memphis made years ago, it wasn't about isolation and noise floor. It was about the groove of the track and the sounds of the instruments.'

Still, Putnam has slipped some acou-

stical design into the facility, though in a very subtle manner. For instance, an armoire in the rear of the control room has a curvilinear front that diffuses sound waves, as do the purposely fixed-angle slats of the plantation shutters on the rear windows, set upwards at 45°. 'Even the bookcases in the control room act as diffusers,' he says. 'The part I really like about them, though, is that they also act as bookcases, too.'

The economic bottom line on Cadre Studios is that Putnam has decided to go with simple, cost-effective (even the VR is leased) off-the-shelf solutions for the \$1m studio's technical requirements, choosing instead to place the emphasis on the attributes of the physical space, such as its acoustical properties, spaciousness and location.

And, Putnam further stresses, the decision to site the studio where it is has everything to do with Memphis and its musical heritage. Cadre's investors hope to tap into a deeply infused sense of connection between the city and its music, on one hand, which encompasses most of America's home->

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<grown genres, and the enduring enthusiasm with which that music has charmed the world for the last hundred years, on the other. 'It's the year 2000 and people still want that music at their weddings and parties,' he observes. But while the company's website could look like one of any number of dot-com music ventures started in the last few years, Putnam says this one has the advantage of a focus on an idea that resonates globally. 'I go to Germany and I say "Memphis", and most any German will immediately think 'soul music' or make the mental and emotional connection with Stax and Sun records,' he explains. 'Memphis isn't just a place—it's a whole way of thinking about music'.

And Cadre Studios is supposed to work the same way, providing good acoustical spaces combined with good

but not necessarily amazing technology that will provide the tools for what Putnam considers the ultimate tool for making great records: talent, both of the artistic and technical types. 'The way this studio is supposed to work is that it allows us to make music the way it's always been made here, using great spaces and great microphones and great talent,' he says. 'In terms of technology, we're just picking the best there is at the best price-point at the moment. The really great thing is that, at these prices, we can keep changing the equipment to accommodate the way the industry moves. What you can't reproduce is a studio space like this one. You can do amazing things in a room like this. It's not a matter of technology; I can use almost anything, though it's nice to have really good digital equipment so affordable now. But it really has more to do

with picking the right music, picking and placing the right microphones, and capturing the right feel and groove. That's something about making records that never changes.' ■

Memphis gets a Museum Mile

CADRE'S ROLL-OUT was serendipitous with the opening, on 28th April 2000, of the Memphis Rock 'n' Soul Museum. Housed in the Gibson Guitar Building half a block south of Beale Street, the museum is on the second floor of a huge atrium space, and provides social, racial, cultural and technological contexts for the development of rock & roll and soul music. The last takes the form of a series of vintage jukeboxes, from which visitors can make virtual selections which are played back on headphone-equipped individual tour guide CD/CD-ROM players, which hold 100 songs ranging from what many consider to be the first rock & roll song ever recorded—Jackie Brenston and the Delta Cats' 'Rocket 88', recorded in 1951 at Memphis Recording Services—to classics from Sam & Dave, Otis Redding, Al Green and the Bar-Kays.

The museum has artefacts galore, including several items from Sam Phillips' original facility, Memphis Recording Service, the forerunner to Sun Studios, as well as record players, Gramophones, radios and other audio artefacts from various eras of Memphis music's evolution. They are arranged in a sequence that follows the course of the conditions which led to what has come to be known as Memphis music, including the sharecropper system that put a lot of really bored yet talented people onto the front porches of shacks throughout the South, creating a rudimentary yet effective network to distribute music, and examples of the Jim Crow era, when 'separate but equal' facilities—from bathrooms to building entrances and railway cars—were established by law in Southern states to separate blacks and whites, creating flourishing musical ghettos on both sides of the racial divide.

John Meehan, senior producer for Smithsonian Productions, says the CD tour guide was developed by Bay Area-based Antenna Productions, and was intended to create as deep an immersion as possible into the milieu that created rock and soul music. 'This has been part of an ongoing series of research and exhibits that the Smithsonian has been doing on American music,' Meehan explains. 'But since this is about the music, it was our intention from the start to make sure it sounded as good as it could.'

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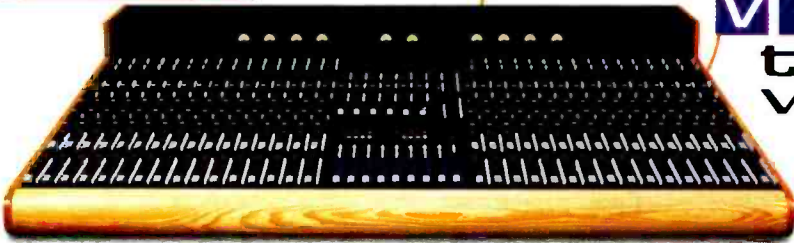
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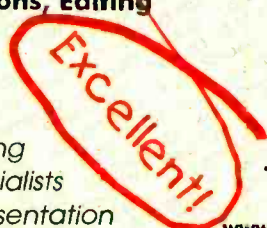
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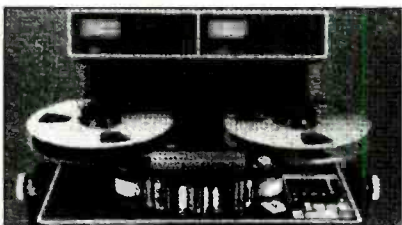


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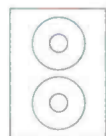
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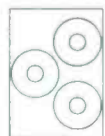
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Moving mountains

Alongside similar moves in a variety of industries, world-wide studio tie-ups are shaping to tomorrow's studio business writes **Dan Daley**

GEORGE ZIPF was not an engineer or a producer. He didn't own a recording studio, or design one, or manufacture a piece of audio gear. But he did discover a mathematically interesting axiom a while back, and that's what gets him into this magazine. Zipf was a linguistics professor at Harvard University who, in the 1930s, found that while a few words are used constantly, most are used only rarely. Further, this pattern can be described mathematically by what's known as power law: in English, the most common word is used 100 times as often as the hundredth most common word.

Zipf's law, as it turns out, can be used to describe all sorts of natural and social phenomena, including city sizes and earthquakes. Xerox researchers have also found that it aptly describes the way people use the Internet: A huge number of websites have very few users, while a few websites have a huge number of users. Even the Internet's stock portfolio follows this model, with the vast majority of Internet market capital accounted for by only a handful of companies. Zipf's law is broadly applicable across a wide range of possible areas. But could it apply to the studio market? The answer might well be yes, if the trend towards larger, geographically diverse facilities continues.

As the world stands, virtually all conventional recording facilities occupy one fixed location, and most of the revenues for each of them are generated within their immediate vicinity. Artists and producers choose facilities based on where their colleagues need or want to work, or because they're close to where they live, or they're en route at propitious times during their travels. But if they went to two cities, they would always be in two different - often very different - studios. The technology platforms that each facility used might be similar or even exactly the same, but there would always be differences in the intangible but critical aspects of vibe and feel.

But we are beginning to see certain facilities become multi-locational. Specifically, The Hit Factory, which has a dozen studios in New York in two separate buildings, also has six more studios in Miami, as a result of its acquisition of Criteria. Quad Recording, also based in New York City, last year bought another facility in Nashville. The facility's five rooms in Manhattan were augmented by four more in Music City. And seemingly old news but which takes on a new significance in this context is the Metropolis-Sterling alliance of mastering and authoring facilities between London and New York. The emphasis on new media in the venture

made it look more like a Silicon Valley move than a pro-audio one, but with these other studios' decisions, it becomes part of a larger studio picture.

Multi-city network of recording facilities is not new. Examples can be found in the EMI network, from Abbey Road in London to Studios 301 in Sydney. And historically, major record labels in the US had networks of recording studios across the country, including Columbia and Decca, which had facilities in both New York and Nashville for many years. The fact that this is taking place once again, and how this round differs from the past, tells us a lot about the nature of the business at the moment. The moves by Hit Factory, Quad and Metropolis come at a time when the same thing is happening in other from clothing (Gap, Old Navy) to office supplies (Staples, Office Depot) to food (too numerous to mention). Projecting a brand and consolidating market share by going into new markets is the overarching business trend of the moment, and it's a page out of the larger book that the studio business is starting to read. The trend can be seen in other areas of the industry, as well; for instance, the SAE schools have been doing it this way for years.

But the strategic implications are the most interesting. Quad owner Lou Gonzales says his ambitions stop at Nashville, but he is fond of saying, 'Never say never'. Troy Germano, Hit Factory's vice president, is less coy. He readily acknowledges that the company is looking for more acquisitions in other US cities. The game plan is quite transparent and it's further underscored by the fact that the renovated Miami version

For better and for worse

We're turning the tide on watermarking but falling for the Love Bug writes **Barry Fox**

WE HAVE THE POWER of the Internet to thank for making Verance, the SDMI and their 4C testing group of Panasonic, Toshiba, Intel and IBM think again about the commercial wisdom of launching DVD-Audio with an analogue waveform audio watermarking system that has never been tested at the quad sampling rates on which Panasonic is promoting DVD-A.

I had many times warned readers of what was happening but no-one took it seriously until Verance made the mistake of not taking Tony Faulkner seriously, and effectively challenging him to spend however much time it took to air the matter on the Web. Now we have an AES session, the promise of open demonstrations so late in the chip development cycle that unfavourable reaction means scrapping the system, and the fun sight of 4C and Verance trying to persuade their tame golden ears to go public and say what they heard or didn't hear at lower rates, even though they had been promised anonymity to protect from embarrassing exposure and challenge.

DVD-Audio looks like a doomed product anyway, until the format is merged into a play-anything-round-and-shiny Universal player, along with Super-Audio CD. Ever-expanding awareness that DVD-A comes bundled with waveform-tampering will just hasten its demise.

But the Internet has its downside too, in the shape of viruses like Melissa and the Love Bug. Not recognising the risks that can lurk in any incoming binary file can cost you the content of your hard disk. Love Bug destroyed MP3 music and JPEG picture files. Shudder at the thought of what a studio network could lose.

Love Bug spread as a binary attachment to emails. Already there are many similar viruses that rely on the Visual Basic Script language used by Windows. Anti-virus software failed to trap any bug that is brand new bug and thus unrecognisable as a virus. Taking life in my hands, I downloaded Love Bug and ran checks which showed that even when my Panda anti-virus software was 100% up to date it was still not looking at VBS files.

AOL and CompuServe say they cannot

block delivery of a virus because it would amount to tampering with private mail. Quite why they can find time to email monthly 'puff' newsletters to subscribers but not find time to warn users about what they are delivering is less clear.

It is possible to go into the Windows Control Panel, Settings, Add/Remove Programs and remove the Windows Scripting Host. The PC is then prevented from ever processing another Visual Basic script until you re-install it.

Microsoft's Web site (<http://officeupdate.microsoft.com/downloaddetails/Out98sec.htm>) is now offering users of Outlook (the Windows email program that

DVD-Audio looks like a doomed product

(LoveYou used to spread itself) a free Security Update download.

Describing the update as a 'significant security enhancement' Microsoft quickly adds that, although 'this update limits certain functionality in Outlook to provide a higher level of security, it was not created to address a security vulnerability within Outlook'. Presumably this is to fend off claims from Windows users who have lost valuable files.

Forget all this complexity. Just say 'No' to binaries by email. Accept only plain text

of the studio used the same designers, acoustical materials and similar technology: a national network of facilities would allow artists to remain in a Hit Factory facility in various parts of the country as they travel on tour or on business. With members of successful bands living in disparate cities, parts can be played in several facilities all networked together, and the differences from being all in the same facility might be the same as if they occupied different iso booths during a take. Here is Zipf's law made manifest: a few studios used more often, because they are more available and fit the most fundamental needs of communication.

Offering consistency at a high technological level is also a move that could help offset the inevitable impact of systems such as Rocket Networks, which has tremendously increased its strategic alliances with audio manufacturers, including Digidesign and Euphonix as it seeks to become the AOL of the studio business. Just as people need to move out of their personal studios every now and then to use larger facilities, they will also need to do the same from their virtual personal studios on these networks.

This is not one of those momentous moments in history in which everyone can participate. But the implications for the entire industry are here. Entire paradigms are shifting, and the mountains are coming to the Mohammeds of music. It just goes to underscore the fact that, in this landscape, having a vision and some imagination are as important as having a console and microphones.

ASCII or RTF (rich text format, with some formatting tags) which cannot yet (never say never) conceal a virus.

Even after the Love Bug wakeup call I've been sent binaries by the UK government's National Health Service, its Department of Trade and Industry, ex-Philips CD-i man Bernie Luskin, Sony Computer Entertainment, Sega's PR, Microsoft's PR, Connextra, someone with 'a new solution for language teachers', Fuji and many more. In each case I send a standard reply asking for ASCII, reminding that it is the words that count, not fancy fonts and bullet points. Often the sender is unable to oblige because they do not know that cutting and pasting from Word transfers safe text, and Saving As can save as plain text. Perhaps this is why Compaq, Novell, Tivoli Systems, Texas Instruments, Electronic Arts, Motorola and British Telecom have all sent me plain text emails peppered with meaningless phrases like ’ Sun Microsystems is sending random question marks.

The best of the lot came from Internet company Alta Vista: 'ALTA VISTA PROPELS THE ‘UNSEEN’ INTERNET REVOLUTION <http://www.ffgb.presscentre.co.uk/news/release-mail.asp?ReleaseID=1810>. Alta Vista ’s new service is just ‘the tip of the iceberg’'

Never mind the quality

Trading off quality against features is the thin end of the trading accomplishment wedge for flash laments **Kevin Hilton**

IN THE BEGINNING there was technology. Everyone looked upon it and said, 'Well, it's not very good, but it is still pretty damn amazing' and so they took it to their hearts. Slowly, painfully the quality improved: it did not seem to matter much that colours could vibrate more violently than a passenger on a pillion or that considerable portions of countries were not able to get broadcast signals at all.

Maybe those were more innocent, accepting times viewers and listeners put up with snow storms, shifting signals and other inadequacies in transmission because of what they were able to experience.

In the last 15 years much has been made of the increasing sophistication of consumers: modern TV and radio transmission and the widespread adoption of CD has, apparently, made people far more aware of good quality and led to a high level of expectation. Ironically, the same time period has seen the deregulation of broadcasting and the removal of such previously written-in-stone standards as the IBA Code of Practice.

Cynics could say that because this freedom made such domestic equipment as CD players and S-VHS acceptable as broadcast standard, the benchmark was lowered. What was once a business all about aspiration, striving to reach the levels set by supposed experts, became a homogeneity: the highs and lows were less obvious; it was becoming a consistency, albeit an occasionally lacklustre one.

Consistency is not a bad thing, despite the belief that people need the peaks and troughs of life to feel alive. Consistency at least sets some kind of standard and introduces a degree of technological democracy. Change one parameter, however, and everything is thrown out of kilter. While audiences were willing to put up with poor quality because the content was compelling, there appears to be a change in attitude now.

Despite the greater capacity of new transmission and delivery formats, there is still trade-off between quality and added value features because there is not, at least at the moment, enough bandwidth to do justice to both. There has been a recent dust-up in the hi-fi/home cinema press over the multichannel sound quality of DVDs. Home Entertainment magazine commissioned a Dolby Digital versus DTS shootout, coming out in favour of Dolby encoded discs because they had more extra features (director's commentary, making-of documentaries, deleted scenes) than DTS titles, despite the latter's recognised uncompressed, superior audio quality.

Trade paper *Inside Hi-Fi* carried a typically vitriolic condemnation of these findings by the determinedly outspoken Ken Kessler: 'Most extras are a waste of time.

Even if they're worthy...they're not worth watching more than once.' Kessler concluded he was criticising bad journalism and warped priorities; Rob Lane, *HE's* senior editor, hit back by accusing Kessler of bad journalism and not reading the original piece. Kessler hit back by... and so on.

It is all a matter of priorities. Digital terrestrial television (DTT) offers new text services, but these are currently just an extension of the programme guide, with a few additional piece of information and links. While DTT has more channels, lame extras and more than occasional instability—what I call the 'digital kiss', an unpleasant smacking noise made when the signal is interrupted—call its added value into question.

More worrying is the effect newer computer and telephone orientated technologies could have. MP3 was recognised as a jump in Internet audio quality; its successors are another step on and do not have the tainting association with piracy. While the source is comparable to CD, the means of listening to it is a retrograde step: music downloading asset management company Reciprocal cites a report that conventional stereo systems are almost unknown today in US college dormitories. This conjures up images of a whole generation of music consumers listening on small, low power, low quality loudspeakers in very near-field conditions.

Then there is WAP (Wireless Application Protocol). Bluetooth made a big splash about the potential of mobile wireless reception around two years ago; since then others have latched onto the concept, leading up to such as Ericsson backing the WAP Forum. Much is claimed in WAP's name: email and web-surfing are already being seen on the new generation of mobile 'phones; the future supposedly promises both digital television and radio.

What kind of quality this will be is uncertain, but given the fact that the capability of WAP is already being called into question does not auger well. Watching analogue micro-TVs is a painful enough experience because of screen size and there is no guarantee that WAP will make the situation any better. In digital radio, the Tell Me More project is attempting to create hyperlinks from what is heard to information databases and e-shops. All well and good, but is this any better than presenters giving out record numbers and listeners going to their local store, particularly if audio quality is compromised.

Drawing attention to such issues could be considered Luddite, but we should consider our priorities. New technology has brought huge possibilities, but does that mean we have to sacrifice what has been built up over the past 50 years?

Ampex

ATR 100 Series

Tony Arnold gives tips and tweaks on a machine that is 25 years old, but has become the most sought after analogue recorder in the professional recording industry

AMPEX FOUNDER Alexander Poniatoff knew how to pick engineers. Examples of his skill include Ray Dolby, John McKnight (founder MRL Magnetic Reference Library test tapes), and product manager Frank Santucci who together with Alastair Heaslet (ATR Electronics), Robert Harshberger Jr (ATR Motors), and Roger Sleger (ATR Mechanical System), were involved in the design of the ATR.

ATR production started in 1976 and the machine was reviewed by *Studio Sound* in October 1976. There have been so many updates since then that it is impossible to include every one, so instead I'll discuss those I feel are most important. Apart from the choice of heads, if everything is in perfect working order and calibrated correctly on your ATR then I defy anyone to tell me they can hear that a certain update is missing.

Ampex hoped that Ferrite heads would mean prolonged life, but the use of them was a total disaster. If you have Ampex Ferrite heads, bin them and replace with metal laminated heads. For 1/2-inch tape there are twin-track and stereo heads and should you change from a ferrite head to a composite type, check that R34 on

the main audio cards has been changed from a 50k preset to a 100k preset. When Flux Magnetic Heads are fitted, due to the extended frequency response, you will need to remove C30 on the main audio card, and also check that R34 is a 100k preset, and change C13 to a 220pf. Do not forget to bridge Pins 1 and A on PWA 4500238 on the headblock when fitting 1/2-inch heads.

Flux Magnetics developed an ME series (Mastering Extended Response) of 2-channel 1/2-inch record and reproduce heads for 30ips operation and an MS series (Mastering Standard) of record and reproduce heads built to OEM specifications. Since the beginning of analogue recording, recording head designs have been compromised to meet large variations in tape types and speeds. ME series record and reproduce heads are designed for 15ips and 30ips operation without compromise and provide increased tape stability for thicker oxide tapes, increased bias and record drive head room, and 20Hz playback wavelength response at 30ips.

The ME record head generally improves the overall frequency response -25Hz to 120Hz ± 1.25 dB, 120Hz to 20kHz ± 0.5 dB, and 20kHz to 36kHz -2dB. The MS replay head extends the low-frequency response to 20Hz-120Hz ± 1 dB.

When checking your machine, we prefer to use Ampex-Quantegy Grand Master 456 even though 499 and GP9 can work at an elevated level. Remember that when the ATR was being designed it was intended to be interfaced with the 456 formulation tape that was launched just prior to the Ampex ATR 100.

The ATR replay headroom approaches +0dB and at the record head it approaches 30dB-35dB, the machine's designers recommended 370nWb/m and +6dB as the working level.

To check the head block, make sure that the indentation or hole on the Phase-Azimuth Cog is pointing to the front on each head (Fig.1). Carry out Phase-Azimuth adjustments from this setting with the highest frequency on your test tape and check other frequencies for consistency.

You can cure static clicks on your new ATR 102 1/2-inch heads when in the vicinity of digital or static equipment. For standard response or extended-response repro-heads, the DC resistance between the core and the head stack base should be less than 10 Ω . The DC resistance between the head-stack base and the bass plate (a non-anodised area), should be less than 1 Ω . Sometimes the bushing that the head hold down screws into makes poor contact with the assembly plate.

When carrying out over-bias adjustments use the appropriate frequency for the speed you are setting up for. Many engineers try to over-bias at 30ips with 10kHz, but this is not recommended. Use 20kHz for 30ips, 10kHz for 15ips and 5kHz for 7.5ips—half the speed, half the frequency, same amount of over-bias.

We have often turned up to service a machine with the complaint being that the machine levels no longer match those of the desk and this is due to misalignment of the vu I-O modules. Readjustment is documented in the manuals. (Manual 4890407-01 Page 5-40 paragraphs 5-52 to 5-54 inclusive; Manual 4890407-02 Page 5-49 paragraphs 5-57 to 5-59 inclusive; Manual 4890407-04 Page 5-14 paragraphs 5-30 to 5-32 inclusive.)

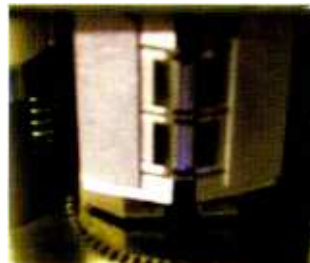


Fig.1

My dad is so enthusiastic about his new outboard device!

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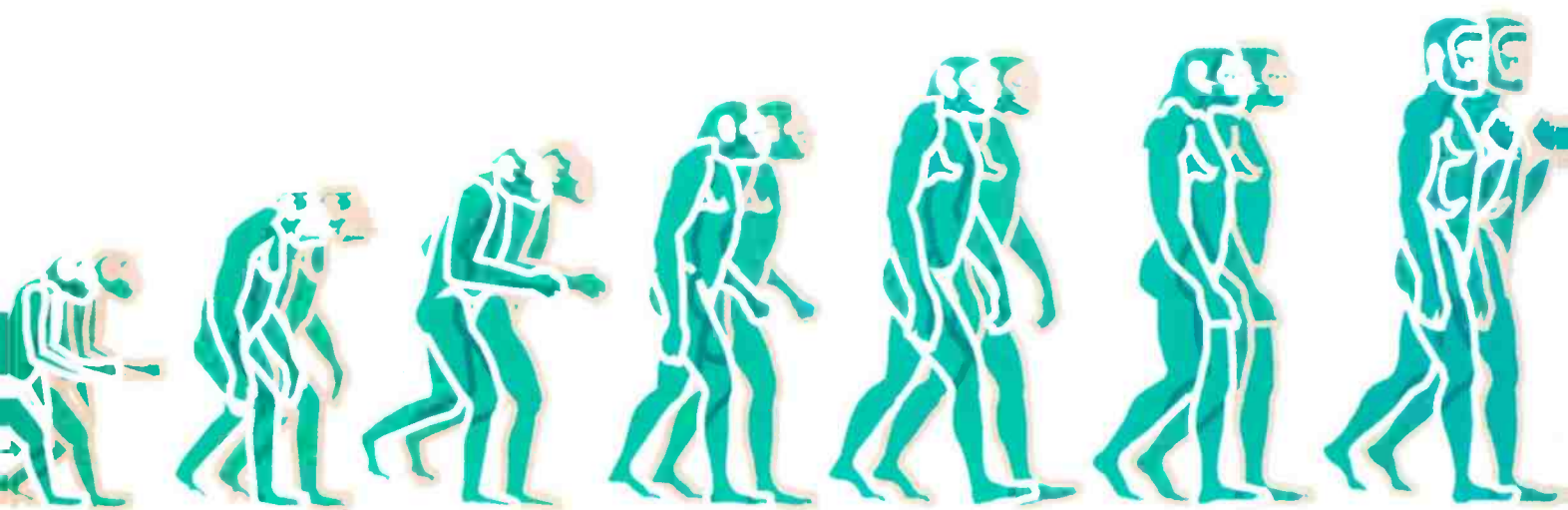
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< We are often asked why Ampex designed Symul-Sync on a stereo record head and I have always presumed that this facility can improve record-replay phase adjustment. It enables me to adjust the record head and the replay head by switching between Sync and Repro while playing the replay test tape. Therefore any interaction between the two heads can be observed.

When I finally check my record head phase while in record, I always take the frequency check up to at least 30kHz. If you really want to get the best from your ATR then I advise the use of a Flux Loop Amplifier, Part No 4020423-01.

The problem with using a test tape is that you could well be setting your levels and pre-emphasis incorrectly. If your transport and tape guides are not set up correctly

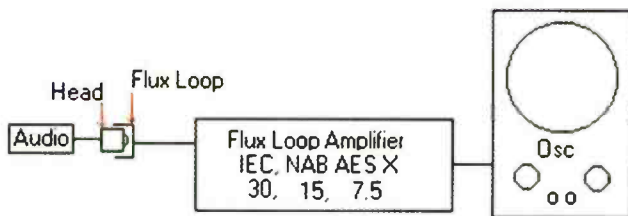


Fig.2

or your heads are worn or shot altogether then you will be calibrating your audio to counteract this problem.

An external oscillator comes with the Flux Loop Amplifier and the correct Flux Loop Part No 4050872-01 enables selected frequency's to be injected into the play head and onto the audio electronics, allowing you to calibrate without the transport running. It also does not matter whether

your replay head is in Azimuth-Phase or not. The amplifier converts your oscillator signals to choose NAB, IEC, AES at 30ips, 15ips or 7.5ips. When your calibration with this unit is complete you

can then go through the normal test tape procedure and if you find you have to alter your presets drastically then you can be sure you have a problem with transport-guides setup or your repro head requires some attention (Fig.2).

To set up the presets in the vu I-O modules without an extender card requires a 6/64-inch allen key to remove the module and a 5/32-inch allen key to remove the complete twin rack that holds the two vu I-O modules. Slacken the cable behind the rack and at the same time pull the rack out towards the front, refit the two vu I-O modules back in the rack and you will be able to see all the presets including the main R21 which adjusts the vu meter itself from the top.

Before adjusting R21 we advise that the machine should have been switched on for at least 30 minutes.

Bearings are often questioned, but should you hear a squeak when turning by hand either of the outboard motors remember that the manual states that this can be normal. However, we always remove the motor and clean the armature and brushes, making sure that the brushes go back in the original sockets in the same orientation that they were removed from. Check that the armature sockets are facing the rear, front or inside the machine but not to the outside.

If you do decide to purchase new bearings for any part of the machine, then be sure you specify EBAC 5 with the correct lubricant. ■

Footnote: In last month's Masterclass, the photographs on page 90 showing normal and lo-crosstable heads were transposed.

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FX sales

These machines are 25 years old, surely parts are no longer available? ON THE CONTRARY all parts are available 90% ex-stock including all the accessories. These include VS10 varispeed, Dolby switching PWA, cue amplifier speaker assembly, single point search to cue, auto locate (Applied Microsystems), 4-speed padnets, and pedestal trolley.

The variable fast forward-rewind speed modification on your capstan PWA is a must. This enables you to use the ordinary library wind (which some engineers find too slow) but when you go into normal Fast Forward or Rewind (which some engineers find too fast) this modification enables you to adjust the speed of the fast wind operation.



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Transforms and Spectra: 4

Continuing the transform theme, **John Watkinson** looks at two more transforms which are useful in audio; the Discrete Cosine Transform behind MP3 and the wavelet transform

AUDIO COMPRESSION, or bit-rate reduction, is becoming popular with accountants if not with listeners because it reduces costs. Reducing the amount of data needed to express an audio waveform makes sound cheaper to record or transmit. Unfortunately the performance of lossless compressors, which have no audible effect, doesn't allow a big enough reduction in bit rate. As a result most audio compressors are lossy; that is, they introduce a quality loss because the output waveform from the decoder is not identical to the original. Better coders make this error as difficult to hear as possible by 'masking'—placing it at times and/or frequencies which will most effectively be masked by the characteristics of human hearing.

As masking is highly frequency dependent, audio compressors must divide the audible spectrum into a number of sub-bands. If the error in each band has a different level, the elevation of the noise floor due to coding can be

shaped so that it corresponds to the masking curve of the ear.

Some coders, such as MPEG Layers 1 and 2, remain in the time domain by the use of sub-band filters which divide the audio spectrum up into 32 bands. However, to obtain higher compression factors, more frequencies are needed, and this suggests the use of transforms as is done in, for example MPEG Layer 3 audio coding, also known as the MP3 coding of Internet audio. Fig.1 shows how a simple transform coder works. The incoming audio is divided into blocks of samples, and these are transformed into the frequency domain. The resulting coefficients are not transmitted to full accuracy, but instead are quantised so that in return for less accuracy a saving in word length is achieved. The decoder performs a reverse transform to produce a version of the original having a higher, shaped, noise floor.

The Fourier transform is not ideal for audio coding because it outputs two coefficients for each frequency, one

representing the amount of sine wave and one representing the amount of cosine wave at that frequency. Fig.2a shows that the relationship between the two coefficients determines the phase of the signal. Fig.2b shows that if the coefficients are quantised in a compressor, it is probable that the quantising error will be different for each coefficient, leading to a phase error which is undesirable.

The DCT is a special case of a discrete Fourier transform in which the sine components of the coefficients have been eliminated leaving a single number. This is actually quite easy. Fig.3a shows a block of input samples to a transform process. By repeating the samples in a time-reversed order and performing a discrete Fourier transform on the double-length sample set a DCT

is obtained. The effect of mirroring the input waveform is to turn it into an even function whose sine coefficients are all zero. The result can be understood by considering the effect of individually transforming the input block and the reversed block.

Fig.3b shows that the phase of all the components of one block are in the opposite sense to those in the other. This means that when the components are added to give the transform of the double-length block all of the sine components cancel out, leaving only the cosine coefficients, hence the name of the transform. In practice the sine component calculation is eliminated. Another advantage is that doubling the block length by mirroring doubles the frequency resolution, so that twice as many useful coefficients are produced. In fact a DCT produces as many useful coefficients as input samples.

For simplicity, the following example of a DCT uses a very small block of only eight samples whereas a real encoder might use several hundred. Fig.3c shows the table of basis functions or wave table for an 8-point DCT.

Adding these waveforms together in different proportions will give any combination of the original eight PCM audio samples. The coefficients of the DCT simply control the proportion and polarity of each wave which is added in the inverse transform. The top wave has no modulation at all because it conveys the DC component of the block. Increasing the DC coefficient adds a constant amount to every sample in the block.

Moving to the right, the coefficients represent increasing frequencies. All of these coefficients are bipolar, where the polarity indicates whether the original waveform at that frequency was inverted. Fig.4 shows an example of an inverse transform. The DC coefficient produces a constant level throughout the sample block. Its value is zero in this example. The remaining waves in the table are AC coefficients. A zero coefficient would result in no modulation, leaving the waveform unchanged. Frequency 1 represents the lowest frequency in the transform which is half a cycle per block. A positive coefficient as shown increases the signal voltage at the left-hand side of the block while reducing it on the right, whereas a negative coefficient would do the opposite. The magnitude of the coefficient determines the amplitude of the wave which is added. The next wave has a frequency of one cycle per block. That is, the waveform

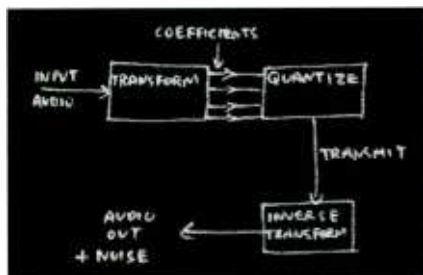


Fig.1: Lossy transform coder

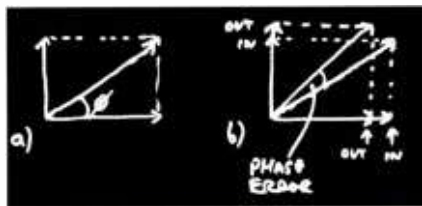


Fig.2: Quantising error in Fourier transform causes phase error

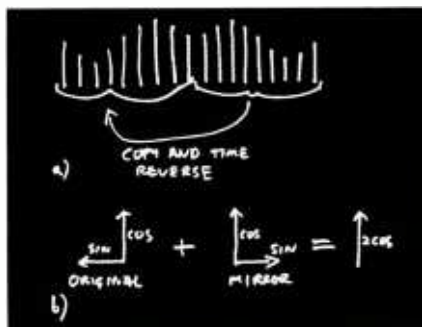


Fig.3a: Block is copied in reverse as if the samples had been reflected. 3b: Sine component is reversed in mirrored block and cancels out

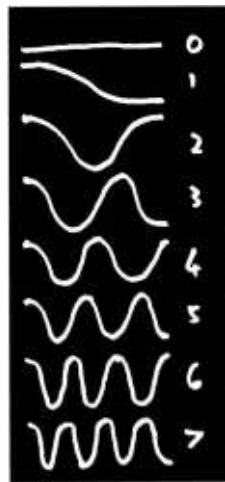


Fig.3c: The eight waves here are mixed in proportion to the coefficient values to produce the output waveform

different for each coefficient, leading to a phase error which is undesirable.

The DCT is a special case of a discrete Fourier transform in which the sine components of the coefficients have been eliminated leaving a single number. This is actually quite easy. Fig.3a shows a block of input samples to a transform process. By repeating the samples in a time-reversed order and performing a discrete Fourier transform on the double-length sample set a DCT

is made more positive at both sides and more negative in the middle. However, in this example coefficient 2 is negative so the wave is inverted. Finally there is a small positive component at F4.

Consequently an inverse DCT is no more than a process of mixing various waveforms from the wave table where the relative amplitudes and polarities of these patterns are controlled by the coefficients. The original transform is simply a mechanism that finds the coefficient amplitudes from the original PCM sample block. If you imagined an 8-channel audio mixer with each input being fed by one of the waveforms from Fig.3c, the coefficients of the DCT would be the equivalent of the outputs of the automation system that determines the position of each fader.

The DCT itself achieves no compression at all. The number of coefficients that are output always equals the number of audio samples in the block. However, in typical programme material, not

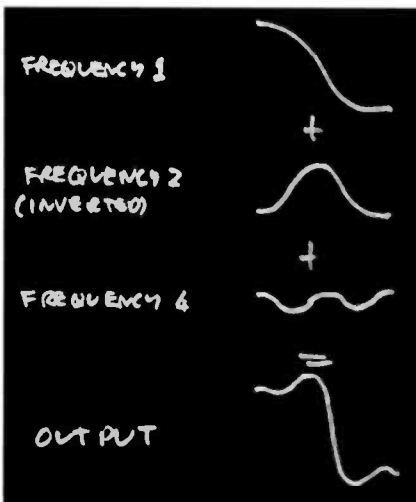


Fig.4: Adding waves from the wave table of Fig.3c

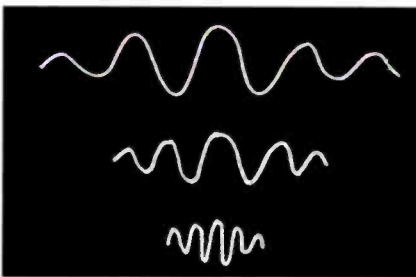


Fig.5: Wavelets contain a fixed number of cycles and so get shorter as frequency goes up

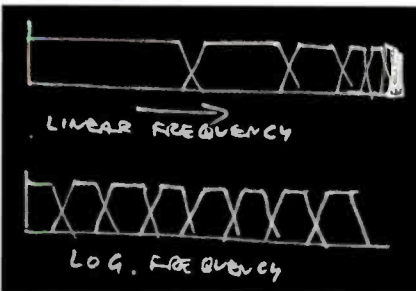


Fig.6: Frequency bands in wavelets are logarithmic or octave based

all coefficients will have significant values; there will often be a few dominant coefficients. The coefficients representing the higher frequencies will often be zero or of small value, due to the typical energy distribution of audio.

Coding gain (the technical term for reduction in the number of bits needed) is achieved by transmitting the low valued coefficients with shorter word lengths. The zero-valued coefficients need not be transmitted at all. Thus it is not the DCT which compresses the audio, it is the subsequent processing. The DCT simply expresses the audio samples in a form which makes the subsequent processing easier.

The wavelet transform was not discovered by any one individual but has evolved via a number of similar ideas and was only given a strong mathematical foundation relatively recently. It is similar to the Fourier transform in that it has basis functions of various frequencies which are multiplied by the input waveform to identify the frequencies it contains. However, the Fourier transform is based on periodic signals and endless basis functions and requires windowing. The wavelet transform is fundamentally windowed, as the basis functions employed are not endless sine waves but are finite on the time axis; hence the name. Wavelet transforms do not use a fixed window, but instead the window period is inversely proportional to the frequency being analysed. As a result a useful combination of time and frequency resolutions is obtained. High frequencies corresponding to transients in audio or edges in video are transformed with short basis functions and therefore are accurately located. Low frequencies are transformed with long basis functions which have good frequency resolution.

Fig.5 shows that a set of wavelets or basis functions can be obtained simply by scaling (stretching or shrinking) a single wavelet on the time axis. Each wavelet contains the same number of cycles such that as the frequency reduces the wavelet gets longer. Thus the frequency discrimination of the wavelet transform is a constant fraction of the signal frequency. In a filter bank such a characteristic would be described as 'constant Q'. Fig.6 shows the division of the frequency domain by a wavelet transform is logarithmic whereas in the Fourier transform the division is uniform. The logarithmic coverage is effectively dividing the frequency domain into octaves and as such parallels the frequency discrimination of human hearing.

As it is relatively recent, the wavelet transform has yet to be widely used, although it shows great promise. It has been successfully used in audio and in commercially available nonlinear video editors and in other fields such as radiology and geology. ■

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No earthly connection

Getting connected is a long-established art and one underscored in audio by ageing technology. Canford Audio's manufacturing director **Chas Kennedy** gives plugs a plug

DO I BELIEVE that most of us in the audio industry assume that technology has passed connectors by and that we give them a passing thought when they let us down? Well, yes. My opinion was reinforced recently when the computer in my car smugly indicated that I had a bulb failure in the right-hand front sidelight without considering the likelihood (nor, patently had his creator) of the connector having fallen off the rear of the light cluster, as was the reality. At least the audio industry is not alone.

New kids on the block are frequently amazed on their first visits to major broadcast installations by rows of patch panels using connectors designed in the early part of the last century, microphone connectors bigger than many of the mics they're attached to and multipin connectors that appear to be either old or over-specified. Is this an indication of an industry soaked in conservatism and served by suppliers who are happy spending nothing on development? No, it ain't.

Of course it is true that if you have a product that isn't broke, then don't fix it. However, even behind the ubiquitous XLR, for example, there is a never-ending search for improvement. In the mid seventies, in a very brave move, Neutrik entered the market, competing head on with Cannon, who by then had made the connector synonymous with its company (as Hoover had done in its field). Nonetheless by excellent engineering, and innovations like the screwless cable clamp, solder-free versions and innovative panel mounting changes, the Swiss secured substantial market share and encouraged

other market leaders such as Switchcraft to respond. What appears to be a very staid product is in fact the subject of intense engineering development.

A major problem inhibiting more radical change in connector design is of course the huge value in infrastructure, particularly in large installations. Rarely is wholesale change possible in these situations which puts great constraints on producers by maintaining systems where physical compatibility has to continue. It isn't until a completely new technology is introduced that options to change become a real possibility. Sometimes it may be, as in the case with video, technology that forces change.

The digital revolution has had a major impact on connector design and use in the audio industry. Well, it has and it hasn't. It is quite fascinating to look at some of the new solid-state portable recorders, for example. Slots for PC cards terminate in 50-way connectors where each pin is less than half a millimetre in length. Cards would routinely be tested to withstand 10,000 insertions and removals and beyond. It's high-tech stuff, but it is run of the mill nowadays. So the analogue and digital audio ports are high tech? Yeah, well XLRs are quite high-tech, but more importantly, are still an enduring standard. Although digitisation of the signal path has not forced change in the audio world, it is having a much greater impact in video.

For many years, video has been terminated using BNC connectors and patched in the UK at least on MUSA connectors. In the US, patching has been mainly on Western Electric or WECO connectors. The introduction of, first, 270Mb/s and more

recently in the US 1.49Gb/s HDTV digital TV signals put stress on existing infrastructure. In the UK the MUSA connector was of dubious impedance (around 600Ω and the early versions had fairly crude junctions with the attached BNC). In the US, similar impedance mismatches were visible and the common US practice of building 'normalising' switching into video connectors caused further impedance and crosstalk problems. It is in this situation that the desire of the major broadcasters to have physical backwards compatibility with existing infrastructure can impose serious engineering constraints. It is possible however, to make a compatible connector which much more closely approximates the desired 75Ω impedance point over a wide frequency range.

It's inevitable that increasingly we will see much more use of fibre technology in our industry, but the need for mobile flexible interconnects will resist this change for some time. In the meantime much background effort continues to go into development of improved connectors to terminate copper.

It was in the course of testing some of those solutions recently that I experienced a great revelation. Intuitively it should have been obvious, but when you begin to concentrate on solving some of the problems associated with connectors, you can take your eye off the ball. So it belatedly became clear to me that even when near perfect connector criteria are met, the great constraint to signal transmission is the cable itself. Now, people in the industry tend to know little about cable (and care less) but that's another story. ■



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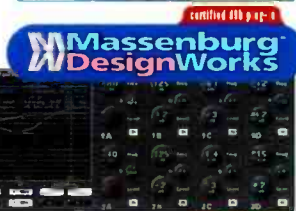
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