December 1997

THE INTERNATIONAL TECHNICAL MAGAZINE FOR PRO AUDIO, POSTPRODUCTION & BROADCAST



EXCLUSIVES

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www.prostudio.com/studiosound

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The door-stop

WHAT DO YOU do with all your old gear, at what point does it qualify finally for long overdue disposal, and where does it all go?

When an item steps out of the mainstream of usability, popularity and practicality it enters a purgatorial state of waiting where it exists for the seemingly very likely occasion when it will be again needed desperately. Audio and current do not pass through it for months until the day comes when the initiative is taken and it is employed for the first time to perform a function for which it was never designed. That springloaded heavy studio door can suddenly be stayed by the unit's mass, its physical dimensions are deemed perfect for tilting the nearfield monitors, or its castors and flattopped surface are adapted as a convenient movable *ad boc* table.

At this point the device's fate is actually sealed, although its owner hasn't spotted it yet. The device may now migrate mysteriously around a facility even appearing regularly in the reception area for brief periods where's it presence is so curious that it is never questioned. Nobody realises that it is attempting to escape in shame.

If it is lucky then it will eventually find a home with someone who appreciates it, if its not, then it will be scrap.

It's easy to get sentimental about old gear, some more so than others, but history has demonstrated that we have at times been too quick to condemn technology in the face of a younger more exciting promise. Integration frequently seems the more difficult option compared to simply starting again and while the latter is facilitated by the new product drives of the manufacturers, the retro revival in modern outboard and mics plus the increasing 'knobbage' of digital desks tells you that there was much that was right with the old.

The key question that must be asked is whether anyone will ever look back with equal affection and sentimentality on the current generation of equipment. Are we using gear that will continue to be appreciated or are they simply door-stops in waiting?

Zenon Schoepe, executive editor

The end is nigh

THE END of the year traditionally invites rationalisation. In professional circles this readily equates to television 'highlights' shows, magazine features and editorials, and achievements awards. On a more personal level, it tends towards New Year's Resolutions and just a hint of soul searching. In a handful of vocations—and I'd readily include pro-audio in this category—the distinction between the two is wont to become somewhat blurred.

For me, the major issues for 1997 include fast and wide digital audio (96kHz... 192kHz... 21-bit... 24-bit...), multichannel carriers, and the Millennium bomb. I invite you all to consider them carefully.

Certainly. *Studio Sound* has devoted space to all of these, but there are considerations that, with Christmas looming large, either have yet to be formally discussed or fall closer to the Personal Considerations category than the Let's Convene a Study Group category. Let me give you an example of the latter: while the fallout from modular digital multitracks covers the lower echelons of the professional recording industry and has settled comfortably on audio-for-video post, there is emerging a third category that offers considerable riches to the astute independent recordist. Having bought a tape-based multitrack as a cheap route into digital multitracking, a keen mind might make the correlation between the machine in the flightcase, those curious bitsplitting boxes that trade tracks for bits and the emergence of multichannel audio-only recording. Assuaged by the impending popularity of DVD, the previously modest multitrack operator could quickly become a forward-thinking sound recordist offering high-bit, discrete multichannel recording overnight. All that would really be at issue would be the bit splitter, a suitable desk and a handful of respectable mics. The career decision is yours.

Alternatively, you might like to take the year end to review the computers that facilitate your high-end recording studio: are you sure that the Millennium bomb is the stuff of hysterical news stories, confined to wreak havoc on financial institutions, air traffic control, or on nothing at all? You are assured that your accounts are safe, and there's been no word from any of the DAW manufacturers or the console automation specialists, but why are insurance companies declining to extend their policies to computer-dependent systems over the millennium end? Think carefully.

Tim Goodyer, editor



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Jan-June 1997

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4

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Inter BEE 97

Tokyo: You have to be impressed by the demeanour of Inter BEE 97. Despite its depiction as an event that belongs to Japan, it is still a heartily regional show that attracts interest from the surrounding area as a type of mini-IBC.

The IBC analogy is apt as the exhibition covers picture and audio representation housed in five halls at the massive Makahuri Messe outside of Tokyo with its phenomenal ceiling height, wide carpeted aisles that you could drive a truck down, and numerous local interpretations of snack bars. The sound contingent accounted for a single hall with all major brands present but often coupled uncomfortably together under local distributor banners.

What was interesting was the presence of Japanese manufacturers that concentrate predominantly on the home market such as Tamura, big in broadcast in Japan, with its ADX7000 and ADX5000 digitallv controlled analogue desks and Qolle with the ever-so-slightly Trident 80c-esque looking AMQ2000 digital console

Then there was Ramsa with its DX1000 digital theatre board, which serves as the technology

behind the circa \$5000 (US) WRDA7 digital desk that is expected to be revealed publicly at NAMM. Ask about the flagship product and Ramsa will tell you that it is intended exclusively for the Japanese market. Ask why and they'll tell you because it is

You're reminded of stories of Japanese manufacturers frequently possessing technology that is beyond their public product lines, and Ramsa clearly has more than it knows what to do with. TOA is another curious case in point with the fabulous looking ix-5000 and ix-11000 digital desks still making appearances, yet why has it not continued to place as it looked like it was certain to after that first prestigious installation outside of Japan in Vienna in 1991?

Yamaha showed a prototype of live-oriented GA(group/aux) a 32/12 analogue board with a matrix and 3-band EQ on the outputs. While I didn't see it myself, there were rumours that the company had also arrived at an 8-track MD personal multitracker.

Recent InterBEEs have been accused of being a little flat, but this year's show underlined its position as the country's premier audio event.

Zenon Schoepe

Austria: A 100-input Cadac J-type console—complete with the new dual-input modules-is one of the unseen stars in Roman Polanski's Dance of the Vampires, a new live production showing at The Raimund Theatre, Vienna. With music by Jim Steinman and sound design by Richard Ryan, the show uses a 35-strong cast and 25-piece orchestra, and has won critical acclaim from both the Austrian and German press. Cadac, UK. Tel: +44 1582 404202



US: The rerecording of the 20th Century Fox 'Fanfare' that accompanies the corporate logo in introducing feature films signalled great changes in California. Along with Fox's first feature-length animation, Anastasia, the Fanfare graced the newly refurbished Newman Scoring Stage and its SSL 9000j-series console, 5-channel Genelec monitoring system and Genex MO recorders. Established in 1928 as a shooting stage and converted to a sound stage a few years later, the 15,000ft² stage gave up some of its floor space to a new control room in 1975. The latest refurbishment by studio bau:ton has recovered the original space and added sight lines between the control room and stage as well as bringing the facility into the 1990s. The studio's John Kurlander, Bill Benton and Doug Botnik have takern to the the SL9000i with the Genex recorders, studio bau:ton, US. Tel: +1 213 251 9791. SSL, US. Tel: +1 213 463 4444.



SATIS

Paris: Continuing the autumn merry-go-round of exhibitions, the 15th. SATIS (Salon des Technologies de l'Image et du Son) took place during October at the Porte de Versailles in Paris.

A long-established show, SATIS has had a bit of a chequered career and seemed at one time to have difficulty in settling into its own niche but carrving on from a successful 1996, this year's show confirmed its place as the premiere French exhibition for the audio and video industries with over 250 **exhibitors**

Also firmly in their places were representatives of the film, multimedia and (to a lesser extent) photography worlds. Combined shows have always been a bit of a lottery but SATIS seems to have found the means to present an integrated show where no one section is left to one side. Certainly, even though many exhibitors showing sound equipment were in the audio section, it was by no means a ghetto and one passed easily from one discipline to the next.

Most of the equipment on show had already been presented at AES but one could note the latest developments in software for the Innova Son Sensory digital console and the digital console from Amptec (Belgium), the Stone-D 001 which features 24-bit A-D conversion, 32-bit internal and 20-bit D-A.

SATIS also features a comprehensive lecture and roundtable programme and one day was dedicated to various applications of digital audio.

These included DAT, MiniDisc and (of course!) DVD, the preservation of audio quality with data compression, comms systems and digital broadcasting from fixed and mobile sites. Radio-France also organised a daily workshop on multichannel sound with examples from the BBC, ZDF, RAI, NHK, CBS and French broadcast.

SATIS looks all set for a successful 16th event scheduled for 3-6 November.

Terry Nelson

December 1997 Studio Sound



Greece: Athens' nomination to host the 2004 Olympic Games recently witnessed a massive rally rolling through its streets. With more than 150,000 people in attendance, the selection of Greek musicians—including George Dalaras—welcomed the support of 6 Midas XL3 consoles and a serious sound reinforcement rig. Supplied by the local Alpha Sound operation, the rig included 100 EAW KF850s with 50 SB850 sub-woofers all driven by QSC PowerLight 1.8 amplifiers handling the highs and Powerlight 4.0s the mids, lows and the subs.



UK: Surrey's residential Ridge Farm Studios has included the Installation of a pair of Genelec 1039 main monitors in the refurbishment of its control room—pictured here with chief engineer, POM. The Genelecs replace a pair of Questeds on monitoring duties, while a 64-channel SSL 9000j console replaces a Neve VR. Genelec, Finland. Tel: +358 17 813311. SSL, UK. Tel: +441865 842300

Recommended reading

Among recent reading matter, three particularly useful volumes have appeared: the 6th edition of the *Tapeless Audio Directory: Wire, Cable* and *Fibre Optics for Video and Audio Engineers*; and *Higb Performance Audio Power Amplifiers*.

The first (ISBN 0-9517826-9-X) continues Sypha's series of directories of nonlinear recorder-editorplayers, with an improved listing format and coverage of machines aimed at everything from location recording and postproduction, to cart replacement and automated radio playout. Coverage includes details of technical and operation specifications, development plans, support and cost. Price \$29.95 (US), \$19.95 (UK); details from

Studio Sound December 1997

Sypha. Tel: +44 181 761 1042.

Written by Stephen Lampen, Cable and *Fibre Optics for Video and Audio Engineers* (ISBN 0-07-038134-8) offers a friendly walk through of one of the least glamourous but most critical technical aspects of all audio and video installations, complete with plentiful diagrams and graphics. Price \$34,95 (US): details from McGraw-Hill at www.ee.mcgraw-hill.com.

Ben Duncan's *High Performance Audio Power Amplifiers* (ISBN 0-7506-2629-1) presents a typically thorough technical investigation of one of the author's pet subjects, and is littered with circuit diagrams and measurement plots, and features comprehensive index, glossary and appendices. Price &40 (UK): details from Butterworth Heineman, Tel: +44 1865 314301. Sweden's NRJ radio network has installed 11 Orban Optimod 2200 broadcast processors. The largest Swedish commercial network, NRJ operates 24 transmitter sites and reaches some 70% of the Swedish population; the new Optimods are located at the main transmitter sites.

NRJ, Sweden.

Tel: +46 8 720 2525. Orban, US. Tel: +1 510 351 3500.

 Chicago Trax Recording has purchased an AMS Neve Capricorn console for its new Capricorn Room. The 72-fader, 128-input desk will work with a new Studer 48-track DASH multitrack machine with forthcoming DVD surround formats in mind.

AMS Neve, US. Tel: +1 212 965 1400/+1 818 753 8789.

■ Singapore's A String Studios has installed a 72-channel SSL SL9000j console. The Taipeibased facility will use the console for music recording and CD premastering, and joins a 64channel SL4000.

Solid State Logic, UK. Tel: +44 1865 842300.

British freelance producer Alan Branch has bought a pair of Harbeth's Xpression! DPM1 monitors for personal reference. The speakers have already seen use on Simply Red, Depeche Mode, Primal Scream and Royal Philharmonic sessions.

Harbeth Acoustics, UK. Tel: +44 1444 440955.

■ Germany's Aalto Theatre took advantage of a schedule break to install a customised 36-channel Calrec T-series console. Located in Essen, the Aalto hosts ballet, opera and theatre productions and required a purpose-built frame to enable the engineers operating the desk to see the stage. Calrec, UK.

Tei: +44 1422 842159.

■ Florida's Retrophonics Studio has purchased a Summit Audio DCL-200 valve comp-limiter, joining another local Summit user, Gloria Estefan's Crescent Moon. Retrophonics in an independent music recording studio whose credits include Don Henley, Tom Petty and Rod Stewart. The DCL-200 will see use alongside an Amek console and a collection of classic mics.

Summit Audio, US. Tel: +1 408 464 2448.

■ London's Metropolis Studios has installed a a dual-screen SA&V Octavia DAW system for its 5.1-channel surround mastering suite. The system complements Metropolis' 5 SADiE editing systems and both make use of the file interchange offered between the SA&V systems and Genex' GX8000 MO recorder. Metropolis also uses Prism Sound AD124 and DA1 convertors.

Metropolls, UK. Tel: +44

SA&V, UK. Tel: +44 1353 64888.

Russia's radio station,

Svobdnaya Russia has opened with an Audioarts R-60 console, Tascam BR20 analogue recorders, beyerdynamic M300 mics, Telos telephone hybrids and Tannoy System 800 and PBM 6.5II monitors. An accompanying production studio boasts a Soundtracs Topaz console, Tascam BR20s and Tannoy System 12 DMTII monitors. The installation was coordinated by MS-Max in Moscow.

MS-Max, Russia. Tel: +7 095 234 0006

The British Twickenham Film Studios facility has followed many US West Coast film studios in installing a Bag End ELF subwoofer system. The first feature film through the new system was Seven Years in Tibet, which is currently enjoying critical acclaim. **De Regle Group.**

Tel: +44 1945 463077.

■ New York's Howard Schwartz Recording has completed work on its Studio 8 featuring a new Harrison Series Twelve console. Occupying a prestige slot in NY's post scene, the fully-automated, digitally-controlled analogue desk is 5.1 surround equipped and was installed with a 200-foot run between the patchbay and the machine room.

Harrison, US.

Tel: +1 615 370 9001. ■ Baden-Baden TV studio

Südwestfunk has ordered a Lawo MC82 digital desk. The 48-fader MC82 will replace the facility's SSL consoles and offers 5-channel and 8-channel output busing. Installation is set for June 1998. Lawo, Germany.

Tel: +49 7222 10020.

■ New London radio station XFM has gone on air with two Audionics ACE mkIV desks.

Audionics, UK. Tel: +44 114 242 2333.

Tascam's MMR-8 handled much of the audio program during postproduction of Francis Ford Coppola's new John Grisham film The Rainmaker. The mix was headed by postproduction supervisor and Zoetrope VP Kim Aubrey, supervising sound editor Michael Kirchberger and picture editor Barry Malkin. At the mixing console was four time Academy Award winner Mark Berger, assisted by Michael Semanick. Coppola's American Zoetrope Mix facility in Napa Valley shares a turn-of-the-century carriage house with the Zoetrope research library and the Niebaum-Coppola Estate winery.





February 1998 5–6

Live! 98

The Roundhouse, London, UK. Contact: Justine Smart. Tel: +44 1322 660070. Fax: +44 1322 615636.

17-19 Integrated Communications 98

Level 2, Olympia 2, London. Contact: Turret RAI. Tel: +44 1895 454438. Fax: +44 1895 454588. Email: 100730.1313@ compuserve.com

March 11–15

Musikmesse

Frankfurt, Germany. Contact: Anke Witte. Tel: +49 69 7575 6596

16-19

Technology India 98 Bombay Exhibition Centre, Mumbai (Bombay). India. Contact: Above & Beyond Exhibitons. Tel: +91 11 651 0205. Email: vikas.gulaty@gems.vsnl.net.in



'Designing

microphones at

Danish Pro Audio.'

Ritz Carlton, Laguna Niguel, Dana Point, California. Tel: +609 279 1700. Fax: +609 279 1999.

23-27

4th BTV China 98 and 3rd COMMTEL China 98 Contact: Business & Industrial Trade Fairs. Tel: +852 2865 2633. Fax: +852 2866 1770.

April 14–16

PLASA: Light and Sound Shanghai Intex Centre, Shanghai. China. Contact: P&O Events. Tel: +44 171 370 8837. Email: shanghai@eco.co.uk

May

16-19

104th AES Convention RAI Conference Centre Amsterdam, The Netherlands. Tel-Fax: +31 35 541 1892. Email: 104th-chairman@aes.org. Net: www.aes.org

18-20

Cable & Satellite 98 Earls Court 2, London, UK. Net: www.cabsat.co.uk

26-28 TV 98 Thermal Hotel Helia. Budapest Hungary. Tel: +361 153 1027. Fax: +361 153 0451. Email: hiradastechnika@mtesz.hu Net: www.mtesz.hu/ hiradastechnika

26-29

Midem Asia 1998 Nusa Dua Beach Resort, Bali Tel: +331 41 90 46 31. Net: www.midem.com

29-31

5th Annual Latin-American Pro Audio & Music Expo Miami 98 Miami Convention Centre, Miami, Florida, US. Contact: Studio Sound International. Email: chris@ssiexpos.com Net: ssiexpos.com

June 2–5

5th Broadcast Asia 98 and others World Trade Centre. Singapore. Contact: Overseas Exhibition Services. Tel: +44 171 486 1951. Fax: +44 171 413 8211. Email: singex@montnet.com

Net: www.montnet.com July 17–19 Pro Audio and Light World Trade Centre, Singapore. Contact: IIR Exhibitions. Tel: +65 227 0688. Fax: +65 227 0913. Email: ruohyi@iirx.com.sg Net: www.mediav.com.sg/pala

September 26–29

105th AES Convention Moscone. Convention Centre, San Francisco. California. US. Tel: +1 415 558 0200. Fax: +1 415 558 0144. Email: 105th-chairman@aes.org Net: www.aes.org

October 12– November 6

ITU Plenipotentiary Conference

Minneapolis. Minnesota, US. Tel: +41 22 730 5969.

November 4–5

22nd Sound Broadcasting Equipment Show (SBES)

Hall 7. National Exhibition Centre, Birmingham, UK. Tel: +44 1491 838575. Fax: +44 1491 832575. Email: dmcv@pointproms.co.uk Net: www.i-way.co.uk/~dmcv/ sbes.htm

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'You used to design microphones for Brüel & Kjær. What are you doing now?'

Morten Siøve and Lestablished Danish Pro Audio and created a business venture with Brüel & Kjær. We have a single aim - to provide professional audio engineers throughout the world with professional microphone solutions. It is a simple philosophy, but one

When I left Brüel & Kjær in 1992, it was

to start something special. My partner

It is a simple philosophy, but one which we take seriously. It is the driving force of our development and the reason for our success.

Anyone who knows anything about microphones will confirm that Bruel & Kjær does not lend its name lightly My designs continue to combine a clear perception of professionals' needs and the technical ingenuity of which I am proud.

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www.americanradiohistorv.com



Freedom of choice

I refer to Tim Goodyer's September editorial, 'Freedom of Choice'.

We have never been complacent about our leadership in the professional audio recording industry, we are there by intention and as a result of being the only manufacturer dedicated to the needs of professionals. The very fact that the largest majority of all recording studios choose Quantegy product would suggest to me that your readers have already exercised their choice and I would like to put forward a few very good reasons for their wisdom in choosing Ampex products manufactured by Quantegy.

We have probably spent more time and money in research and development into professional audio recording media than any of our competitors past or present, and as a consequence, understand the requirement of the professional recording market better than any.

For many years we have remained fully committed to our customers' needs, offering a range of configurations in all formats far greater than any of our competitors. For analogue recording alone we offer a choice of four different products ranging from our flagship 499 GrandMaster Gold, through 456 the original GrandMaster, 406 for the less demanding applications, and 478 low print product designed specifically for speech recording.

You raise an interesting point regarding the way the market is viewed by some of our competitors when you say, 'if enough of "us" pick up on that choice, we will retain it. But if we neglect it, we run the risk risk of having it withdrawn'.

Unlike our competitors past and present, we have determined our direction and strategy to serve only the professional market, this has created a unique interdependency between Quantegy and its customers. Manufacturers with more diverse interests have readily quit this market despite holding a significant share of the market, for Quantegy to consider doing so would be implausible, in short we need each other equally.

We were there at the beginning of professional analogue recording and we will be there at the end offering the same dedication and highest product for as long as there is an analogue machine pulling tape, can our competitors make the same claim? **Peter Goldsmith, Director of Sales, Quantegy UK**

Tim Goodyer writes: I readily acknowledge the legitimacy of Quantegy's position in the tape market. That it is committed to the future of audio was certainly born out when Steve Smith revealed of the company's plans to launch a new analogue tape at the Analogue Reality Check forum (reported in Studio Sound, October 1997) held during the recent New York AES Convention. In fact, the transcription of the whole discussion makes worthwhile reading for anyone interested in the state of analogue, and indeed, in both Quantegy's and BASF's position as well as that of a number of other well-informed panelists. However, I feel that the observation made in my Editorial stands. Although I quoted the tape market in the course of pointing out that dependency on

one source (of media, equipment, expertise or many other resources) can be dangerous, the principle applies across the board—in equipment, media and beyond.

On location

Thank you for the article on our ME-DART digital recorder and editor (*Studio Sound*, September 1997). We are not sure whether we understood the whole message of the introduction expressed in superb English humour but...

Originally, DART was intended to mean Digital Audio Recorder and Transmitter but as Switzerland is not a member of the EU, whenever anyone crosses the border with a DART the customs officers were suspicious of the name and it was not easy to convince the customs officers that there was no radio transmitter built in. The reason for this unusual interpretation of the name is, therefore, more historical and political than anthropological.

The reason the ME-DART and Kudelski's Ares C look alike is that both are digital successors to successful portable tape recorders. Mandozzi Electronics' intention was to keep the appearance and operation of the new device as similar to the old one as possible in order to facilitate the transition from conventional tape recorders to digital ones.

Maybe you have realised that the door of the disk slot is electronically conductive in order to guarantee EMC. In fact, you can record with ME-DART in the rain while operating a GSM cellular telephone without affecting the recorder's operation (as long as the microphone cable is shielded). The headphones can be used to check audio signal in a kind of read-after-write manner because they are fed from behind the A–D convertor and correspond to the recording on the PCMIA disk; digital clipping can thus be detected immediately.

Exploring a take was intentionally made similar to working with a conventional tape recorder—turn the cue wheel forward or backward and the sound produced is the same. One of the advantages of recording in 'linear mode' is that the jog and shuttle functions produce similar sounds both in the forward and backward directions at speeds varying from almost zero to twice real time. In addition, the time indications of the Takes and Clips are very accurate. All regular users are convinced that there is no other equipment that can be operated as easily as ME-DART.

Finally, the journalist's position when using ME-DART as a recorder must only be that of the Homo erectus if he doesn't have any place to rest the lower end of his spine or if he wants to play the alp horn at the same time. During recording, ME-DART can be carried comfortably on its shoulder straps and must not necessarily compress the pot belly. The recorder can also be operated if the reporter is seated or even lying on a comfortable chaise-longue. For editing, ME-DART can rest on a table top and the reporter must not bend his neck to operate controls and see what happens on the front panel (what is unfortunately needed for other modern recorders).

GG Bardola, Mandozzi Electronics, Switzerland

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360 Systems Short/cut, Digicart II Plus

Nonlinear broadcast systems owe a debt of gratitude to 360 Systems for its reconciliation of technology and practice. **Rob James** tracks the latest refinements in the broadcaster's art

HORT/CUT IS a stereo or 2-track desktop hard-disk recorder and editor, clearly designed with radio applications in mind. That said, many other areas —such as theatre and TV—will appreciate its virtues. Recording is to an internal hard drive and storage time for the Model SC-180-1 is around 90 minutes. Model SC-180-2 increases this to around 160 minutes.

The machine is intended to be an intuitive, productive audio tool. It can be operated completely 'stand alone' since there are monitor amps, small speakers and a mic preamp built in. Equally, it has connectivity functions to integrate with other 360 Systems devices, designed to provide complete station solutions to production and delivery. The case is a substantial sculpted metal affair. This contributes to a sense of bullet proof solidity. The unit's dimensions and weight make it a good proposition for field work. Add a microphone and it becomes perfectly possible to edit a piece, record links and complete in a hotel room.

The waveform display can be 'zoomed' horizontally to give 2s, 5s, 10s or 20s across the screen. Pressing the AET key in conjunction with the Zoom keys changes the height of the waveform from x1 up to x16. The rest of the commendably uncluttered front panel divides into three sections. The centre section has edit keys arranged in two ares with large CUT, COPY, INSERT and FRASE keys below which are MARK. i<< (go to previous mark) EDIT-IN, FDIT-OUT, I>> (go to next mark) and t sho keys. The prominent, blue button on the right is the BLEEP key which can be used to insert or replace a tone or silence. The jog wheel is a nicely weighted device with a finger depression and rubber outer ring set into a recess, all of which keeps everything at a low profile to minimise fatigue. in intensive use.

The Rec Gain controls have ARM and solo keys and the transport controls include conventional functions plus PAUSE. The Arm keys select tracks for insert recording and editing but hitting RECORD automatically arms both channels. New recordings are not destructive and may be initiated at any time without losing anything. All keys except QWERTY and BLEEP are internally illuminated.

The rear panel has analogue and digital inputs and outputs on NLRs. If a microphone is connected to the left analogue input a mic amp can be switched in with an adjacent, mechanically latching button. Nominal analogue level is +4dBu. An interesting addition to the digital I-O is the provision of BNC connectors to the putative AES 3-1-D standard. This enables 75Ω video circuits and cabling to be used for digital audio. Both flavours of digital 1-O are also used for the 360 Systems proprietary D-Net networking under which Short cut is a transmit only device.

A 15-pin D-connector provides for GPI connections to enable Short cut to be integrated



My children (aged 6 and 8)

managed to successfully

edit some speech after a few

minutes tuition. It is a pure

editor and does not have

the bells and whistles of

complex mixing, equalisation

and dynamics

into a broadcast and transmission environment. Play, Record, Rewind and Fast Forward can all be remotely controlled while Tally outputs are provided together with a 12V source to drive lamps if necessary. In addition, the internal speakers can be muted. The GPIs are user programmable to other functions. External storage is catered for by a 25-pin D SCSI connector, Hard disks, MOs including the new 3¹¹-inch LIMDOW, or Jonega Zip—perhaps the most obvious since

D-CARTII plus has a Zip drive.

All defaults and preferences, together with the quick reference to editing functions, are found by pressing MENT. The display changes to a list with one highlighted item. The list of options will be different depending on whether MENT is

invoked when FILES or DIRectories are lit. The highlight may be moved with the jog wheel. ENTER opens the submenu and a further press highlights the parameter to be changed. The jog wheel scrolls through the available options. ENTER accepts the option and EXIT takes you back up the menu tree. Far easier to do than explain and completely intuitive.

The storage model is one of Directories and Files. The maximum number of directories is 10, each of which can contain up to 200 files. Each directory except the Public one may be password protected. There is a master password to allow 'supervisor' access to change passwords or re-format the hard disk. The editor can work at 44.1kHz sampling or 48kHz and this may be set on a per directory basis. There is no sample-rate conversion so recordings made at differing rates cannot be intercut. In addition Dolby AC-2 encoding is available when exporting 48kHz material via D-Net. Once any recording or editing has been performed, save lights until you either save or deliberately abort. To perform simple recording all that is required

is to press rec; the resultant file will be given a default title which can be user defined the or Short/cut can be set to prompt for a title after each recording. In Threshold recording the machine waits, after rec is pressed, until audio reaches a preset level, then comrecording. mences Recording can be

forced' in this mode by a second press on the rec key. There are two Insert record modes. The basic is quite unlike the normal, video style insert. Recording commences at the cursor position and 'pushes' existing audio later in time. Replacement insert recording records over a region defined by the edit in and edit out keys. This mode is destructive to any preexisting audio in the region.

Editing makes considerable use of marks. The MARK key inserts location marks during any process and are used for quick navigation. Marks are tied to audio not the time-line. The EDIT IN and FDIT OUT keys define the edit markers. Only one pair exists for any file at one time. Performing an edit operation other



DigiCart II Plus and the Short/cut make a powerful combo for fast production

than ERASE places the audio between the edit marks onto the clipboard from where it can be inserted in a similar manner to the INSERT record operations. There is a single level of Undo and a second press will Redo. This works with all edit operations except erasing a directory from which there is no return. The Erase function actually replaces a highlighted section with silence, although this can later be restored by highlighting the erased or bleeped area and pressing ERASE or BLEEP, respectively, while holding UNDO. All edit and record operations can be performed on one or both tracks allowing the two tracks to be used for mono and slipped and so on.

Edit operations take a finite time to perform, say two or three seconds, before you can do anything else.

Any audio on the clipboard can be mapped to one of the 10 hot keys for instant playback. If a CLIPBOARD file is created, this can be mapped to a hot key so the contents of the clipboard can be auditioned at any time.

From a cold start. Short/cut boots in a mere 15 seconds or so---very much faster than any PC-based system. The boot cycle includes an attractive lamp test routine which runs through all the illuminated buttons. Meanwhile the display keeps you in touch with progress after the initial logo. The machine automatically opens the Public directory and illuminates the FILES key inviting you to open a file.

In use the Short/cut is highly intuitive. I managed to open a file, record

and edit some material and save the result without even looking at the Menu, Edit or Directories quick reference let alone opening the manual. However, when you do get around to it, opening the manual is a rewarding experience. Like the machine it is clear and concise. For simple operation everything you really need is pretty obvious. Closer examination of the manual reveals considerably greater depth with a raft of 'power user' key combinations. At the most basic this gives you reverse play by pressing the rewind key while holding the play key. As a film dubbing mixer I don't need convincing of the utility of this but if you haven't had the facility before, try it. I believe it is one of the quickest ways to edit.

The REWIND key will also take you to the beginning of the file if pressed twice in quick succession. PAUSE temporarily interrupts recording or replay, a second press resumes and there are many other 'smarts'.

DIGICART IS a replacement for NAB cart machines but, as you would expect, offers far more than the humble cart. The front panel has a large SELECT knob, three chunky illuminated buttons for REC STOP and PLAY and a bright display. At the top there is a Zip drive. The rest of the panel has 12 internally-illuminated keys, a pair of input level controls and level bar graphs. The rear panel has much the same connections as Short/cut so I will only highlight the differences.

Unsurprisingly for this application there is no mic pre-amp. The SCSI connector is a 50pin Centronics and the GPI s are on a 25-pin D-connector. There are also two 9-pin D-connector serial sockets; one is for factory diagnostics and the other for serial remote control. The remote control uses the ES.bus data format. This may be used for third party control or to connect the optional RC-220 remote which can control up to four machines. A 6-pin mini DIN connects a QWERTY keyboard. This may be a standard PC board or 360 Systems own mini, the RC205.

Recording formats are 32kHz 44.1kHz or 48kHz 16-bit linear or 48kHz Dolby AC-2 compressed. This is the default format and offers roughly 5.1 compression.

Each recording is termed a Cut. Supplementary infor-

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<<<< mation may be stored with each cut in a header. This contains an ID for Directory. Drive and Cut, an alphanumeric name of up to 15 characters, the total running time, the recording format and editing information (head and tail times, fade in, fade out, gain and secondary cue timing).

Cuts can be played individually or they can be assembled into plavlists or 'Stacks' There can be up to 1000 stacks per disk or drive and up to 1000 cuts per directory. Stacks may be Linear or rotating. A Linear Stack will play a list of items from a single play command. A Rotating Stack plays a single item and 'rotates' the pointer to the next item which will be played on the next Play command. Stacks can be nested one level deep within another Stack. If a Linear Stack contains a Rotating Stack each time the Stack is played the rotating element will be different. The converse also applies. Stacks may be converted into cuts

Edit functions: Output gain which sets the level for the cut anywhere in a 96dB range in either 1dB or 0.1dB steps. The head and tail trim, fade-in and fade-out, and Pre-roll functions are accessed through the EDIT MENU key and parameters are set using PLAY and STOP keys and or the SELECT knob. All edit functions are nondestructive. To save space, an edited Cut may be copied in edited form and the original deleted. Ideal for extracting sound bites from lengthy speeches. There is also a direct Overwrite Record mode which erases an existing cut.

Secondary cues can be inserted at any point within a cut. These are used to control external equipment via a relay closure which appears on the remote socket. Cues are inserted using the SEC key. LOOP invokes looping on playback only.

With the capacity of this device some way of searching and managing information is obligatory. Directories may be sorted by ID or alphabetically by name and there are powerful search routines which allow cuts or stacks to be located by number or name although this requires an alphanumeric kevboard. The optional remote controllers allow searching by ID number. Presets, Cuts and Stacks on hard disks can be assigned to hot keys on either the alphanumeric keyboard or the remote.

D-Net is 360 Systems proprietary networking protocol-machines conforming to the protocol may be interconnected using either the normal AES connections or the AES 3-1-D BNC connections. D-Net transfers are always initiated by the source machine. If several D-Net equipped devices are daisy chained together transfers may be made to a

specific target machine or broadcast to all machines. The network can also be set up in star configurations using video distribution amps or routers.

The target may be identified by its exact name or 'wild cards' can be used, so if you have machines named Master 1. Master 2 and Studio 1. Studio 2. a transfer to Master will hit all the Master machines. There are two modes of transfer: Mailbox transfers are always non-14



destructive and end up in Directory 9. Alternatively transfers may be made to specific locations. This will overwrite any existing material in the location. For safety, machines can be set to receive to mailbox only. Single Cuts. Stacks. Directories or entire drives can be transferred. Communication is one-way only so the transmitting machine does not have any indication as to whether the transfer was successful.

Short/cut does not operate natively in AC-2 mode which is probably a good thing for an origination device. Provision of AC-2 coding when exporting via D-Net allows standardisation on the format for playout if so desired.

Considerable thought and effort has been put in to making the D-Cart II plus easy to maintain in the field. The construction is modular and should be inherently reliable, far more so than analogue machines. Both machines are refined to a specific purpose and, as a result, operational complexity is kept to a minimum

The machines will find a home wherever a

360 Systems, 5321 Sterling Centre Drive, Westlake Village, California 91361, US. Tel: +1 818 991 0360 Fax: +1 818 991 1360. Europe: 360 Systems, H Meijerstraat 13, Pumerend 1444 HA, The Netherlands. Tel/fax: +31 299 433627. E-mail: pjlee@worldonline.nl

cart and or 1/4-inch machine were formerly used. In a wellrun radio situation with good housekeeping these machines offer an effective solution. With a rugged feel and clean user interface The Short/cut looks the part and is one of the easiest editors I have ever used. In fact my children (aged 6 and 8) managed to successfully edit some speech after a few minutes tuition. It

is a pure editor and does not have the bells and whistles of complex mixing, equalisation and dynamics. There are faster machines around but not in this price bracket and not so tightly aimed. The networking capabilities add spice to the feast. It is a thoroughly competent editor which quickly inspires confidence. The DigiCart II Plus is similarly impressive, quick to learn, with powerful features for preparation and on-air use.



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Fostex D-160

A new digital nonlinear multitrack machine from Fostex combines capability and convenience. **Rob James** thinks it may even signal the end of budget tape machines

OSTEX ENJOYS a long history of delivering cost effective recording devices in quirky guises. The company was among the first to produce affordable analogue multitracks, pioneering the 8 tracks on \forall_4 -inch, 16 on \forall_2 -inch and 24 tracks on 1-inch formats. The professionals were sceptical; the pragmatists bought the machines and made music (and money). Fostex was also the first manufacturer to produce a time-code R-DAT machine, the D-20, using a proprietary method of recording code later modified to comply with the emergent standard.

The D-160 is a 16-track digital machine aimed squarely at music recording, packaged in a 3U-high rack-mount box. The front panel contains nothing more than a power switch, recessed 15-pin D-connector for the control panel, a 3.5-inch removable disk slot and four evenly spaced stainless steel mushrooms. These allow the control panel to be easily and positively slotted on to the front panel. The connector makes it look a bit odd but this is amply compensated by the utility of one control panel useable on the machine or up to ten metres away. The disk carrier is a robust, metal cased affair and Fostex sensibly sells the empty caddies at a reasonable price so you can fit your choice of E-IDE drive. It also supplies a list of recommended E-IDE drives and devices for the SCSI-2 port. This is intended primarily for backup, although a suitable drive may be used for real-time playback. There are three format options for SCSI devices. Type 0, 2 or 3. Type 0 allows real-time record and playback from a suitable device. The other two are intended for backup only. The D-160 analyses the attached device and suggests format 0 or 2.

The rear panel carries phono connectors for unbalanced analogue I-O. 8 inputs and 16 outputs. MIDI In. Out and Thru are on the usual DIN sockets.

The SCSI 2 interface is on a 25-pin D-connector. Unusually for a hard-disk recorder there are no SPDIF co-axial or AES connections; instead digital I-O is on four optical connectors. Two of these carry 8 inputs apiece and the other two 8 outputs in Alesis ADAT format. In addition the machine's Data Out 1-8 and Data In 1–8 connections may be configured as optical SPDIF I-O. Fostex can supply an optional convertor for machines not equipped with optical SPDIF.

Two blank panels are available for option boards. The first, Model 5041, adds 8 inputs and 16 outputs, analogue, balanced +4dBu via three 25-pin D-connectors. These mirror the phono I-O. The other option is the Model 8345 time-code and sync board which adds digital and video sync and XLRs for time-code I-O and BNCs for wordclock I-O and video In and Thru. Both video and wordclock have switchable 75Ω termination.

The A–D convertors are 18-bit sigma-delta 64x oversampling types and the D–As are 20-bit sigma-delta 128x oversampling. All tracks can be recorded simultaneously, either via the digital inputs or 8 analogue and 8 digital.

A large, bright FL tube' display dominates with record track select buttons in two banks of four (plus SHIFT key) to access the 16 tracks. Below this are three LEDS marked LOCKED, HD ACCESS and MTC in. Transport controls are bottom right with Fostex's quirky arrangement of Record Stop Play Rewind and Fast Forward—a standard arrangement of transport controls would be welcome since in a studio with several machines from different manufacturers, it is all to easy to hit the wrong key in the heat of the moment.

The concentric jog-shuttle dial is smallish but more than adequate. Edit functions are on a row of dedicated dark grey buttons labelled COPY, MOVE, PASTE, ERASE, UNDO and REDO.

AUTOPLAY cycles through the autoplay and autoreturn options, and two associated LEDS indicate the selected functions. Two keys select Clipboard in and out times. The Auto-Punch facility has keys for START, IN, OUT and END and a separate AUTOPUNCH mode key above the RECORD key with two LEDS to indicate rehearse or take. The LOCATE key takes you to the most recently located position or the value can be edited. VARI-PITCH is invoked with a dedicated key and indicating LED. The final block of six keys at top right are the most obviously multi functional. DISPlay SELECT and EXECUTE/YES pressed together access Timebase Select. HOLD > and STORE together access Programme Change. The last button in the group is RECALL.

The display consists of 16 level bar graphs and 16, 35-dot alphanumeric characters which variously display messages, time or bar-beatclick information, and the amount of time



remaining on the disk. The right of the display is taken up with dedicated status and mode indicators. There is no track display. In view of the likely uses of the D-160 this is unlikely to be much of a handicap.

The D-160 uses Fostex' FDMS-3 (Fostex Disk Management System) This differs from the system employed on the earlier DMT series and D-80/D-90 which allocate disk space on a fixed basis. On these machines if you have sufficient storage for 8 tracks 15 minutes long, that is what you get. The D-160 allows recording anywhere in a 24 hour window on any track up to the maximum capacity of the disk which allows longer projects to be undertaken. Additionally there are 8 'ghost' or additional tracks. Complete 'real' tracks may be swapped with ghost tracks to free space for further recording on an individual or 8-track bank basis. Each disk can contain a maximum of 99 Programme files. Each file can be given a meaningful name. Programme files can be write protected to avoid accidental editing. Similarly tracks can be protected against recording.

Since the D-160 records individual files for each recorded 'event' it is possible to insert silence between existing events. This silence does not use hard disk space but does count as an event. The maximum number of events per track is 512 but the machine will not record after 508 on a given Track. When editing, each edit point splits existing events, increasing the number. The number of events can be monitored using the Setup menu. One suggested strategy for reducing the number of events is to copy a range of events to the clipboard and pasting the range back in the same place whereupon it becomes one event.

There are two monitoring modes Input Monitor and Repro Monitor. Arming a track for recording and pressing the REC key while stationary or commencing recording switches selected tracks to monitor source otherwise output is always Repro. There are no internal mixing options for track bouncing or monitoring or level controls since it is assumed the D-160 will be used with a mixer.

Basic recording can be done on up to 16 tracks at once if you have either digital sources or 8 analogue and 8 digital sources. Sampling rate may be either 44.1kHz or 48kHz. Punch-in recording can be either automatic or manual from the control panel or optional foot switch. Rehearse mode switches the monitoring on the tracks selected for recording from replay to source at the In point and back to replay at the Out point. On a take, the track(s) monitoring switches to replay about two seconds after the actual punch out.

Time can be displayed as ABSolute which is based on each new programme starting at zero hours and continuing to a maximum of

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24 hours, as MTC which is an offset added to ABS or MIDI bar-beat-click which corresponds to the MIDI clock-Song Position Pointer which can be created using the internal tempo map.

If the optional time code and sync card is fitted, the D-160 may be slaved to external LTC or MTC. Play speed code only is allowed at 24fps, 25fps, 30fps Drop frame and non-drop frame and 29.97fps Drop and non-drop. The digital 1-O can be locked to internal or external wordclock or external video reference.

Most of the setup options are accessed by first pressing the display select key then EXE-CUTE/YES. The jog wheel will then scroll through the possibilities. When the required parameter is displayed pressing EXECUTE YES allows it to be altered. A further press sets the value and returns you to the menu.

In addition to tape recorder-style transport controls there are a number of 'smarts'. Locate points are memory data points in time or barbeat-click depending on mode which are accessed using the CLIPBOARD IN/OUT, AUTO PUNCH IN OUT, AUTO RTN START/END and LOCATE keys plus the stop, rewind and FFWD keys in combination. Auto Play mode will start playing from the direct locate point. AUTO-RETURN allows automatic return to a preset start point when a preset end point is reached. In combination, the Auto Return and Auto Play functions allow looping; if they are used with Auto Punch Rehearse, a punch-in may be rehearsed repeatedly until satisfied and then taken. The transport has to be stopped after punching out before it will go into record again.

Editing is a simple matter of defining In

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and Out points then deciding what you want to do. COPY AND PASTE makes a copy of the data and pastes it to another location. MOVE AND PASTE does the same thing but erases the source data including the clipboard. ERASE allows erasure of data in three ways. Erase data from a specified point to the end of program on a track or tracks. Erase data on all tracks between two specified points and erase all data from a project (this one cannot be undone) All edit points can be edited using a combination of keys and the jog-shuttle dials.

Fostex Corporation,

Tel: +81 425 45 611.

UK: SCV Electronics.

6-24-Southgate Road,

Tel: +44 171 923 1892.

Fax: +44 171 241 3644.

US: Fostex Corporation of

America, 15431 Blackburn

Tel: +1 310 921 1112.

Avenue, Norwalk, CA 90650.

Tokyo, Japan 196.

London N1 3JJ

3-2-35 Musashine, Akishima,

Using Copy and Paste up to 99 copies can be pasted in one operation to extend a sound. Mono or adjacent odd and even pairs of source clips can be pasted to different tracks otherwise up to 16 track wide clips can be copied and pasted or moved or erased. There is a single level of undo-redo. All events are recorded with a fixed 10ms facte including edits.

VARI-SPIED allows plus or minus 6% variation although this function is sensibly locked out in certain modes since it

varies the sampling rate which would result in problems if using the digital I-O. Backup can be to any SCSI device. ADAT or DAT. Recommended SCSI storage includes hard disks, Zip, Jaz, MOs, SyQuest and PD. If ADAT or DAT is used as the backup medium it is possible to save the data as eight track blocks which are compatible with the D-90. The sampling rate of the target machine needs to be set the same as the programme to be saved.

So what have we here? Basically, it's a replacement for an analogue multitrack with the added benefits of basic cut-and-paste editing at an attractive price. Where the D-160 really comes into its own is in conjunction with mixers and tape based multitracks which use the Alesis LightPipe optical interface standard. In addition there is a good symbiosis with an ADAT machine which can easily be used for archiving as well as in production with the D-160. The only rather sad thing is Fostex does

not have a compatible DAW to cover more comprehensive editing and manipulation. If more tracks are required several machines may be daisy-chained together using MTC and wordclock to make a 32-track, 48-track and so on.

The D-160 is neat and practical. The user-interface is uncluttered and reasonably straightforward: although the manual could be clearer. The removable control panel is a master stroke. The unusual appearance when attached to

the main unit is a small price to pay for the convenience of using the same panel on the machine and in front of the operator. The machine emphatically does not use data compression, the convertors are more than half way decent. Throw comprehensive backup options into the equation and it should be a winner. Does this spell the final end of the line for budget analogue multitrack?



Denon DN-M1050R

Derided by the purists, the MiniDisc format is continuing to build its profile in professional applications. **Jim Betteridge** puts the format and the latest Denon machine to the test

F THE DIFFERENCE between 24-bit/96kHz and 24-bit/192kHz is significant at the mastering end of the market, when it comes to a play-out format for radio, television and theatre, it's not an issue. Witness the growing success of MiniDisc—not only is it 16-bit, 44.1kHz, but it also suffers the indignities of ATRAC data compression.

We're up to version 4 of ATRAC (only the replay algorithms have been altered, so all versions are compatible with each other) and

I must say it sounds pretty darned good. Listen carefully to the tail of a sustained piano note drifting into silence and you don't need golden ears to tell that something is afoot, but for the vast majority of broadcast and theatre requirements, MD does very nicely. It's also, apparently, got a projected life span of around 30 years, which is longer than CD and much longer than CD-R.

Denon is unquestionably the MD market leader in the broadcast sector and has found

You don't have to wait several lifetimes to achieve purity



particular success with its DN-990R. Built primarily as a play-out machine, the 990R is very compact (4U-high, 1/3 rd rack width), offers a front panel of foolproof simplicity and can take ongoing 24-hour use without complaint. You can see these machines in theatres and radio stations and they are also in use in television for wild (non-sync) audio play-out. Where once a DJ or engineer reached for cassette. 1/, inch, NAB cart or DAT, the MD has largely taken over. And with good reason. It'll edit like a slick reel-to-reel machine, you can properly name your tracks, the discs are small, inexpensive, robust (no fear of corrosive finger marks as with CD-Rs) and will give you 74 minutes stereo, 148 minutes mono. It's rerecordable, and if you erase a track, that time is given back as free disc space. In the field, a reporter can carry a low-cost portable machine on which to record, edit and, with an appropriate interface, send the programme back down a telephone line to an MD machine at the station-including all edit information. Broadcasting House, for example, has 12 DN-990Rs permanently on line ready for just such occasions.

If production staff have been quick to recognise the operational advantages, engineers have been harder to convince. One of the main concerns has been what happens to a signal that is passed through multiple data compression systems—ATRAC, ISDN and satellite (one encode-decode at a time, of course). It seems, however, that tests carried out by the technical department of a major radio station show quite acceptable results with no aliasing problems apparent.

The DN-M1050R supersedes the DN-1100R and offers more facilities, more options and a 20% lower price tag—that's progress. It is designed to be a workstation rather than a simple play-out machine and thus is presented in a full-width rackmount format with a larger display and generally more buttons. This is not to say that the front panel is cluttered, far from it. Effort has been made to provide one knob per major function to aid quick and easy use.

Indeed, operationally things are pretty straightforward with large illuminating buttons bearing legends in ordinary English. Just as well really—the manual is less than fabulous with tiny print, an abrupt style and no index. On the positive side, it does contain all manner of technical data normally only found in a workshop manual including schematics, pin-outs and exploded diagrams of its mechanical construction.

So, approaching the machine with a job to do: You put it into record by pressing record (switches monitoring from playback to line-in) and then play; you can also set an incoming audio threshold at which the machine will automatically start recording. In either case the next available track number is allotted up to a

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DN-M1050R - a unique intelligent and skilled designed. Production staff will be keen on the MD machine

maximum of 255 per MD blank. If you're recording analogue you can adjust the left and right inputs independently with two rotary controls. It would be nice to have a 'calibrate' setting here rather than having to rely solely on the position of slightly vague knobs and couple of chinagraph marks each time you record. Track numbers can be set to increment during recording by various means: each time you press RECORD again; if the signal drops below an adjustable threshold for more than two seconds; or if you connect a DAT or CD via SPDIF it'll copy across the track IDs.

Having recorded your programme, let's say it's Track 1. you can top and tail it by dividing it at the relevant points using the DIVIDE button. The edited track then becomes Track 2. with Tracks 1 and 3 (unwanted air) at either end. These you can quickly erase. When you bin Track 1. all the track numbers shuffle up one so that the programme becomes Track 1 and the air at the end, Track 2; erase Track 2 and you're left with your programme. Track 1. This could be scary with a discful of tracks idented with numbers alone and so it's recommended that you name your tracks a bit sharpish after, or indeed while. recording. This is done either from the front panel (relatively painless) or a standard PCstyle PS2 QWERTY keyboard. Once named, it's very easy to track skip up and down with a rotary control to find the required track. When a track finishes playing it can automatically cue up again or cue up to the next track or just stop. An END MON button plays the last few seconds (adjustable) of a track so you know what to expect-a brilliant facility. Up to 25 tracks can be put nose-to-tail (tracks of silence can be used as spacers) and tracks can be combined (undivided), all very quickly and easily. A track can also be inserted in an existing sequence. A sequence of up to 25 tracks is a Program and you can have up to three programs. Varispeed is available on playback, ±9.9%.

An edit point is found using the 'repeating frame' technique employed by Fostex DAT machines. It is not as accurate as a good quality hard-disk system, but it is very quick to use and if you have a nice clear edge to cut on plus a little bit of luck (where the frame falls in the music), quite acceptable music edits are possible. Assembling tracks and editing inter->>>>

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16 IN+16 OUT-DIGITALDOMAIN-REALTIME-SYSTEM

CreamWare announce the launch of TDAT16 - but be forewarned... you'll have to add your name to the list of recording engineers who are ahead of you in the queue for one! Why? TDAT16's all new, yet mature and stable - we've taken our proven tripleDAT interface and added the powerful capacity of 16 I/O's, whilst keeping all the features that earned tripleDAT a 1997 T.E.C. Award nomination! Why gamble with new products that are not proven in the field? Don't risk your valuable studio time and recording talents with one piece of hardware combined with some 'off-the-shelf' software interface. TDAT16 is a finely tuned hardware AND software solution. No installation problems and no waiting for an update that actually works! Install TDAT16 and your creativity starts flowing. The fastest editing with all the power tools that make tripleDAT so outstanding: integrated CD-Writing and a whole suite of excellent REALTIME effects. You won't find a product like this at any price! IDAT16 is the HDR solution for the digital studio. Now you're digital, stay digital. Link your digital mixing consoles, your tape (like Alesis ADAT or DAT) and your TDAT16 powered PC together without ever having to turn back to analog.

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The DN-1050R is part of a new generation of products from Denon, which includes the DN-C680 CD player

<<<< views, on the other hand, is very easy. The rear panel provides balanced and unbalanced analogue plus AES-EBU and SPDIF digital ins and outs. A 15-pin D-type connector offers a comprehensive parallel interface and a 9-pin connector provides serial control, switchable between RS232 and basic RS422.

There are three optional boards available, all simply fitted by the user. The first is a RAM board providing an instant start or Hot Play mode by loading the first few seconds of up to 20 tracks into memory. Keys 1–0 and Q–P on the QWERTY keyboard can then be used as triggers. A Hot Key pad providing 20 large dedicated trigger buttons is due in January. The second option is a 48kHz-32kHz sample rate convertor I-O card (MD only actually records at 44.1kHz).

The third option is a control card, ACD-27, currently in preproduction and expected in numbers in the Spring. It provides video sync or word clock in-out (as standard, the DN-1050R has no word clock input and so incorporating it into a digital system may be tricky), better Sony 9-pin P2 control, and will allow the DN-1050R to chase and lock to SMPTE-EBU time code as well as record material in against time code (25, 29,97 and 30 frame rates). The MD format has no facility to record or replay time code to or from disc, and so the timing information is held in the machine and is lost at power down. The time-code DAT format, then, is not under threat.

While admitting to having once hardened my heart against MD and its compressed ways, it's now very clear to me why production staff are so keen, and, having heard ATRAC Version 4, I must admit to being something of a

Denon, Hayden Laboratories, Hayden House, Chiltern Hill, Chalfont St Peter, Gerrards Cross, Bucks SL9 9UG, UK. Tel: +44 1753 880109. Fax: +44 1727 846490. convert. Quick. cheap, simple, flexible, compact, robust—this is what the people want. This is absolutely not to say that higher sampling and bit rates aren't the way to go for future mastering

formats, they are. For everyday acquisition, simple editing and playout, however. MD has a lot going for it.

The DN-1050R is generally a very intelligently and skilfully designed machine with

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lots of large, clear, dedicated buttons and knobs to make operation simple and immediate. Denon has been bold in its vision. creation and capture of a new professional market for MD in broadcast and live events and resultant success is no mystery.





🗱 TL Audio EQ-5013, VP-5051

The arrival of autumn colours herald the introduction of a new line of valve outboard from this British design house. **George Shilling** tries on the lvory parametric and voice channel

L AUDIO is now well known for its Classic series of valve outboard equipment. As digital recording has grown in popularity, so, too, has the antidote: valve outboard equipment to sweeten and warm the perceived 'clinical' sound of digital. This burgeoning market in affordable digital recording has been a Godsend for TL Audio which has rapidly expanded its budget valve and transistor ranges: the unusually coloured Indigo and Crimson ranges to be specific.

Around 10 or 15 years ago, most outboard toys were designed along Ford's Model T approach: available in black only. Upstarts such as Focusrite and Joemeek changed all that, and now just about all the primary and secondary colours have been tried by one company or another. TL Audio's latest range is the Ivory series—the off-white approach has already been more convincingly employed by Amek, but let's not get too hung up about looks. The new Ivories are priced similarly to TL's Indigos, and like the Classic and Indigo ranges, all feature valve technology. They look like particularly good value as, unlike the unswervingly 1U-sized Indigoes, all five models in the lvory series are 2U-high rackmounters. This affords more space for features missing from the Indigo equivalents. The other

In use, I found the EQ-5013 superb. As you ease up the gain on any of the frequency bands the effect is subtly progressive for the first 45° of travel. At 90° the EQ is very audible, and immensely powerful.

advantage to their size is that vents are no longer needed on the top panel; they are instead sited on the front and sides, enabling units to be placed above and below their sisters with no cooling problems. My guess is that eventually the Indigo range may be left by the wayside, but personally, I prefer the look of the old deep-blue. Not only is the colour smarter, but the bare steel cases with recessed screws seem cleaner cut than the Ivories', black cases with clumsy protruding screw-heads. Still, once they're in the rack you don't see these bits, and build quality is undeniably high, with large toroidal transformers adding substantial weight. Both units reviewed employ 12ANTA-ECC83 valves on their circuit boards.

Externally, both feature the same black plastic knobs found on the Indigo range. They are sensibly given colour-coded caps for their different functions, but it is possible that after a few knocks in a busy studio the caps could come adrift. Variable pots are pleasantly damped, but their legending could benefit from a finer scale. Double the number of indicator marks would enable more accurate matching of settings on stereo units, and make recalling settings much easier. Push-buttons feel cheap and clunky but work well enough. Both units have a voltage selector switch and IEC mains socket



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EQ-5013

with separate adjacent fuse holder. Unfortunately no manuals were available for these early review models (both featuring a serial number ending in 001). Judging from past experience with TL Audio equipment I am sure they will be clear and straightforward.

The EQ-5013 is a 2-channel valve parametric equaliser; each channel having four bands of EQ. On the rear there are unbalanced jack inputs and outputs and balanced XLRs. I agree that there is no need for balanced TRS jacks: if you want a balanced connection then vou use an XLR, don't vou? Both input and output sections feature push-button selectors which change operating level. They switch the jack sockets between -10dBu and +4dBu. and simultaneously the XLR connectors between +4dBu and +18dBu. You might wonder why anyone would need the latter: this is for ADAT users wanting to achieve maximum recording levels without struggling with front panel controls', extreme settings.

On the front panel the two channels have completely independent controls and are set one above the other. From left to right they each feature the following.

First, an Instrument input jack provides more gain and good matching for, say, an electric guitar. Two LEDs marked DRIVE and PEAK accompany an input gain pot which has a range of ±20dB with a centre detent at zero. The parametric EQ controls follow. Each band has a ±15dB GAIN pot with centre detent. FRE-QUENCY knob with approximately a 5-octave range (30Hz-1kHz, 100Hz-3kHz, 1kHz-12kHz and 3kHz-20kHz) and BANDWIDTH knob with adequate range, probably the same as that found on the Classic series EQ-2 of 0.5-5. Next there is an EQ ON button and LED for each channel, and an OUTPUT pot with the same range as the INPUT pot. Finally there is a Power pushbutton with LED.

In use, I found the EQ-5013 superb. As you ease up the gain on any of the frequency bands the effect is subtly progressive for the first 45° of travel. At 90° the EQ is very audible, and immensely powerful at 15dB cut or boost. The bandwidth control goes to a narrow enough setting to give a good wah-wah effect using the low mid band, and equally goes wide enough for broad warmth or presence to be added or taken away. The INPUT and OUTPUT gain pots remain active whether or not the EQ is switched in, and as such there is no way of bypassing the effect of the valve circuit onboard. The wide range of gain settings enables you to fully control the amount of valve drive, added to any signal. whether or not you wish to EQ. At normal levels, the yellow Drive LEDS flash on and off with the occasional red peak LED flicker. How-

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ever, by judicious use of the gain controls you can drive the valves harder until they start to audibly distort. That famous Edwyn Collins. 'Girl Like You' guitar sound (achieved on record by overloading a vintage Neve input channel) is attainable with this unit, simply by plugging into the Instrument jack and cranking the knobs up. The VP-5051 features separate XLRs on the back for mic and line inputs, and an unbalanced jack line input. The line inputs feature a level switch similar to that found on the EQ but, inconsistently, the output jack and XLR are fixed at -10dBu and +4dBu respectively. There are additionally a TRS side chain insert and a Link socket for stereo operation >>>>







Soho starts at less than £30,000

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- Dual Machine Control Ports plus MIDI machine control
- Multi-channel FX Rack (Reverb, Delay, Chorus, Flanging, 6/12 Band EQ, Mastering Compressor)
- 19 inch Pod a rack unit option to house DAW controllers or outboard FX devices

Soho is the latest addition to Amek's range of fully specified digital mixing consoles. Developed specifically for audio post production applications, Soho is designed to be integrated with any existing or favoured Digital Audio Workstation. The sleek and ergonomic design and highly impressive specification makes it ideal for companies who require a cost-effective digital console, while maintaining the quality, professional image and functionality of their post production operation.





VP-5051

<<<< of two units. On the front there is an Instrument jack, and a selector switch to choose between the three inputs. A fourth position enables a phantom-powered mic to be used. A single GAIN control operates for all inputs, centre detented for zero line level. Drive and Peak LEDs are found nearby. There is a switchable 90Hz high-pass filter: I would have preferred a lower frequency. The Compressor section features an ON switch, THRES-HOLD, RATIO (1:1.5 to 1:30), and GAIN MAKE-UP knobs, Attack and Release characteristics can each be set to four different positions, although even at the fastest settings the compressor is not lightning fast. The expandergate has a single control setting threshold from Off through -40dB to a maximum -20dB, with a Gate Shut LED. This worked well, although I would be reluctant to use it when recording vocals. A 4-band EQ section has four preset frequencies for each band with some overlapping of bands, and ±12dB gain on each. The two middle frequencies have a fairly broad preset bandwidth of 0.5, with 12dB/octave shelving top and bottom. This is fine for gentle shaping, brightening and warming of vocals, and is in fact the same arrangement as that found on the popular Classic series. EQ-1. On the right there is a round Focusrite Redstyle illuminated vu meter with a switch to select Input, Output and Gain Reduction, with additionally an Output +10dB position. This position indicates 10dB lower than actual level on the meter, the latter useful if you need a high output level to interface with certain digital equipment. The OUTPUT GAIN knob cuts the signal at full anticlockwise, always a bonus when connected directly to a multitrack with-

Tony Larking Professional

Sales, Letchworth, Herts SG6 1AN, UK. Tel: +44 1462 490600. Fax: +44 1462 490700. US: Sascom Marketing. Tel: +1 905 469 8080. Fax: +1 905 469 1129. Germany: SEA, Auf dem Diek,6, 48488 Emsbüren. Tel: +49 5903 9388 0. Fax: +49 5903 6141. out going through a desk fader. There is an EQ ON button, with an EQ PRE button adjacent which places the EQ before the compressor. A LINK button enables stereo operation when another unit is connected. Finally there is a power ON switch and LED.

The overall sound quality is unmistakably valve, but per-

haps a little honky and less refined compared with more expensive competitors. There is a danger of overloading and generating a nasty crackly distortion if gains are not set carefully to match any EQ boosts. You cannot obtain the nice progressive distortion of the EQ-5013

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if you overload the EQ section. Strangely, this distortion can occur without the peak LED lighting. Another potential problem I found was that the mic input was not particularly sensitive. With a very quiet vocalist you might struggle to get the compressor to work hard enough using microphones which have a low output, even with input gain and threshold at

full tilt.

Despite a few niggles with the VP-5051, both Ivories represent good value, and the EQ-5013, while not including any filters or shelving bands, beats off competing units with much higher price tags. Now, how does that Edwyn Collins riff go?



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NEW TECHNOLOGIES

Broadcast Phoenix

A broadcast console of modular design, the Phoenix has been designed to allow the user to choose the number and type of modules and where they are placed. It can be installed with 12 to 48 mono or stereo input modules, 2 to 8 group modules each comprising 1 to 4 additional stereo inputs, six



aux modules, two master modules, one talkback module and 1 to 4 monitor modules.

The mono module has one mic and one line input with an additional mic input available as an option, one remote control unit, a balanced insert and 4-band EQ. Stereo modules have two stereo inputs, two remote control units, a balanced stereo insert and 2-band Baxandall EQ. Each aux module has one stereo and two mono auxes.

Fougerolle, France. Tel: +33 1 39327350.

Earthworks uni

The Z30X cardioid mic has a claimed smooth response to 30kHz and very accurate impulse response. Its polar pattern is said to be unusually flat to some 75° offaxis while handling noise and proximity effect are said to be lower than most directional mics.

The mics are complemented by the LAB101 single channel preamp and the LAB102 2-channel version. Features include: polarity reverse, phantom power, standby, stepped gain control, clip LED and variable gain to the output. Earthworks, US. Tel: +1 603 654 6427.

Tube voice channel

Lydkraft has revealed the Tube Tech MEC1A single-channel voice processor combining a mic preamp, equaliser and compressor in traditional blue livery.

The device draws from component parts of the existing Tube Tech range including the preamp from the MP1A, the compressor from the CL1B and the equaliser from the impressive EQ1A. Shipping is expected at the beginning of the year. Tube Tech, Denmark. Tel: +45 3871 0021.

Octavia 8/24

SAVL's Octavia 8/24 is a multiple output tracklaying and dubbing system that integrates with large mixing consoles but retains the editing facilities of Octavia 8/08. As with Sadie and Octavia 8/08, Octavia 8/24 operates with the recently released V3.02 software. It has 8 inputs and 24 outputs in analogue and digital and can play at least 24 tracks off one SCSI disk. All >>>>>

BSS Opal DPR522

Commonly reduced to compressor 'extras', gates have currently all but lost their profile. Zenon Schoepe finds their saviour

SS HAS NOT HAD a dual-channel power gate in its catalogue since the long-in-the-tooth DPR502. Somewhat ahead of the game, this unit employed a toe-curlingly involved means of generating a MIDI note, complete with an envelope, from a channel's gated signal. Like many such things, it all seemed fairly reasonable and acceptable for the time, but if the novelty of pulling the device out of the rack in order to throw the rear-panel dip switches wore off, then you could always resort to the unit's ADE (Auto Dynamic Envelope) button which served as means of accentuating the leading edge information in the incoming transient and perking it up a bit. That feature earned an unqualified all-round 'Good!', but you have had to wait until the release of the Opal DPR522 gate to enjoy it again.

This is only one of the features of this new unit which joins the company's compressorde-esser in the more financially accessible Opal range. The look is handsome and typical with the those characteristic illuminated winegum channel selectors positioned centrally on useful inclusion of a paralleled balanced output jack signal on each channel. For those who really like to knit their own connectors then each channel comes with an 8-pin socket for remoting such functions as gate opening.

Other twists include a DUCKING switch, for flipping a channel's modus operandi on its head, and OPEN CHECK which fires the envelope manually so you can inspect the process.

You're left in no doubt that this is a premier league processor as nothing has been left out. In practice, there really is nothing that you could throw at this device and give it a headache. It's perfectly balanced and wonderfully fine in it level of control.

BSS' ADE is more pronounced and dramatic than the equivalent feature on competing units-give it a good snare and you'll make it that much better while retaining the variation in dynamics of the original. Often it is all that is required to pull a track along. ADE is best suited to percussion, but it has a use with bass and sharp rhythm guitar parts



a box that offers dual fully-featured noise gates with frequency assisted keying and stereo linking. The last of these combines signals from both inputs and hands the control of both signal paths over to the pots of Channel 1.

Metering is again excellent with a multicoloured LED display showing the relationship between the Threshold value set and the signal that is working it below and above it. BSS always does a great job in metering and the particular arrangement on the DPR522 contributes most to the rapid optimisation of a channel for the desired effect. The pivotal Threshold LED always glows. whether exceeded or not, so there's never any doubt about how close to triggering the gate you are. Additionally, individual LEDS show the gate's open and closed states, and extinguish in keeping with the created envelope. Excellent.

Both channels have exactly the same features and vou're presented with fully-variable Threshold (-50dB to +20dB), Range (-80dB to

BSS Audio, Linkside House,

Tel: +44 1707 660667.

Fax: +44 1707 660755.

EN6 3JB, UK.

Summit Road, Potters Bar, Herts

0dB), Attack (20µs to 1.5s), Hold (5ms to 2s) and Release (1ms to 5s). The filter section involves a FRE-QUENCY control fully variable 60Hz-13kHz and an accompanying wiDTH pot that works between 0.1 and 8 octaves

There's a KEY LISTEN switch for monitoring the section and a button that injects the external key input arriving on a balanced jack. Connectors are on balanced XLRs with the providing you're prepared to accept that it doesn't always work creatively for these.

A really good dedicated gate is still quite a marvel to behold and the Opal stirs the memories with some of those features that have now been committed to history with the popular trend for auto oriented boxes and simple gates that are just stuck on the end of a compressor chain to bolt down the inevitable noise build-up. The DPR522 tugs at the heart strings with fully variable ATTACK and HOLD-essential if you want to get serious-and the sort of razor-fine filter that can delve into complex programme and produce unerring and repeatable triggering with uncanny precision. It's an arrangement that I find preferable to the highpass and low-pass filter combo, although theoretically the results ought to be similar.

You can keep your multichannel auto-time constant devices for the rudimentary gating duties, but no rack is complete without at least a couple of channels of clever stuff. BSS' new 'power gate' qualifies with flying colours.

This is the unit that the original DPR502 always should have evolved into, and you've got to wonder why it has been such a long time coming. If you want extended gating control then this box should be high on

your list-indeed it is increasingly difficult to find advanced gating functionality without other bits of dynamics processing hanging off the back. The Opal DPR522 is up there with the best established processors of its type.





Trantec S4000

Breaking cover in radio-mic manufacture, Trantec are pitching against some of the big boys. **Neil Hillman** seeks counsel

SHOULD HAVE felt better about myself than I did. It was just a confidence thing I suppose, but it became so bad I needed therapy. So one Saturday morning, above a dry cleaners in Moseley, I stood head bowed, surrounded by eight people seated on stackable plastic chairs and introduced myself, through the quietly powerful Belinda: 'My name is Neil Hillman and I am, erm, a sound recordist.'

Very good Neil. Well done. Remember no-one here sits in judgement of you—we are here as neural networkers supporting you back to emotional enrichment and a fulfilled life. In a silence broken only by three washing machines accelerating into their spin cycle below us. I began: 'I suffer from feelings of inner turmoil—anxiety, anguish, panic, fear and loathing—if I am forced into using radio



microphones. I've tried convincing directors they can cover the words in a close-up and coerced cameramen that a shot says much more with less headroom so that I can use a boom mic: I've moved from VHF to UHF and even changed manufacturers, spending more and more money while my underlying insecurity still feeds off the conviction that no other recordist is plagued by the same snap, crackle and pop that I'm convinced will appear halfway through a major scene.

No one spoke. Finally, a large lady in her mid-50s, freshly boulfonted in pearls and a tweed twin-set asked, I wonder, have you considered the all-new Trantec S4000 UHF diversity-receiver radio microphone?

More confidently she went on, it brings the latest UHF PLL synthesised tuning technology to within the reach of all radio mic users and comes programmed with 16 intermodulationfree frequencies contained within the UK TV Channel 69, but it can be configured with up

to 32 frequencies to increase its capacity for the worldwide market. All this for a whisker over £800 including a personal mic.'

The meeting broke up shortly after that, but at home the \$4000 specifications made impressive reading: frequency response for the receiver being 30Hz–18kHz, the transmitter 70Hz–18kHz, distortion less than 0.3% and a dynamic range greater than 110dBA. The

output of the belt-worn transmitter is 30mW

with an RF sensitivity down to 5μ V, powered by a 9V MN1604-type battery. At present only the mains receiver is available; although a matching pocket transmitter-receiver system will be available by next spring.

The system is easy to configure, the transmitter controls are neatly tucked behind a sliding flap that when pushed up from the bottom allows access to the battery compartment, or when pulled down reveals the transmitter gain control knob, large enough to be operated without a screwdriver, but, strangely, without a scale of numbers to record optimum settings for individual artists. Either side of this are two recessed buttons that allow for changing between up to four banks of intermodulation-free channels (shown by BAnc in the transmitter's LCD window when the top button is depressed), or scrolling through the individual channel frequencies (displayed as FrEq).

I did wonder just how robust in the field this sliding clip would be, but certainly when closed the plastic housing, while light, gave the impression of withstanding all but the severest of maltreatment.

The tuned-to-length antenna is securely screwed into the top of the transmitter and the microphone is connected via a Lemo 4pin plug. A red LED indicates that the unit is on through a switch located on the top face.

The receiver configuration is achieved through two-front panel push-buttons and a setup menu. These allow quick and easy changing of Channel and Bank. display software Program Info. and receiver Mute Level, or Name Input which can be used to identify the user of that unit with 8 alpha-numeric characters and the Display Name? function. Three leds on the front panel show which of the two receiving aerials is active, and audio peak from the transmitter just before distortion is likely.

Range testing gave a credible 100m, but the frequency response of the audio chain did not appear to live up to the specifications. The unit sounded noticeably bass-light around 125Hz–130Hz, either as a result of the unmarked personal microphone supplied, or, perhaps, the trade-off between a system that utilises a squeezed bandwidth to fit in more intermodulation-free channels in the allocated 8MHz of the UK's Channel 69.

Given the stunning looks, the overall standard of this product and the price; with a

Trantec Systems, Kestrel House, Garth Road, Morden, Surrey SM4 4LP, UK. Tel: +44 181 330 3111. range including guitar transmitters and hand-held microphones. Trantec has placed itself perfectly for the price versus quality battle the radio market is about to

> embark on with the launch of, among others, Sony's Freedom range. I'm just not sure about the quality for critical recording applications.'

Belinda's eyes lit up: 'We have several workshops for indecisiveness...'

NFW TECHNOLOGIES

<<<< connectors are on the back of the 5U processing unit with a single high-speed serial connection to the host PC. All audio processing is performed on SAVL purposebuilt hardware and within Octavia 8/24 there are four AT&T 32-bit floating point DSPs for 33Mflops of processing power.

The system comes with 20-bit A–D and D–A Delta Sigma convertors on all inputs and outputs but is designed to operate from 11.025kHz to 96kHz sampling rates and from 8 to 24 bits. It chases or trigger locks to all LTC and VITC time code rates and can be referenced to AES or video sync. **SAVL, UK. Tel: +44 1353 648888.**

Speaker setup

Genelec now includes a quick reference guide for monitor setup with all its units. Using clear diagrams the company draws attention to erroneous positioning and gives an indication of the correct and preferable location for speaker angle and distance, vertical distance, and the symmetry of the room installation complete with bass tilt adjustment suggestions. Bass tilt and bass roll-off settings are also recommended for speakers in corners and against front walls. Speaker positioning on stands and above the mixing console surface are also tackled together with the recommended critical distances of freestanding monitors from back walls.

1092A and 1094A subwoofers are now equipped with a new discrete subwoofer signal input channel. The new crossover filter retains the three front channel LCR input and output connectors, but gives an



improved overall system response when used with digital surround systems. The additional subwoofer input is required by decoders with a separate low frequency effects channel output. The upgrade makes the subwoofer equally applicable to analogue matrix type and digital discrete type surround formats.

The diminutive 1029A monitor is now available in a grey finish.

Genelec, Finland. Tel: +358 17 813311.

Wireless talkback

The IFB 1624 is a UHF wireless system with two channels that can be employed in sportscaster situations where split feeds are required or in a mix to mono mode giving TV presenters the ability to adjust the relative levels between two IFB sources on one earpiece.

The stereo transmitter is programmed with 16 UHF frequencies between 518–865MHz and users have a >>>>>

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Drawmer MX40

A 4-channel punch-gate has joined the MX30 at the affordable end of things. **Zenon Schoepe** investigates

The SECOND in Drawmer's new affordable dynasty, the MX40 is the natural accomplice to the MX30 dual-channel compressor-limiter-expander (*Studio Sound*, July 1997) in supplying four nicely condensed gates in the same 1U-high rackmount. Gone is the champagne-brushed finish of the early MX30 models in favour of a plainer brushed silver background for the compartmentalised, black, processing sections. The MX40 differs also in having only balanced XLR connectors, rather than the MX30's jack and XLR combinations, and these are accompanied by a single jack key input for the whole unit.

Access to this is selected from each channel's EXTERNAL KEY switch meaning that you can effectively drive all the gates simultaneously from one key input or combinations thereof. Not the most flexible arrangement, but one that is actually convenient should you wish to group together, say, any two channels without having to dabble around with separate triggers.

As an aside, when no external key is plugged into the box then Channel 1 can be

Peak Punch amounts to a processing tool, and if you've never heard it, but regularly find yourself attempting to massage a bit of attack out of flabby kicks and snares then this is something to look at. It immediately makes things apparently more lively and aggressive.

That concludes the lesson on the features and operation of the MX40 which is pleasantly organised and ever-so-slightly expensive in feel; basically it feels like a Drawmer and there are no shortcuts here.

Gates ought to be either open or closed, and the worst types are cursed with the tendency of not being able to make up their minds consistently about which way they ought to swing. Certain combinations of programme, level and noise gate can dip you into an excruciating never-never land. Pots equal control and theoretically the more you have the more chance you have of avoiding the aforementioned situation. Lesser be-knobbed 'auto'-style gates do some of the thinking for you, and are better at general cleaning duties than they are at the fancy stuff because, quite



UK: Drawmer, Charlotte Street,

Tel: +44 1924 201089

Wakefield, West Yorkshire, WF1 1UH

used, by selecting the EXTERNAL KEY buttons on the other channels, as the trigger source on the other channels. This arrangement means you can use it to tighten up something like a horn section, where you assign the tightest player to Channel 1 and have the remainder follow his channel's dynamics. Additionally, Channels 1&2 and 3&4 can be linked via a SLAVE LINK switch, in which case total control of the envelope for the paired channels is handed over to the odd numbered one.

The majority of switches have associated LEDs, there is neat and clear 3-LED metering for gate activity, and useful front-panel graphics that tell you what each channel's three pots do. Like the MX30, the MX40 gets by with few knobs: TRIGGER FREQUENCY, THRESHOLD and RELEASE. All channels are identical.

Trigger frequency (50Hz-8kHz on an octave Q) works with a FILTER IN switch and KEY LISTEN, while Threshold scans from -60dB to +20dB. Release time is fully variable from 10ms to 4s clamping down to switch selectable ranges of -90dB and -20dB. The whole section can be bypassed, but the unusual inclusion, at this end of the price scale, is a PEAK PUNCH

switch. This puts a rocket up the leading edge of the envelope causing it to become more pronounced and dramatic. It's a process that Drawmer has

employed in other units and, of course, BSS has also pioneered the cause with its own ADE system, interestingly enough also implemented on its own less budget Opal series gate (see review in this issue). simply, they can go a little soft in the envelope at extremes. The Peak Punch feature, what must be some clever Attack time programming and a fair resolution frequency filter makes the MX40 quite animated sounding and solidly resilient to chatter.

It's in power-user gating that you lose out: the lack of a width control on the filter frequency means that by definition tuning can be a little general while the addition of fully variable attack would permit leading edges to be shaped subtlety.

Of course, I am fully aware that one should reserve true unadulterated and unrestricted excitement for boxes asking many times the cost because the cheap stuff shouldn't be encouraged or even acknowledged in pro circles. Well, I'm sorry, but I am, again, extremely impressed by what Drawmer has come up with for the job, here, and I don't really care how much it costs; the fact that it is cheap is yet another achievement. I was knocked out by the performance and simplicity of the MX30. The MX40 is not four channels of DS201 for beer-and-peanuts money and you shouldn't kid yourself that there is any-

where near the precision and control available. However, what the MX40 does do is present accessible, effective and, most importantly, reliable and results with a minimum of

predictable gating results with a minimum of fuss and drama. For a lot of people this will be all they need for the mundane majority of tasks. The dilemma is that they might still expect to pay more. Try it.

NEW TECHNOLOGIES

The IFB 1624 can be used with Garwood's standard size 16-frequency receiver or with a miniature bodypack featuring two independent volume controls. The LV2 miniature receiver weighs less than 70g with battery life of over 3 hours from a single AA cell. Garwood, UK. Tel: +44 171 328 1211.

A&H GL3300

The 8-group GL3300 sound reinforcement console, like the GL2000 and GL4000, is a dual function mixer, allowing it to be used in FOH, monitor roles, or combining both tasks simultaneously.

GL3300 builds on the GL3000, the most notable additions are 4 mute groups, group mute switches, direct outputs on all channels, an 8 group out option on a D connector, groups 1-4 inserts, +6dB boost on auxes and matrix sends, and the choice of different mono/stereo input configurations. GL3300 is available in 16, 24, 32 and 40 channel frame sizes.

The MixWizard Series WZ14:4:2 is designed for live sound and recording applications and is also a dual function console. It has 4-band EQ, 100mm faders, vertical PCB construction, nutted pots and MSP (Minimum Signal Path) architecture.

Allen & Heath has extended its Contractor Series with the GR05, a four-zone mixer featuring 2 mic inputs and 3 stereo line inputs, with front panel level control on all inputs and outputs. Mic inputs are on balanced XLRs and have rear gain trims, 3-colour LED meters and EQ specially designed for speech.

A number of internal jumpers and trimmers, which the installer can use to create a custom package for each installation, con-



figure resources such as phantom power, pads, filters, routing matrix, priority ducking, and ganging of multiple zone outputs. **A&H**, UK. Tel: +44 1326 372070.

Harbeth 30s and 40s

Harbeth's Monitor 30 and 40 are the results of ground-up redesigns of the BBC LS5/8 and 5/9 monitors. Both retain roughly the same dimensions of the original models allowing continued use of stands and ceiling cradles and both are available as active models. Harbeth claims that coloration has been dramatically reduced across the range by the use of its patented cones and advanced computer modelling.

The HL-Monitor 40 is the company's first 3-way system and uses a 300mm woofer and Radial midrange cone. It boasts a large 'traditional' box with classic BBC components like a thin-wall cabinet (here with tensioning rods), complex network and >>>>

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AT 4033a

Audio-Technica AT4049, 4051, 4053, 4033a

Updates to its 40-series mark Audio-Technica's pursuit of the professional mic market. **Dave Foister** checks them out

S INCE FRIGHTENING the competition with the 4033, Audio-Technica hasn't stood still for a moment. The surprisingly smooth and classy all-rounder was followed by a multi-pattern model, the 4050, which quickly found favour in high places. The range expanded last year with the 4041, a long thin cardioid with a similarly disproportionate quality and range, and in the same way this has now been augmented with a multipattern modular system, the 4051. If only A-T could explain the logic of the numbering system joy would be complete.

In fact, this is not strictly a new model; it has been reintroduced after an absence from the

A-T catalogue of a few years; although it never had a particularly high profile first time around. The fact remains that the timing of its relaunch fits well with the development of the whole range, and is bound to attract more attention following the success of the other models.

The 'new' series is centred on a body

(which is quaintly referred to in the literature as a handle) containing the preamp electronics. on to the end of which can be fitted a choice of three capsule elements. This is, of course, in the style of the AKG 451, Blue Line and 460 series, the Neumann KM80s. Sennheiser's K6 models and others. The principle is well established, and the criteria are the same-given a good audio performance. How easy is it to change heads, and can it be done frequently without damaging screw threads? Is it truly a modular system where patterns can be changed as readily as on a switched-pattern model, or is it a set-and-forget arrangement where you buy the bits you want and leave them mated semi-permanently?

In the case of the 4051 the answer can be guessed at by looking at the overall mechani-

cal quality of the parts. This is no cheap lightweight; the body is machined brass, plated with black chrome, which gives it a reassuring solidity and weight. The heads are built to a similar standard, and mate to the body with a substantial screw thread which for once inspires confidence. Changing heads is no 'kid gloves and tweezers' job, and the

threads look likely to survive many changes —even hurried ones—without stripping. The carrying case for the microphone has inserts in the foam for all three heads, making it clear that they are designed to be changed at will. The case also includes a simple foam windshield as standard, which is probably just as well as I found the cardioid capsule I was using for voice to be particularly susceptible to blasts. The only other standard accessory is A-T's own rugged metal-based stand mount, making for a slim convenient package in an adequate plastic case.

The only control on the microphone is for bass roll-off (6dB per octave below 80Hz); no pad is thought necessary as SPLs of 143dB can be handled within 1% distortion figures.

The three available capsule heads are the 4051 cardioid, the 4049 omni and the 4053 hypercardioid, all clearly identified by white graphics of their patterns printed on the side. In the case of the cardioid and hypercardioid this is virtually the only visible difference, but of course the omni head lacks the grille slots in the sides, and, consequently, is unmistakable. All perform extremely impressively, with a notable neutrality both on and off axis whatever the pattern.

The cardioid will find itself the usual allrounder, and could make itself so useful in its own right that it will be harder to sacrifice it when one of the other patterns is needed. Its principal characteristic is clarity, with the expected HF extension matched by a commendable warmth at the other end. Significantly, the sonic character remains more or less constant right across all the capsules, making it particularly easy to switch to hypercardioid when circumstances demand it. I found myself selecting this head more often than I expected, knowing that the improvement in spill rejection would not be compromised by sacrifices in the sound. The omni is particularly impressive, with all the openness and smoothness you need when a microphone is giving you all the room ambience.

Any one of the three possible microphones in this range is a strong contender in its own right; put the package together as a complete kit and it becomes very attractive indeed. The only danger is that you'll want to use all the heads so often that you'll end up buying three bodies.

Still not resting on any laurels, Audio-Technica has also upgraded the

model that began it all, the US: Audio Technica, 1221 established 4033. According Commerce Drive, Stow, OH 44224. to A-T this was prompted not by any adverse com-Tel: +1 216 686 2600. Fax: +1 216 688 3752. ments from users or UK: Audio Technica, Technica reviewers but because they House, Royal London Industrial found themselves able to Estate, Old Lane, Leeds LS11 8AG. improve a couple of areas Tel: +44 1132 771441. and wanted to pass the Fax: +44 1132 704836. upgrades on. The result is Email: sales@audio-technica.co.uk the 4033a, with a higher

> a greater resistance to mechanical noise. Certainly nothing has been lost in the changes; the new version remains as impressive a performer as the original, and will no doubt continue to win friends.

SPL handling capability and

NEW TECHNOLOGIES

Harbeth, UK. Tel: +44 1444 440955.

Weiss DS1

The Weiss Gambit Series de-esser-compressor-limiter is based on the design of the EQ1 equaliser and uses virtually the same user interface differing only in the array of processing controls.

It can operate as a frequency selective compressor or broadband compressor the latter being set from linear to limiting with adjustable predelays, auto release and softknee transfer function. The frequency crossover is implemented using ultra-steep linear phase FIR filters.

Weiss, Switzerland. Tel: +411 940 2006.

Tascam MDs

Tascam has introduced two additions to its MD range. The MD501 a mid-price 2-track recorder-player designed for radio, live theatre and installations while the MD301 is a budget version of it.

The MD501 has balanced XLR I-Os, a PS/2 PC keyboard connector for disc and track naming while a port for fader or event start allows remote activation. Auto ready and program play make the unit suitable for broadcast while an SRC converts incoming digital to 44.1kHz and there is digital level control and programmable fade.

The MD301 offers the same playback and



editing facilities with coax and TOS optical digital I-O but unbalanced jack connectors. **Tascam, UK. Tel: +44 1923 819630.**

Millennia mixer

Offering full compatibility with the GML HRT-9100 mixing system, the Millennia Mixing Suite claims audio excellence and modular flexibility. With an expanding selection of input and outputs options, the mixer is intended for critical location recording, submixing, MIDI rig mixing and multisource mastering. Mixers can be combined for a total of 80 inputs. Employing circuitry similar to that found in the company's mic preamps, the Mixing Suite is almost entirely DC coupled without the addition of servo amplifiers.

Millennia, US. Tel: +1 916 647 0750.

Digital clocking

Sync DA allows a digital studio to be clocked from one device and can resolve word clock and 256 superclock from any AES-EBU signal in addition to distributing word clock to six devices.

It is used to distribute the clock of Aardvark's low-jitter master clock generator the AardSync II and can distribute from DAT, DAW or digital mixers. Front panel >>>>

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MX2442 24 Channel 4 Bus Console

MX8000

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Korg SoundLink 1212I/O

The I-O options offered by a new I-O card make Korg's SoundLink workstation a hot prospect. Dave Foister reports

HEN KORG SURPRISED us with the 168RC digital mixer, it was clear from the SoundLink badge that there was more to come. Outboard A-D convertors and D-A convertors were available to accompany the mixer, linked via ADAT-format 8-channel optical interfaces, and the scene was set for a broad-based system of integrated units, now joined by the 1212I-O card for the Macintosh (and others).

This PCI-bus card derives its inspiring name from the fact that it offers 12 channels of input and output in various forms to the computer. Eight of these use the ADAT interface, augmented by a stereo SPDIF interface and two channels of analogue. Without the ADAT optical connector it's hard to see how so much could have been fitted on to the edge of the card, and a few extra measures have had to be taken to squeeze it all in. SPDIF is on a mini-DIN with a breakout cable, and is backed up by wordclock in and outthe spec for the card is pretty comprehensive and intended to help it integrate into any environment. The wordclock connections are on another special breakout lead, alongside an ADAT sync input which will allow suitable software to become an additional transport on a multimachine system.

Analogue input and output is unbalanced on TRS jacks, but can cope with +4dBu or -10dBu levels. Jumpers on the card alter the operating level, which can be set independently for inputs and outputs.

A single floppy disk carries both the installation software for the card and a comprehensive utility package allowing various basic functions to be carried out with the card separately from any full-blown audio system. A single window carries controls for a surprising range of parameters, making the card a useful toolkit and problem solver in its own right. Any input can be routed to any output, and each channel's volume is independently adjustable; even phase reverse is provided. Analogue input trim. clock source and sample rate are all controlled from here, and the utility can handle multiple 1212 cards. This setup allows control of digital bouncing between ADAT machines, copying of SPDIF sources to a pair of ADAT tracks and vice versa, and even simple mixing perhaps for monitoring purposes-multiple sources can be routed to the same output, although the lack of either panning or built-in headroom means it can hardly be called a mixer.



LEDs show the exact sample rate. Aardvark, US, Tel: +1 313 665 8899. The UK Office. Tel: +44 1442 870103.

Micro amp

The updated Micro system from ARX includes the MicroDrive amp which is a 3-channel unit with on-board speaker processing. It contains all the circuitry of the MicroPro loudspeaker processor plus amplification to power one or two MicroMax speakers per side. MicroMax on-board processing provides phase correction EQ contour, crossover to sub functions and Interactive System Control which monitors the amp output level and protects the speakers. Extra outputs permit slaving up to other amplifiers while Enhance circuitry creates the dynamic impact of a much larger system.

The LSP2 dual-channel, dual-band loudspeaker processor supplies Band 1 (low frequency) and Band 2 level control, variable HF EQ trim and ISC fail-safe speaker protection. Designed to complement >>>>



0 000 000 000 00 0 EQ-2 Dual Parametric Valve Equaliser



EQ-1 Dual Valve Equaliser

www.americanradiohistory.com

VI-1 8 Channel Valve Interface

Web: http://www.sascom.com



<<<<< the company's Concert Series Loudspeakers, the LSP2's operating parameters are preset by a range of plug-in model-specific modules for each of ARX's products. ARX, Australia. Tel: +61 3 9555 7859.

Cheap effects

DigiTech Studio's \$100 is an affordable dual multi-effects processor that can be configured in series, parallel or combined. Effects include stereo reverbs, multitap delays, chorus, flange, tremolo, detune and parametric EQ with 99 factory programs and 99



user presets. The device is MIDI and footswitch controllable.

DigiTech Studio, US. Tel: +1 801 566 8800.

Phase check

The Turbosound TPC-1252 Phase Checker Set is designed to check the acoustic phasing and electronic polarity of multiple loudspeaker arrays. The set also performs a variety of other test functions, including cable continuity, cable polarity, locating lines in multicore cables and verifying signal flow through components in the audio path.

It consists of two units: the TPG-12 Pulse Generator and the complementary TPD-52 Pulse Detector. The TPG-12 plugs directly into a power amplifier, electronic crossover, mixing desk or other input via an extended male XLR connector (no patch leads are needed) and generates a unipolar pulse.

The rugged, hand-held TPD-52 detector carries a condenser mic for measuring acoustic phase, and a female XLR for electronic polarity. The microphone's low sensitivity ensures that ambient noises will not affect the detector. It senses the positive or negative leading edge of an incoming pulse to determine its polarity, and displays this information with either a red or a green LED.

The TFM-330 is a high specification professional active floor monitor that >>>>

Send///!

The key to the next step is the panel for Sound Manager routeing: any two inputs and any two outputs can be nominated as the Mac's primary audio interface for the Apple Sound Manager. This means that any Mac application that deals in sound in any way, even for beeps and alerts, can use the 1212 for its I-O, allowing, for instance, SPDIF to be fed directly to the computer and +4dBu analogue outputs to be obtained from it. There are sev-

eral dedicated audio packages, such as Adobe Premier and Bias Peak, dealing purely in stereo via Sound Manager, and these can be given considerably more functionality by routing in and out via the 1212.

The ultimate goal of course is integration with a full-blown DAW, and several familiar names now support the card. Logic Audio v3.0.

Digital Performer v2.11, Deck II v2.6 and Steinberg VST v3.5 can all use the card's full facilities, giving all kinds of enhanced operation. Material can be digitally transferred backwards and forwards between the DAW and an ADAT system: the DAW and its associated MIDI side can be synchronised seamlessly with an ADAT system, becoming effectively another transport with the same format digital output; and the workstation's audio can be fed digitally to the Korg 168 mixer for real handson digital mixing. Since the 168 has two sets of ADAT optical inputs, the workstation can be mixed alongside a single transport as a completely integrated system. The SPDIF and analogue I-O allows aux sends to be set up within the computer if required, and the use of a combined audio-sequencing package allows the 168 to be automated via MIDI. On a simpler level, Korg's 880 A–Ds and D–As can be interfaced directly to the workstation, probably offering a significant improvement over those previously fitted, and including a few

phantom powered microphone inputs. As with the Sound Manager setup, telling the workstation software to use the 1212 as its interface is a simple set-and-forget operation. although routing remains flexible and comprehensive.

The latest development is that the card now also works on the PC. Windows drivers are already available from

Korg's Internet site and will be bundled with the cards shortly. Software support from several major workstations will not be far behind, bringing full ADAT digital interfacing to yet more familiar packages.

Korg's decision to go whole-heartedly for the ADAT format as the foundation for an entire collection of equipment was a bold one, and the concept has clearly been used imaginatively. This card makes sense of the whole SoundLink system, offering true and comprehensive integration of a wide range of equipment in a simple and highly flexible form.



UK: Korg UK, 9 Newmarket

Tel: +44 1908 857100.

Fax: +44 1908 857199.

Tel: +1 516 393 8520.

Email: soundtracs@korgusa.com

NY 11747-3201

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Studio Sound December 1997

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Røde NT-1



The 1990s mic feeding-frenzy is beginning to calm. Instead **Dave Foister** is finding a generation of rational microphones

HE SPLASH Rode has made must be the envy of the microphone world. To have achieved such success with its first two models—the NT-2 and Classic mics—is quite remarkable, and a testament to the Australian company's ability to spot a gap in the market and fill it with something special.

When I reviewed the original NT-2, I was



puzzled as to what had happened to the NT-1: had it, perhaps, been an unsuccessful design project scrapped in favour of what finally appeared, or was the numbering simply intended to make Rode look more established than it was? (A common trick in record catalogue numbering), Neither would seem to be the case, as the NT-1 is now with us, slotting in below the NT-2 in terms of facilities and cost. which was, perhaps, the idea all along.

After the sojourn into

valve territory with the Classic. Rode is back in transistorland for the NT-1. Straightforward solid-state circuitry sits in a straightforward solidly-built body, which itself owes nothing to either of the existing models. Gone is the satin finish of the other two in favour of a battleship grey paint: gone too is any attempt to flatter by imitation a familiar standard. Instead the body has simple elegant lines with a heavy wire grille over the capsule, producing a shape halfway between the classic impressions of an ice cream cone and a shaver. The sleek contours are unmarred by any kind of switches or controls, and even the Rode badge is surprisingly modestly printed.

The arrangement for stand mounting is unusual and effective. The attachment consists of a swivel-mounted ring which fixes to the body at its base, locked on with a knurled

screw collar that allows for rotation of the microphone in the mount. The swivel is secured with a large knob, and the whole thing is quick to adjust and very solid once set. It doesn't offer any kind of shock protection, so a shock mount (the SM-1, presumably the same as that used on the NT-2) is available as an optional accessory. No form of windshield is provided or available separately, even though

Rode recommends using some sort of protection when recording vocals: in practice the existing grille seems to do a pretty good job, keeping blasts out more effectively than some other models in a straight shoot-out.

Since there is nothing on the microphone to adjust, there seems little point in producing a manual for it: nevertheless, Røde has managed to write a longer Instruction Guide than some manufacturers provide on far more complex models. The result is an idiot's guide to condenser microphones, most of which is sound enough; although it does contain one strange exhortation. Rode asks that all connections should be made to the NT-1 before power is applied, and that it should never be disconnected while powered. I know no-one who switches phantom on and off while rigging and striking like this, and if it were really necessary it would be a major drawback. The closest I've come to having to worry about such things was a dodgy old concert hall PA console whose preamps were liable to expire if a microphone was plugged in with the phantom on, which everyone regarded as a real pain. I have to admit that without thinking I disregarded the instruction concerning the NT-1 a couple of times, with no discernible ill effects: if there is a reason it requires this special treatment it should be made clear, and if not then the manual shouldn't ring the alarm bells as it does.

The reason the NT-1 has no controls is the fact that it is a single-diaphragm design and therefore operates only a cardioid pattern. Not long ago, a big side-fire microphone with only one polar pattern was something of a rarity -such simplicity was the domain of smaller end-fire models-more recently, though, several have appeared from a wide variety of manufacturers. Yet almost every major player now has one, presumably on the grounds that lots of people rarely use any other polar pattern and therefore can't justify paying bigmicrophone prices for double diaphragms and extra switches. It's certainly an attractive way of offering your wares, and introducing their benefits, to a new audience.

This, then, is what Rode has done, in a microphone which although styled very differently embodies the same essential character as its more complex precursor. Here again is the

overall smoothness with a soft presence lift—nothing too colourful or brash, but enough to help a vocal poke through effortlessly. It's a good deal flatter than some of the comparably-priced competition, with a class to the sound that approaches the familiar favourites. Its looks are a lot less pretentious than some of the new would-be stars, but it still looks substantial and reassuring stick it in front of a singer and

they won't feel short-changed (always half the battle to my mind). And you won't feel short-changed by the sound: everything that put the NT-2 where it is can be found in the NT-1, which must surely follow in its footsteps.

New Technologies

<<<<< incorporates a custom designed 4inch voice coil, 15-inch low frequency driver and a high power 2-inch high frequency compression driver on a custom waveguide. The HF horn pattern is designed to give essentially the same response when used very close to, or standing back from, the monitor.

The THL-811W is a 2-way passive full range loudspeaker enclosure designed for use in professional sound reinforcement applications. It incorporates two customdesigned front-loaded 10-inch mid/low frequency loudspeakers, matched via a third-order internal passive crossover net work to a high efficiency, high frequency compression driver loaded with a wide dispersion horn.

Turbosound, UK. Tel: +44 1403 711447.

Zeus intercom

The RTS Zeus 2400 DSP intercom is aimed at auditorium and stadium installations, broadcast and simulator situations and military and industrial applications. The system is designed to give 24 channels of beuer than CD quality audio in a 2U package. Heart of the system is a TDM ASIC for audio switching and mixing in the digital domain which works at 20-bit, 44.1kHz.

Zeus-Edit, a Windows-based configuration package, is included for on-line and off-line user defined configuration capabilities. It ships with a library of presets All channels are compatible with existing Telex matrixing keypanels and 4-wire 1-Os for programme and IFB feeds. Zeus offers eight

nts -

GPI inputs and outputs to trigger external devices and a serial port is provided for connection of Telex accessories. Telex, US. Tel: +1 612 884 4051.

Monster monitors

Described as 'the world's largest control room monitors', the Alderson Acoustics AA15(1283) is a solfit mount system capable of very high SPLs. Each cabinet measures 44 inches wide by 92 inches high by 20 inches deep and weighs 650lbs. The cabinets are divided in to two sections for ease of mounting and positioning with the driver component consisting of four 15-inch woofers, twelve 7-inch low midnanges, eight 2-inch dome high mid ranges and eight 1-inch dome tweeters per cabinet. Cabs are fitted with ten 1500W stereo amps and two 4-way digital crossover-equalisers.

Frequency response is claimed to be linear, from 25Hz to 19kHz, and the system is said to achieve an SPL of 125dB continuous at 5m.

Less hairy chested is the AA412 (444) freestanding monitor for large mastering rooms which is built up of four 12-inch woofers, four 7-inch low mid >>>>

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UK: HHB Communications, 73–75 Scrubbs Lane, London NW10 6QU. Tel: +44 181 962 5000. Fax: +44 181 962 5050. Email: sales@hhb.co.uk US: Event Electronics, PO Box 4189, Santa Barbara, CA 93140. Tel: +1 805 962 6926. Fax: +1 805 962 3830.
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TASCAM 5 Marlin House, The Croxley Centre, Watford, Herts, WDI 8YA. Brochure Hotline 01923 819630

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Calistan Solutions Wave Safe

The audio security offered by DAT can now be extended to your PC's audio. **Dave Foister** reports

ESPITE the head start enjoyed by the Macintosh, the PC as an audio tool is coming of age. Besides the big highend workstations, there are more and more applications dealing in audio, and thanks to Mr Gates the *de facto* standard for audio files, from little beep beep noises to full CDs, is the WAV format in its various permutations. Like any other format, audio files are prone to errors in transfer and problems with hardware, perhaps more so because of their enormous size, and bad or missing data can render an entire file

useless. Unlike most dedicated audio storage media, a PC's hard disk incorporates no error correction whatsoever, so a tiny discrepancy that corrupts the overall structure of the file can be fatal. In particular, difficulties can arise when using various backup media, from CD-R to DAT, where media problems can lead to errors in the restored file.

Help is at hand in the form of Wave Safe from Calistan Solutions. In a sense, adopting the role of a dedicated audio medium's error correction, this remarkably simple and inexpensive package can store enough details about an audio file to reconstruct it in the event of subsequent damage, recovering as much data as is physically possible given the nature of the problem. Three versions, from Shareware to Pro, are available, and it is the capabilities of the full-blown (but remarkably inexpensive) Pro version we shall be looking at here.

Wave Safe's operation is a 2-stage process, beginning with a step to protect the file (or batches of files). or in other words to log its details in a database for future reference. Three degrees of protection are available, trading thoroughness against speed: the most detailed version takes twice as long to do its work as the medium version, but has substantially more potential to deal with problems

should they arise. On a Pentium 90, seemingly regarded as entry-level these days, even the most thorough option does its job remarkably quickly, certainly fast enough not to deter a user from bothering. As an idea of its power.

if one byte in a 50Gb file has been changed, the full protection will find it. The protection process generates a database file which is tiny compared with the original audio file, and the software can deal with multiple databases for different projects.

Once a protection database exists for a .WAV file, it can be checked against the original data at any time and any corruption will be reported. The check will find both damaged and missing data, and report both their exis-

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the structure of the file, where the bytes appear to begin and end. for instance, turning anything after the problem into unplayable gibberish. If such difficulties are encountered, the next stage of the process is to fix the file. This probably won't correct the actual damaged area, but it will restore the subsequent structure so that the rest of the file becomes playable. One of Wave Safe's demo examples has bad data very near the start, and what follows is completely

tence, and their positions, within the file to

within a fraction of a second. The full signifi-

cance of this is only apparent when you

realise that bad or lost data are likely to affect

follows is completely unusable, producing unpleasant garbage when read into a standard editor. Fixing it by means of the protection file gives a few milliseconds of glitch, the unrecoverable error, followed by perfectly useable audio.

Another scenario is the occassional corrupted byte here and there, which may otherwise escape detection, although the audio integrity will be compromised. Wave Safe will find and report these, providing the opportunity to read a backup again after, say, cleaning the heads.

Perhaps the most powerful of many features exclusive to the Pro version of Wave Safe is the Rescue function. Sometimes, thanks to damaged backup media, power problems or other catastrophes, files can be so badly damaged or truncated that Windows can't even recognise them as useable files, never mind load them into an application. Even in these cases Wave Safe Pro is able to put the pieces back together, as was graphically demonstrated with a deliberately-scratched floppy disk—simulating the kind of damage found on a crashed hard drive or a damaged optical medium—which appeared to be beyond

redemption. Wave Safe

reconstituted a recognis-

able file in which most of

the audio was as good as

new. In all these contexts,

this could make the differ-

ence between saving and

losing that great solo, or

between a small vocal

Calistan Solutions PO Box 17550, London NW2 5ZA, UK (sae for shareware version). Fax: +44 171 691 7823. E-mail: sales@calistan.demon.co.uk. Net: www.calistan.demon.co.uk

patch and a complete new session.

With .WAW files becoming ever more central to computer-based audio, the vulnerability of conventional media is bound to become more of an issue. Audio generates particularly large files, making it statistically more susceptible to problems as well as less likely to be routinely backed up. Wave Safe would like to see itself become indispensable in this environment, and it would appear to have the power, the speed and the ease of use to succeed.

NEW TECHNOLOGIES

Alderson Acoustics, US. Tel: +1 212 691 0027.

PortaMonitor

RTW's PortaMonitor complements the company's existing product line and uses three DSPs for a wide variety of measurement functions and analysis. It provides level indicators as peak or vu. an audio vector scope, a correlator, an AES-EBU status monitor (showing all the status and data bytes) and a ¹/₃-octave analyser. A surround option helps the depiction of 3:1, 3:0 or 2:1 signals in a special display mode on the vectorscope and when used in this mode the device calculates the Lt and Rt stereo signals 40 examine the stereo compatibility of the surround mix.

The PortaMonitor uses a 320 x 240 pixel TFT colour display for measurement results, bar graphs and figures and is distinguished by high-contrast images and a wide viewing angle.

RTW, Germany. Tel: +49 2 21709130.

Valve bias

Biasking is a simple but accurate diagnostic tool designed to read current flowing through the output tube of an amplifier. It works on most octal socket tube amps, is small enough to carry in a guitar case, requires no power source other than the



amp it is plugged in to, and can be used to match tubes into pairs. Svetlana, US. Tel: +1 415 233 0429.

ISDN remote

The ISDN remote audio router control usesthe auxiliary data facility available on many ISDN audio codecs to control an Audionics audio router at the far end allowing full remote control over which audio feed is sent on the audio codec's return line.

It can be used to remotely monitor studios around the world from a central position or from anywhere where an ISDN line and compatible codec are available. Engineers can monitor the audio paths around distant transmitter sites, remote access can be provided to programme material for redistribution or editing and outside broadcast presenters can select their reverse feeds without intervention at the studio end.

Rather than being stored locally, the source names are received in real-time from the matrix. This means that for each matrix accessed, the remote module will display a new set of available source names, exactly the same names as those shown in >>>>

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DX-5050

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Introducing the

Studio Audio Industry Recognition awards

STUDIO SOUND will host the first Audio Industry Recognition (AIR) awards during the course of the Amsterdam AES Convention May 1998.

The *Studio Sound* AIR Awards, or SSAIRAs as they are the already being referred to, are the result of repeated requests from the industry for the world's leading international and longest established professional audio publication to put its weight, reputation and integrity behind an awards system that recognises the technical achievement of the industry it serves.

The selection process will be democratic and unbiased and will allow *Studio Sound's* readership to vote and judge the outcome.

The SSAIRAs will combine product category awards with wider industry specific awards.

The months leading up to May will see nomination and voting forms appear in the magazine and **only registered readers of** *Studio Sound* **will be eligible to vote**, although theoretically anyone can nominate a product for consideration within a category.

Manufacturers will be free to nominate product but will not be able to vote and this will be policed through the requirement of all voters to include their unique identification number, which is included on the address label of the magazine, with any correspondence via mail or Email. This unique number contains information on, among other things, its owner's employment and job function so security, authenticity and the avoidance of duplication are ensured.

Only products launched at and after the AES Convention in Munich 1997 qualify for inclusion although it is worth pointing out that only those that are deemed by the *Studio Sound* editorial team to be in a deliverable state by May 1998 will ever make it through to the final selection process. *Studio Sound* is not interested in perpetuating software myths; either a product exists and can be bought or it waits for the following year's SSAIRAs.

The first stage of the process is the nomination of products for categories so look out for the form in *Studio Sound* and the precise details of what to do. This will be followed by a voting form of the nominations in the magazine and on the *Studio Sound* web-site.

The winners will be announced at an Awards ceremony during the course of the Amsterdam AES Convention in May.

Anyone wishing to discuss the nomination procedure should contact myself or Tim Goodyer at Studio Sound. Zenon Schoepe

NEW TECHNOLOGIES

<<<< the studios at the far end. Although it is envisaged that ISDN lines will be the most popular way to use this system, it can also be used with any digital data line that supports RS232 transfer.

The MCX1 series of 1U audio switchers offers 24x1 mono or 12x2 stereo switching facilities. An 8 digit alphanumeric version which indicates each selected source is available and the source names can be custom configured to suit user's needs before it leaves the factory. An update service is available to allow for future changes to a system.

Typically the MCX1 series can be configured for transmission selectors, outside source selection, monitoring panels or as a workstation selector.

Audionics, UK. Tel: +44 114 242 2333.

Hi-bit convertors

Developments of established Prism convertors, the AD2 and DA2 are 24-bit/96kHz capable A-D and D-A convertors respectively.

Included in the AD2 is a synchronous sample rate convertor along with a stereo peak-reading bar graph. Many of the features are accessible via a menu system and an LCD while a range of stored configurations can be selected using dedicated selector buttons.

The DA2 has up to seven inputs supporting 96kHz and other high sampling rates with double speed and 2-wire AES3-type interfaces. An 'exceptionally' stable PLL has high jitter attenuation and the unit's digital outputs can provide a dejittered digital feed-through for reliable digital transfers plus a master clock mode in which the effects of incoming clock jitter are eliminated completely. **Prism, UK. Tel: +44 1223 424988.**

Multipurpose box

Audio Centron's CE6 multipurpose enclosure features a 6.5-inch polypropylene long throw woofer with rubber surrounds and an extended voice coil. For the top end it uses a 1-inch impedance transformer dome tweeter which produces a claimed nominal frequency range up to 22kHz. A rear mounted threaded template accepts a variety of omni-mount, wall mount or ceiling mount adaptors. The optional CE6MSM permits the cabinet to be attached to a mic stand. **Audio Centron, US. Tel: +1 314 727 4512.**

Acoustic modules

IAS-A1/A2 and IAS-D1 are acoustic modules for the low-priced acoustic optimisation of smaller studios. The modules are accompanied by general installation instructions free of charge. An alternative to on-the-spot consultation is the measurement CD which can be used for simple measurement of reverberation time in the rooms which can then be evaluated by the acoustic engineers of IAS.

The modules are extremely flat in design and provide absorption of resonances and the optimisation of reverberation time. The A1 is a low/mid frequency absorber, the A2 a broadband absorber for reverberation time correction while the D1 is a diffuser for mixing the generated sound energy. IAS, Germany. Tel: +49 2241 62918.

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AD-2

96KHz 24-BIT A/D CONVERTER

State of the art performance with Prism Sound's proprietary A/D conversion system

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Built-in digital-to-digital synchronous sampling-rate converter with SNS noise-shaping system for high-quality down-conversion for CD

Comprehensive range of interface formats: TOSLINK, SPDIF, SDIF-2 & AES ports with both 2-wire and 2x speed 96kHz formats

High-quality internal master clock plus proprietary high-order, high-attenuation triple PLL system to eliminate external clock jitter without incurring long lock-up times

Fully impedance and voltage balanced transformerless inputs

DRE encoder and decoder for hi-bit performance using 16-bit media

DA-2

96KHz 24-BIT D/A CONVERTER

State of the art performance with Prism Sound's proprietary D/A conversion system

Full 24-bit processing for all inputs avoids truncation distortion or extra dithering

Comprehensive range of interface formats: TOSŁINK, SPDIF, SDIF-2 & AES ports with both 2-wire and 2x speed 96kHz formats

Interface jitter effects 100% eliminated with clock master mode, plus proprietary high attenuation triple PLL system

Independently floating, transformerless balanced outputs

7-way multi-format source selector, with feed-through digital output

DRE decoder for hi-bit performance using 16-bit recording media



O UNDERSTAND the passion with which vintage outboard is regarded is to witness the manifestation of a primeval instinct present in most engineers. It can be distilled down to a yearning for simpler times when choices were fewer and the mind and ears were better able to concentrate on fundamentals and basics.

Most engineers could eventually be persuaded to admit that they believe vintage equipment had a contributing effect in a past

life to the creation of the wonderful sounds on the great records of the 1960s and 1970s.

like Words round 'smooth' and "thick' have now become standard terms used when describing the performance attribute of the vintage and it is surely not by chance that they represent rough antidotes to the 'sharp', 'harsh' and 'thin' that is now equally well cliched in the description of digital.

There is also mythology associated with vintage equipment, mythology that is so strong that people feel comfortable to expound on its virtues even though they might never have used it much in the same way that you don't have to have even sat in a particular classic car to lust after it. Indeed increasingly this is the case because the true vintage is rare, probably getting rarer. and as a consequence is prohibitively expensive for most. Add to this the likelihood that of those who have used vintage outboard there will be a percentage that has encountered ragged and unserviced units and you uncover the origins of the reputation it can have for being unreliable but brilliant when it does work.

It's emotive stuff driven by an undoubted fashion element that brings outboard units in from obscurity, elevates them to hit maker, and

then either places them in an enduring favourable light or sprinkles dust on them and consigns them back to the history from whence they came.

Predicting these trends is a difficult game but there remains an unchallenged superleague of processors that are always in demand and always well regarded and these among the most desirable pieces of rack filler around

Ignorance is the biggest obstacle for anyone looking to buy vintage equipment, using the stuff is simple, and this is because you really need to know your old gear and your electronics to spot a good piece. You also have to have a good grasp of the day's market to make an informed guess on whether a unit is worth what it is asking. These issues conspire to

make buying on the open market an exercise only for the brave and informed. The rest are best advised to buy from reputable vintage gear specialists.

It should not be forgotten that there are some modern equivalents to established classics-even AMS Neve is now remanufacturing the 1081 mic pre EQs after a lay off of many years-and anyone seriously considering vintage originals who has not investigated the modern alternatives is not being scientific in approach.

Vintage outboard

Not just a polite word employed to mask the obsolescent and the unusable, 'vintage' generates a very definite response when used in connection with outboard processing. Studio Sound asks about the realities and practicalities of ownership

> A point worth making is that the use of transformers contributes to a lot of vintage gear's sound, an incredible irony as it was the move away from transformers in the search of sharper transients that consigned much of the old generation equipment to the skip and from there to vintage status in the first place. Still it now seems criminal that so much equipment was trashed, dumped and broken apart in the name of progress. Perhaps there is a lesson in this. Analogue tape anyone?

> Studio Sound asked for the comments and experience of three UK-based vintage specialists: Funky Junk with its large service facilities and as one of the most active operators in vintage gear: used equipment specialist Sounds Incorporated; and used gear dealer and now

TL Audio manufacturer Tony Larking Professional Sales.

Mark Thompson, Funky Junk: 'We're slightly different to others in that the only equipment that interests us is equipment that does a job and does it well - we're not in the antiques business. When we sell vintage equipment we have engineers go through it, set it up, line it up, replace capacitors, so that it's working to modern standards. We don't sell Hitler bottle mics because they're a load of crap and not usable. We have stuff like that in our museum.

With commonly known products like the Fairchild and Pultec the essence of the sound lies in the output transformers rather than the valves. Therefore if you get a Fairchild with stuffed output transformers it is no use to man nor beast because you can't get the originals and you've got big problems.

'Our service department can't repair everything but we know a man who can unseal those output transformers from the epoxy and rewind and put them back together.

Buver beware is the advice because with a lot of this equipment you're paying a premium for the fact that it has investment value. That investment value is only worthwhile if it is in good working order and completely original. The moment a part is replaced with something non-original, while it may still be perfectly good, its investment value goes. People should only buy through a reputable outlet that will give a warranty for the unit's condition and its pedigree and authenticity there's a lot of stolen stuff about, for instance - and will give a service backup. What is the point in spending \$10,000 upwards on a Fairchild if it breaks down ten days later and there is no service backup?

The pool in the world of units like the Fairchild is not

that large: they made less than 1000 mono compressors and about 3000 stereo. We have this vear sold five stereos and four monos and that's probably a lot more than anyone else as there may perhaps be 20 or so changing hands in a year, which is why they attract such a premium.

Performance-wise there are other things –EAR and Fairman—that compare. We do AB tests for customers with Fairchilds against the EAR compressor. Half can't tell a difference, half can hear a difference in the bottom end and of those half prefer the Fairchild and half the EAR. Tim de Paravicini at EAR hand winds all his transformers and it's what sets his stuff apart from the other tube manufacturers.

There are a lot of things that people haven't heard of like the Gates com- >>>>







Studio Sound December 1997





<<<< pressors, a contemporary of Fairchild, and certain models are beginning to attract a lot of attention. The Lexicon PCM42 delay is attaining classic status in that they're going for horrendous prices even though they're a digital device.

You have to separate the myth from the reality. A lot of it is fashion. We have a separate department that specialises in vintage keyboards and two or three years ago there was a big vogue for many of the earlier analogue keyboards. That has pretty much ended now. Part of that is down to the manufacturers wising up to the fact that people were after these sounds and produced modern units that gave a very good approximation but with modern facilities and specs. We re-just about to get involved with the Fairman range and we're sole stockists of EAR in the L'K which performs as well as the originals. It may well be that we are reaching a peak with the vintage stuff.

The list of most desirable units reads Faurchild, Pultec, the original Langevin stuff, the original Telefunken (or

Siemens) VT2 and VT6, and some of the Roland 3 Series rackmount stuff which includes the Dimension D. phaser, flanger and vocoder. You'd also have to include the early Neve EQs and compressors but that ties in with what I was saying earlier as AMS Nevel has started to remanufacture the 1081. I don't think that will effect the value of the originals because of their investment value. The new one may do the same job and sound similar, but people will always claim that the originals sound better.

One of the things that concerns me is that we do get units through from top studios and guys who swear blind it's the best thing since sliced bread and you put it on the bench and the valves are gone. The sound is appalling but they've been shoving vocals through it because it's a Fairchild. We've seen an LA2A in which the opto-isolator had gone and therefore the thing hadn't actually been doing anything at all even though the meter claimed it was limiting. All the guy had been doing was warming up the signal a little bit by valve.

A lot of it is bull, with some people going to mad lengths to use all this wonderful equipment and then mastering onto DAT. You do need a decent mic preamp and to me the Neves are the best I know of. People come in saying they want to warm up their sound and we tell them to go out a buy a nice second hand analogue tape recorder, a Studer B6⁺. A80 or even a B⁺⁺ and master on that. If it's properly set up you're not going to get noise, run at 30ips or 15ips with Dolby A and it's likely that that will give you the degree of warmth you're looking for.

We spend more time trying to persuade people not to spend money on the esoteric stuff. Yes, if they know what they want or are after a particular thing then we'll lend them stuff to try and if it does the job then fine. The reality is that getting the sound that Led Zeppelin had in 1972 is not necessarily a question of spending $\pounds 25,000$, it's a question of having the song, performing it well, and the ears of the guy behind

We've seen an LA2A in which the opto-isolator had gone and therefore the thing hadn't actually been doing anything at all even though the meter claimed it was limiting tion of naving the song, perid the ears of the guy behind the desk are always going to be the most important investment. It may well be that sorting out your patchbay, mastering analogue, good mic pres, for the sake of £300–£400 is going to give you the result you want.

If vintage equipment is properly maintained, the valves are changed when they need replacing, if capacitors are changed on discrete stuff, then it's incredibly reliable. It was built to a standard, not a price and the fact that it's still here 30 years later is a testament to its reliability.

But if you were to buy a \$300 TL Audio Indigo valve compressor there's no reason why it shouldn't still be working in 20 years as long as it's been subject to proper routine maintenance."

Nick Ryan, Sounds Incorporated: "Fashion is more than 50% of it but somethings tend to stay in for longer. Neve 80 Series EQs and mic amps are always popular as are Pultees and Urei 1176s, and 1m finding more >>>>



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<<<< people asking for Telefunken valve mic amps. Then there are other items that people ask for but don't realise what they're going to cost.

Virtually everybody is interested in vintage gear because even someone with a Yamaha

02R can make it sound a hell of a lot sweeter if they put the outputs through a couple of V^2 Telefunken mic pres. It brings up the low frequencies and seems to put the bits back where they ought to be in the upper end.

Many buy vintage gear to make digital sound more acceptable. They buy the old stuff for "rounded" sound which they associate with quality. We've all listened to the old records and asked how they got those sounds. We know it had to be the partly due to the equipment but, of course, there were the



players, the acoustics and the engineer. People forget that in those days there was no such thing as outboard equipment, you had a desk, compressors, fairly basic EQ, and if they wanted echo they used plates, natural room reverb or tape. That approach made the engineer concentrate rather than be confused by a plethora of effects. You have a fantastic choice available these days and by taking away that choice you are forced to concentrate on the essentials.

There is interest in old delay and effects units but people soon find that they are actually quite limited and suited best to certain tasks. These were early attempts at technology that has since been bettered whereas compressors have always attempted to do the same sort of thing but come in different flavours.

Plates are not in great demand because when people discover how wonderful they are they then have to be fortunate enough to have the room to accommodate them. They can vary very widely depending on how well

they've been looked after, cleaned and how well they're tensioned. I've worked in studios where we had three identical plates in perfect condition and all sounded quite different.

From the serviceability aspect many of the biggest problems exist with more recent equipment like early Lexicons and Eventides as the chips are no longer readily available. Anything that is discrete is fine but

anything that's combined into an EPROM or op-amp, you can be well and truly stuffed. The old mer am interface empfortable with

The old gear can interface comfortably with modern studio equipment and levels. One of the most important points is that while the major manufacturers like SSL and AMS Neve produce very high quality products, they are nevertheless built to a budget because everybody has to watch their budgets. Although the mic amps on those desks are very high quality they do not compete with the specialist items.

There are some very high quality modern Class A discrete and valve products around as well as state of the art solid state stuff. If you're going to AB then you have to recognise that the new stuff will be quieter because. even if you're following the old-style circuitry. advances have been made in improving noise. You also have to be aware of the cost because some of the vintage gear is going up, but it gives you the sound you want and when it comes to selling it the chances are you'll get your money back. That's a bonus these days.

In terms of future classics, units like Tube Tech spring to mind, Basically anything that is built with quality and soul. Budget units are likely to always remain budget units.

The list of most desirable units includes the Neve 10⁻³. Pultec EQP1A. Telefunken V72 and V⁻⁶, Fairchild 670 (if were looking without budgetary constraints), and the Teletronix LA2A.⁺

Tony Larking, Tony Larking Professional Sales: There isn't a limitless supply of the old gear around and there are certain things that everyone wants to get hold of but can't. We do a bit of used gear now but it can be so hard to find.

The five most desirable pieces of vintage outboard are the Fairchild 670; old Neve EQs that have been repackaged like the 1081s; Pultec EQs; the dbx 160 is always sought after; and, of course, TL Audio EQs!

With regard to effects units there are certain things that come back into fashion and then the price rockets. I remember years ago we col-

lected loads of stuff from Mike Oldfield that he'd finished with, like keyboards and effects, and we had a lot of trouble selling them yet now they'd be worth a fortune. It's very trend driven.

'A very important point with used equipment is that anyone will sell it to you even though they're not a specialist and might not know a lot about it, so it's always worthwhile buying from a reputable dealer.

There are many myths. People talk about classic gear and classic consoles such as the Trident A range. The thing is that Trident only ever built 13 of them. In those days the markets were much smaller.

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A Harman International Company Copyright 1997 JBL Professional From the confluence of R&B and rock, Barry Beckett has had a hand in dozens of seminal rock recordings and ventured into country. **Dan Daley** finds that the way to be prolific is to take it easy

ONSIDERING THE RECORDS he has produced and played on in a career that spans four decades -and counting-Barry Beckett's office is relatively free of the sort of buddy photos that adorn Nashville's Music Row offices. There are a couple of shots of him and Bob Dylan at a formal dinner table. a young Beckett looking somewhat uncomfortable in a tux and somewhat envious of Dylan, who looks relaxed in a knit Dred cap and jeans. Beckett's hair, moustache and the neatly trimmed thatch beneath his lower lip are all silver now, but he still has the natural modesty that comes with a Southern upbringing, and neither his walls nor his demeanour would suggest that he has a discography that would put this writer well over his word count, including Bob Dylan, Paul Simon, Etta James, Bob Seger, Dire Straits, Glenn Frey, Jason & the Scorchers, Joe Cocker, Elton John, Lynyrd Skynyrd. Phish, and a host of country acts from his current Nashville base, including Tammy Wynette, Neal McCov, Lee Roy Parnell, Hank Williams Jr and Alabama. And if he had to sum up all of it in a single phrase, he save they all have, that back porch kind of feel'. And that's just what an afternoon with the avuncular Beckett is like.

Beckett's career was nurtured in that small corner of rural North-Western Alabama---the Quad Cities area comprised of the little towns of Muscle Shoals, Florence, Sheffield and Tuscumbia-whose geographical insignificance belies their cultural importance. This is where producer-publisher Rick Hall sited his Fame Studios (FAME was an acronym for Florence, Alabama Music Enterprises), which produced hundreds of seminal R&B, rock in roll and soul records in the late 1950s and 1960s. The studio attracted the cream of what was a surprisingly deep pool of natural talent from the surrounding area, people like bassist Norbert Putnam, keyboardists Spooner Oldham and David Briggs, and guitarist Jimmy Johnson, who formed the protean backing bands for literally hundreds of recording artists, from the late Arthur Alexander to Aretha Franklin.

A guitar player in local bands as a teenager, Johnson started working for Hall as an engineer at the 4-track studio in 1962, but three years later he went back to the other side of the glass to join former Del-Rays bandmate drummer Roger Hawkins as a session guitarist. David Hood came to the group as a trombonist, but transitioned to bass when Norbert Putnam moved north to Nashville to open Quad Recording Studios, where he would produce Neil Young, Dan Fogelberg and Bob Dylan's Nashville Skyline. Birmingham, Alabama native Beckett, who



had been playing keyboards in various bands in the panhandle of Florida (another odd music hotbed that also spawned Larry Butler, who produced Kenny Rogers' The Gambler and became the only Nashville producer ever to win a Grammy), came north to Muscle Shoals to cut tracks with DJ-turned-producer Papa Don Schroeder, whose hit Tim Your Puppet by James & Bobby Purify in 1966 was the first of hundreds of Top 10 hits that Beckett would play on. And it was at that session that Beckett met Johnson, Hawkins and Hood, Within months, they had asked him to come to Muscle Shoals to stay, and Beckett replaced Oldham as the premier keyboardist for what had become known as the Muscle Shoals Rhythm Section.

"I remember that I tried to figure how much it would take for me to make the move," Beckett recalls, "I had a new baby, I figured that I could make it on about \$100 a week, which would have been two sessions back then."

Two sessions a week was hardly what Beckett got; the Muscle Shoals Rhythm Section is credited in the Rolling Stone Encyclopedia of rock 'n' roll as having played on more than 500 records, probably more than any other formal studio session group. (The only other potential contender is the crew at American Studios in Memphis during the same period.)

Within a year, the prolific Hall had turned his first production chore over to Beckett -The Wallace Brothers on Jewel Records. It was the start of a long production career. but one built upon the solid foundation of his decade-plus stint with what was then the World's Most Dangerous Band, '[Fame] was where I learned how to go through a song and understand it.' says Beckett, 'It was where I learned things like the numbered music chart system. The sessions were done to +-track, so we tried to get the song in an entire take, even doing the lead vocal at the same time. Fame's recording room was about 15 feet by 30 feet and playing in that place day after day, we got very, very tight as a band. We all set up in the room-I was playing a Hammond MI, a baby version of the B3-and the lead singer was the only one in an isolation booth, which they would put together whenever they needed it. It was me. Roger, David, and Junior Lowe on guitar and sometimes Jimmy would play guitar, sometimes he would engineer, depending upon what the song needed." >>>>>

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Left to right: Bruce Hinton chairman of MCA Records; JP Pennington, artist formerly with Exile; and Barry Beckett

<<<< Beckett says that he was not a technically inclined producer then, nor is he now. Rather, he relies on engineers to handle those chores while he concentrates on the music and the artist. It also allows him to play a lot on sessions he produces, something he says he tries to do as often as the situation allows. He does, however, like to mix himself, an art he learned while working sessions for a trio of producers who used Fame often: Jerry Wexler, Arif Mardin and Tom Dowd.</p>

Jerry, Tom and Arif had been coming to Muscle Shoals for a while: they knew about us, and we all learned a lot from them,' he says. What influenced me more than anything 1 think was learning to mix records myself. Tom taught me panning, but he also showed me something else: how he could make a mix more listenable by taking the stereo mix down to mono. When you take the stereo mix down to mono, the instruments that you've panned left and right will tend to jump off the mix, which is very useful thing considering that radio was still pretty much mono in those days. So we always did a mono version of the mix. What you had to do was to set the lead vocal back a bit from where it would otherwise be, and if it sounded great in mono, the stereo mix was guaranteed to sound anazing."

Jerry Wexler taught Beckett something less tangible, but just as useful. Jerry showed me the power of the song, and how to make an artist comfortable in the studio, and how to get them to interpret a song,' he recalls. He taught me the psychology of the studio. Jerry needed Tom and Arif to communicate with the musicians on their technical level, but he was the one who could communicate directly with the artists like no one else. And I consider myself lucky to have seen him doing that and learning something about that.'

The Muscle Shoals experience was the equivalent of a shingle hung outside an office door. Beckett's dance card as a producer rapidly filled up with artists looking to capture the magic of Muscle Shoals. Each of the four session musicians went on to become producers (Johnson made records with the Amazing Rhythm Aces and Levon Helm from The Band: Hood for Wayne Perkins: Hawkins for Canned Heat), and each often would use the others on sessions and credit them as coproducers. Paul Simon's 1973 There Goes Rhymin' Simon was one such collaboration. Beckett tracked it at Muscle Shoals Sound-the studio the Section had opened in Sheffield after they left Hall's orbit-using his Section-mates on the basics, then brought it to New York City and A&R Studios, which was then part-owned by Phil Ramone, who engineered the overdubs. The record spawned two hit singles: She Loves Me Like a Rock' and 'Kodachrome.'

"I remember 'Kodachrome' well because it was so different than anything we had ever done before,' says Beckett. 'It was a real challenge to come up with parts for that."

Bob Seger's Beautiful Loser in 1975 was the first record the Detroit-based artist would do



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with the Silver Bullet Band. It gave Seger a regional hit in 'Katmandu'. But the Beckettproduced *Night Mores* three years later put Seger over the top, with hit singles in 'Mainstreet' and 'Rock and Roll Never Forgets', as well as the title song. Beckett was also there in in 1980 for *Against the Wind*. Seger's only No.1 album.

Beautiful Loser was done at Muscle Shoals Sound, co-produced again by Beckett and the Muscle Shoals Rhythm Section, which traded off tracks with the Silver Bullet Band. Beckett was particularly pleased to work with the maniacally perfectionist Seger, to the point where he says. I remember that my fingers were in constant pain anytime I worked with Bob. That's how hard we played. It was a great way to let off steam and pressure. We played that record and the others as though we were in a live situation-the 5-piece band set up on the floor, and Bob playing and singing. We cut Night Mores in Miami, at Criteria. Funny thing was even though it was a very piano-based sound, the piano that we used at Criteria was kind of a joke. It was very small and hard to keep in tune and we had to keep the lid shut on it and mic it from within to keep leakage out. We had to use a lot of EQ to make that piano sound as big as it does."

Beckett had been making his records at 15ips without noise reduction, a practice that dated back to the Fame days. At one point, he and engineer Jimmy Johnson experimented with 30ips, but Johnson suggested they go back to 15ips to keep the low end punchy.

We were making some pretty noisy



Left to right: Paul Nelson, songwriter at Sony ATV-Tree Publishing; Barry Beckett and Matthew Beckett, his son and professional manager of the Beckett Music Group

records by today's standards, but we didn't care.' Beckett says. 'Tape hiss was the glue that held a lot of those records together. I used to play on a Wurlitzer electric piano and it had this huge hum. It drove the engineers crazy, but I didn't care—that hum was part of the sound to me.'

In 1978, Beckett worked with Joe Cocker on Luxury You Can Afford and the next year with Bob Dylan on Slow Train Coming. In both instances. Beckett was faced with eccentric vocalists and put to good use the lessons he had learned from Jerry Wexler a decade before.

'With both of them, you had to get them on a single take,' says Beckett, 'especially with Dylan. The timbre of his voice changes throughout the song. We were getting maybe two, three takes the most out of him, and we found that we couldn't splice parts of vocals from one take on to another because his vocal sound changed so much. But Dylan was fast. We usually got the song done in a couple of takes, which is the way I prefer to do it anyway. That's part of the magic we had at Musele Shoals. With Dylan, I found myself just sitting back and steering him very little. But it still took Bob three or four days, while we were cutting, to even trust me. When he finally did, he let me know with a very slight smile, or something like that, to say I was okay.

Beckett had come into the Cocker project with the basic tracks already done, by Allen Toussaint, who had left the project in midstream due to creative differences.>>>>





VI-1 8 Channel Valve Interface

EQ-1 Dual Valve Equaliser

PA-1 Dual Pentode Valve Pre Amplifier

C-1 Dual Valve Compresso





Barry Beckett with RCA country artist Lorrie Morgan

'He was an awesome singer,' he recalls, Tve never heard a singer that good. As strange as he might seem on stage, he was very straight in the studio. While he was singing, he would look straight at me in the control room and I could never tell if he was uncomfortable or not. On one song, he broke into a vocal that was so incredible that I just sat there listening, enthralled. And I forgot the structure of the song as it was going by. So at one point in the song, he just stops singing and stands there, looking at me. I don't exactly know how to react. He's just standing there, not singing. I stopped the tape and asked him what was wrong and he says, "Nothing. That's just where the solo is". To this day, that's my biggest weakness as a producer. I think: being nervous in front of the artist. Because if I'm nervous, I can't put the artist at ease and get the performance I want out of him.

Dire straits was as much a vehicle for its guitar-playing guru. Mark Knopfler, as it was a true band endeavour. On *Communique*, produced by Beckett and Jerry Wexler at Compass Point Studios in the Bahamas. Beckett was aware that Knopfler was the dominant musical force in the band, but seems to appreciate that Knopfler didn't unduly impose that dominance during the making of the record. Rather than being emboldened by the massive success of their first record, which was made in 19⁻⁻ in 12 days for \$25,000, Knopfler might have felt the sophomore pressure that often follows successful first ventures.

Mark Knopfler was always trying to make his guitar sound and playing perfect. says Beckett of the record. The whole track would

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track was good or not, so you just had to stay with him until he got his playing and his sound to where he wanted it."

The Communique experience, perhaps, tested Beckett's limits, if not his limitations. It underscored to him how used he was to being in control of the music tracks, a state that harkened back to the Fame days when the same session players guided every track. But through the early 1980s, Beckett watched music change. Even Muscle Shoals had now become a disco haven. Phil Ramone asked him to come to New York and work with him there, but Beckett disdained what he calls the drug scene of New York in the Yuppified early-1980s. LA was equally uninteresting to him because he saw it becoming a postproduction city where sound supported film and television instead of music. Surprisingly, for an Alabama boy, considering Nashville as his next move seemed to be result of a process of elimination rather than a conscious decision.

We always tried to play so un-country in Muscle Shoals," he says. We didn't care for that safe, conservative approach to making records they had up there. We were usually busy making fun of it."

But Nashville was changing: younger, edgier country artists like Rodney Crowell and John Anderson caught Beckett's car, and took some of the sting out of making the move to Nashville, which he did in 1985. Nashville's semi-feudal hierarchy was initially opaque to Beckett: membership in the Nashville club was not a Southern birthright. But he did not have a problem literally starting over, which is essentially what he did, playing on song demos for writer Bob DiPiero and others as his entree to the Nashville system. The strict

revolve around his guitar playing. He was always concerned about his sound. Mark is a very touch-type player. Every little nuance about the sound and the feel has to be right. It takes time to get those things right. And me, I'm a spontaneous sort of producer. I like to get things in the first couple of takes. In fact, if you can't get it in the first few takes, then you're probably not going to get it. Most artists hit their stride on a song in that time. Right or wrong, mistakes or no mistakes, that's where the magic in a track is. The moment is gonna happen on the second or third take every time. So the problem for me was to keep Mark magic through all the takes. The thing is, Mark believes that how he played would determine whether the

traditionalists who were still pretty much running the business in Nashville were not about to hire a rock 'n' roll piano player to produce their records,' he observes. 'But I knew that there were people in Nashville and in country who liked rock 'n' roll and that it was going to be part of country music at some point or another.'

Jim Ed Norman, a Los Angeles producer who had migrated to Nashville several years earlier and was now running Warner Bros Records there, was responsible for bringing a number of other LA expatriates to Nashville, including producer Jimmy Bowen, who would eventually run Capitol Records there and who brought digital recording and loud drums to country music with an evangelical zeal. Norman gave Beckett two things: an A&R job at Warners and Hank Williams, Jr, a rowdy redneck who was a perfect fit for rocking up country music. The pairing worked, garnering Williams a CMA award and giving Beckett the kind of credentials Nashville requires.

While his discography has become predominantly country in recent years, the artists tend to be soulful ones, like Delbert McClinton and T Graham Brown. But Beckett has kept his *bona fides* in the pop camp, as well, producing Grateful Dead-*manque* Phish's second record, an experience that seemed to leave him almost bewildered.

"I didn't know much about them except that they spelled their name weird," he says, sounding every bit like a member of Nashville's own aristocracy. But that illusion lasts only a moment. "I went to hear them play in a theatre up in Burlington [Vermont] and the place was packed shoulder to shoulder and filled with marijuana smoke and I said to myself. I haven't seen anything like this since the 1970s! This is great!"

Beckett was initially confounded by Phish's tenuous song structures, which often include a verse and chorus only as dressing for protracted solos. I knew that, unless I had the most amazing engineer on earth, and did lots of looping and editing. I would not be able to make a single with them, and I told them and the A&R person that,' he says. 'And they said let's do it, so we did. And there was no single, and that probably hurt the record's sales a bit. But they're the kind of band that builds itself steadily, album by album, not single by single. It was an experience.'

Beckett travels enough, preferring to land in Tom Hidley rooms wherever he goes, and prefers vintage Neve consoles, like the Flying Faders-equipped 8068 that he used at Muscle Shoals Sound (Fame had a Universal Audio console); although he admits he prefers the 'convenience' of SSE's automation system. He has started two new independent record labels, one for blues-rock artists and one for country, and while he would not dismiss the idea of studio ownership once again, he states flatly. 'There's just no money to be made in the studio business. He splits his engineering work in Nashville between Pete Green, whom he uses for the more bombastic projects, and Csaba Petocz, whom Beckett calls more nuance-oriented'.

1 defer to engineers technically. Always have, But I still trust myself when it comes to the songs and the artists.

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on **BASF** tape

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Richard

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The 1997 MTV Europe Music Awards were held on 6th November at the Ahoy Stadium in Rotterdam, the Netherlands and broadcast live to an estimated audience of one billion. Recipients of this year's baubles included Jon Bon Jovi, Will Smith, The Prodigy (who picked up three awards) and Oasis, with Backstreet Boys picking up the Select Award for being the choice of the viewers on MTV's interactive juke box selection (see side bar for full list).

As if in answer to Mark Knopfler's barbs about the MTV generation, the station proved that it has a social conscience with the Free Your Mind Award, designed to highlight humanitarian issues and encourage people to break away from intolerance and prejudice. This year the award was presented by counterculture cult hero Dennis Hopper and went to the Landmine Survivors' Network. It was accepted on behalf of the organisation by Plamenko Priganica, a Bosnian landmine survivor.

Now firmly established in the roster of such ceremonies, the MTV Europe Music Awards are unusual in that they are broadcast live; even nine years on from the infamous UK Brits when everything fell apart, many organisers and broadcasters are wary of leaving so much open to Murphy's Law. But the satellite station courageously ignores both the dictum 'If anything can go wrong, it will' and O'Toole's Observation ('Murphy was an optimist') and relays the whole event as it happens, recording it as live for both its own later transmissions and those of terrestrial services around the world (which usually air it a few days later).

Despite the trepidation with which such events are viewed, those involved agree that there is a basic formula by which they are done. This is very much the view of Andy Rose, who has worked on the MTV Europe Music Awards since their inception. He takes the role of supervising sound supervisor and in this capacity puts together the facilities and crew for the audio production. Each year the show moves around Europe, with a different host nation each time, something that, to a degree, dictates the equipment and personnel selection. 'We always try to use the local facilities of the country,' says Rose, 'it's a kind of detente.' But he adds that this European fraternity has to be tempered with expediency, using facilities that offer everything to do the job, which often means taking them from wherever they are available.

In this instance, Dutch public service broadcaster NOB provided the vision scanner, which also houses a 48-channel Studer console that was used for the overall production mix. Although NOB has a SSL sound truck and there are numerous recognised music recording mobiles in the country. Rose specifically wanted mixing desks that were assignable and resetable. Twe worked in NOB's sound truck and it's very nice but it doesn't have the recall options that you need for a something like this—and you have to have two trucks for these events. Resetability is vital because the way we work is to record the rehearsals of the live acts, spin those back and rehearse In the interests of internationality, Europe's premier music awards show tours the continent, making each event a mix of consistency and chaos. **Kevin Hilton** signs in



the mixes, storing the various settings.

With this in mind, Rose settled on BBC OBs' 88-fader Calrec assignable MSC3 truck (which worked on the Awards last year when they were held in London) and the Euphonixequipped Fleetwood mobile. The British bias continued with the choice of live sound company, an important element, not only because the event is staged before an audience, but because there has to be good communication and understanding between those working in the house and those in the TV trucks. This year Rose contracted Detrick Zieba's Dimension Audio, which is experienced in such television-orientated events (and worked on this year's BRITS), sub-hiring in a sound >>>>>







<<<< system from Britannia Row.

This show is very music heavy," observes Rose, 'and so it is useful if there is an overall sound supervisor as the single point of contact. I worked closely with Bob Lopez, the music mixer for Brit Row, because having that kind of interface is very important—everybody has to know what is going on and when it is going to happen." The stage crew also included Dave Morton and Steve Blinco on radio mics. We decided that we needed 12 wireless mics on the show," says Rose, which meant that three could break down but we would still have the lead vocal on the same channel."

N MSC3. Rose was the main music engineer, working with the BBC's Simon Scrivener and Gary Stewart, while Keith Mayes was primary mixer in the Fleetwood. assisted by Tim Morgan, along with the mobile's Tim Summerhaves. Mayes confirms Rose's decision to opt for resetable consoles. saving. There was only a potential of ten minutes between bands and there's no way you could work it by just writing everything down at rehearsal and then setting it all back up again on the night. What you need to be able to do is have everything come straight in, get it on tape leach truck housed two 48-track digital machines] and then spin it back and rehearse the mixes during down-time. This gives the opportunity to fine-tune things and then store it all."

The set and technical areas for this year's Awards were rigged on the Monday, rehearsals were set aside for Tuesday and Wednesday, with Thursday being show day, plus a dress rehearsal. Ten acts were lined up to play on the night, nine of which were completely live (although some of the rap acts did have backing tracks mixing in with the onstage instruments), while the Spice Girls had live microphones. The first day of rehearsals were used to rehearse all the bands going through MSC3, while Tuesday was Mayes' day.

Of the 130 active splits on stage, the trucks were sent 40 lines. These were then evened out between MSC3 (designated $\langle X\rangle$ and the Fleetwood ('B'), with 42 splits each. Additional compressor and reverb units were installed in both mobiles, meaning that each had pretty much the same equipment. Andy set the style for the show.' says Mayes, 'which mainly meant the way the audience reaction was treated and how much was used. The three mixers [Rose, Mayes and BBC TV OBs] sound supervisor Tim Davies in the NOB scannerl had access to the same audience mics. The house style had to dictate whether the reaction was to be open or close miked: there's no point doing it differently. This time round we went for an atmos style."

On the all-important subject of audience



response. Rose adds. The way I prefer to work with applause is to treat it as part of the band sound. Keith and I would use our feeds while the bands were on

and Tim would just put up his two faders. At the end of the songs, everything would go over to Tim and there would be a crossfade, with the idea being that nobody would be able to tell the difference.' Tim Davies confirms the approach, adding, 'We all had a feed of the same audience mix but I had channels of my own 'canned' audience effects [played in by grams operator Rick Simon from an Akai DD1000] and two mics over the bear pit at the front of the stage where the die hard fans were. It was relatively simple.'

The atmosphere-sound style extended to the music, in keeping with TV director Julia Knowles' decision to shoot the show using big framed shots. The style was also influenced by the varying requirements of the bands and in particular, the opening act. U2, with whom Rose has worked eight times this summer during their Pop Mart tour. They have a very ambient sound anyway, says Rose, and that's what the band wanted. Bono in particular was keen on the big, ambient, all-round sound.' Mayes adds. The very open sound worked well with someone like the Backstreet Boys as

The winners

Janet Jackson

Best live U2

The Prodigy

Best female artist

Best rap Will Smith

Best rock act Oasis

Best R&B Blackstreet

Best dance The Prodigy

As Long As You Love Me

Free Your Mind Award The

Landmine Survivors' Network

Best video Breathe,

Select MTV Award

Backstreet Boys,

Best group Spice Girls

Best male artist Jon Bon Jovi

Best song MMMBop, Hanson

Best alternative The Prodigy

well because it picked up the excitement of the kids in the crowd."

Mayes says that the director's choice of shots to a certain extent dictated what elements were given prominence in the mix. We had to cheat slightly in some cases," he says. With Blackstreet. Slash was the featured guest artist and the director used a lot of shots of him and so we had to favour him more in the mix.' Rose adds, Bjork likes to have her vocals quite low in the mix but because of the shots used, we mixed her up and made sure that you could hear the words more. It's a bit of a compromise sometimes.

Of band influence on what the mixers did during the show. Rose says, 'All the bands had as much input as they wanted. Most sent a representative along to sit in the truck during the performance and they would pass on comments, pointers and suggestions on what effects they wanted. A few of the artists popped in because they wanted to see the mobiles. The only ones who came in during the rehearsal mixes and contributed were U2.





who are very into the way things sound."

Even though playing live, some acts incorporated backing tracks into their performances. These were played in from a

SADiE DAW, with a backup version held on DAT, with the two running as much in sync as was possible.

The start buttons were hit at the same time,' says Rose, 'and if the SADiE had failed for whatever reason, the DAT would have been faded up. There would have been a slight gap, but at least it gave us something to fall back on in the event of anything happening.' If anything had gone wrong. Rose said that everyone would have had to cope with it but on the night, everything worked. The show was totally tight,' he says, 'although it did over-run but that's MTV's decision.'

While saying that everything worked for the live show, Rose and Mayes admit that the dress rehearsal did give them a few heartstopping moments. Unlike other 'live' awards shows, these do not receive any audio postproduction afterwards: if there were any repairs needed, these would be done in the trucks rather than in a sound dubbing suite. As well as the recordings of the rehearsals, while one truck was on air, the other recorded the output onto 48-track.

The stereo output of the two mobiles was fed to the NOB scanner, coming up on the 36-channel Studer desk. Tim Davies also had the audience feeds, the grams (which additionally included two-MiniDisc players for walk-up and walk-down music and stings), eight hand-held stage mics in case any of the acts wanted to shout to the crowd after their appearances and a hand-held mic for Boyzone. lead singer Ronan Keating, who handled presentation duties, presumably after his success in this capacity at the Eurovision Song Contest. There was an additional Beta 57 for a voice-over artist, who was positioned by the side of the stage, which caused some coloration on the feed, but created the vibe the producer wanted.

Davies, who was assisted by a NOB engineer, ran everything through Neve compressors and limiters, with the complete output running via a Finalizer, of which he says, T like the sound it makes. It gives 3-band compression and limiting and makes for a punchier sound. From the scanner the production mix of vision and audio was fed to a near-by up-link vehicle and beamed back to MTV's studio complex in London.

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Gregorian MOODS

Setting up in an abbey to record a selection of Gregorian chants presented numerous challenges to the Fleetwood crew—all of them complicated by a television broadcast. **Simon Croft** reports

AST YEAR, the Benedictine monks oF Downside Abbey made it to the No.1 slot in the British classical music charts with an album of Gregorian chants entitled simply The Abbey. The follow-up album, Gregorian Moods, hit UK high streets on 10th November this year, accompanied by a full-scale television advertising campaign from Virgin Records. There is also a BBC television documentary, to be screened 21st December. Part of the Everyman series, it covers the whole project from a number of perspectives, including the recording from which some of the broadcast soundtrack is derived.

Viewers are in for a visual, as well as musical, treat. The word 'picturesque' hardly does justice to the Abbey, set as it is into the countryside of Somerset; arguably some of the prettiest England has to offer.

Purchasers of the audio CD need hardly feel short-changed, however. As with *The Abbey, Gregorian Moods* features alternating tracks from the monks performing plainsong and a boys' choir singing in polyphony.

All the material was recorded, mixed and edited at Downside Abbey by Tim Summerhayes and Ian Dyckhoff of Fleetwood Mobiles. The pair were also responsible for



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recording *The Abbey* album the year before, as a silver disc on the office wall confirms. Although the experience proved invaluable, they decided to take a slightly different tack this year, as Summerhayes explains.

The Abbe was recorded on Mitsubishi 32-track, but we thought we'd push the boat out on the new one,' he says. The recording was technically 8-track audio, but it occupied 16 data tracks on a pair of Tascam DA-88s. Prism convertors were used to split the data across a pair of tracks for each channel of audio signal, in glorious 24-bit resolution.

For safety, we used a third DA-88 to record the eight audio tracks through its own internal 16-bit system. As a result, we were able to A-B the 24-bit and 16-bit systems, Summerhayes notes. There was a marked difference.

The low-level, high frequency detail was quite awesome,' says Summerhayes, 'really very, very good on the 24-bit. I am not knocking the Tascam system, which is great for normal stuff, but in this situation, you could hear the difference. His opinion is supported by Ian Dyckhoff, who edited the material on a SADiE system, and also Simon Hayworth of Chop 'Em Out, who was responsible for the mastering (see side bar Mastering Gregorian Moods').

Before we enter into a discussion of the technicalities of the recording, Summerhayes and Dyckhoff recall some of the more idiosyncratic aspects of the *Gregorian Moods* sessions: As you go into the crypt for the power, the doors creak open, says Dyckhoff, "to reveal all the coffins and casks down there."

He says the experience was very strange', adding, 'The power was okay, it must have been phantom power!'>>>>







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The mastering of Gregorian Moods

Simon Heyworth, director and chief mastering engineer, at Chop 'Em Out recalls, 'The only difficult thing was the ambience because there wasn't really enough of it (in terms of duration). I think they couldn't leave the mics open as long as they would have liked.

'It was the end of term and the boys would tend to make a noise before the ambience had really finished, so I had to construct it out of what there was. Other than that it was fine: It sounds great.

'I used the Sonic Solutions. It is popular for mastering because it sounds better. Whether 24-bit sound better in the end depends on the material and if dithering or noise shaping is applied. I think I used either the Sonic Solutions Turbo Dither or Super Bit mapping.

'Turbo Dither is not as simple as pushing a button. There are a number of parameters you have to adjust and there is the question of whether signal level or bit resolution is the criteria. I prefer bit resolution, which is slightly different.'

Heyworth also points out that the quality of 24-bit recording depends greatly on the A–D convertor. He transferred the edited master to 1630 format. <<<< There was actually a power cut, caused by a thunder storm in the middle of the week, when the whole village lost its electricity for about two hours. So it was back into the refectory for more teal, recalls Dyckhoff with some affection.</p>

We also had to work around the monks' various services.' Summerhayes explains. They gave us a timetable and there was no way we could infringe on those. We would just leave everything set. Although it was open to the public, it was fairly secure.'

Both choirs were approximately 20 strong, although it varied slightly from session to session. The monks' choir was headed by Father Dunstan O'Keef, while the boys' choir was led by David Lawson, the head of music at Downside Abbey.

Dyckhoff points out that the boys were actually '20 strapping lads' who were not adverse to a night on the tiles, as it was out of term time. 'The next day they would be lying around, all black coffee and sore heads,' he remembers.

They were very disciplined when they were working on the actual sessions, says Summerhayes. I think David Lawson put the Fear of God into them—literally.

Summerhaves makes no bones about the fact that his first attempt at a miking plan during the recording of *The Abbe* last year was a bit of a disaster: 1 put in



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what I thought would be the right rig and it was appalling, he admits. Fortunately, that was just a test day so we went back and worked out Plan B, which went quite well. Using basically two pairs, we captured the diction and the presence, as well as the ambience with a pair of outriggers.

This year's recording used a miking scheme evolved from the lessons learned last year.

Two sections of the Abbey were used for recording: the north wall of the choir area for the monks and the central section for the boys' choir and the organ (see diagram above). All microphones were on stands, rather than suspended. There were two rows of monks. Summerhaves recalls. We had a pair of 87s covering the two rows of pews, or misericords to use the correct term. For the soloists we used a U47.

We bunched the monks together as close as we could. We had another pair of distant mics [Neumann KM1408] by the altar looking into the choir area. For the big reverbs we used the distant mics that were for the boy's choir, in a different location in the nave.

The choir was about halfway down the second section of the nave, against one wall. We used that solid wall behind them to reflect the sound. We used three close mics, Schoeps CMC5 cardioids. We also used a stereo Schoeps at about 20ft-25ft. That was the main microphone. The close mics were used simply to get some definition. And we used two distance mics [AKG +1+s] at about 30ft along the nave, to get the reverb, which was awesome. I don't know what the reverb time was, probably seven or eight seconds. It was a fantastic sound."

The location of the choir was in some ways a slightly grisly choice, situated as it was next to the three lead boxes of the remains of Oliver Plunkett, the Archbishop of >>>>













From the top. 1: The choir assembles. 2: David Lawson and the choir. 3: Right, production engineer, Tim Summerhayes with Mike Christie, executive producer. 4: Prisms for 24-bit.

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The conductor was between the choir and the organ console,' Summerhayes continues. 'The choir was in a semicircle around him in three rows. We had a spot soloist mic in the choir, That was a Russian Mikrofon MKE-13M that we had an demo from Ravencourt. The guy who was doing the sound for the Everyman documentary just happened to have a few about his person.

It was very quiet, very clean. It sounded a bit like a KM84 if anything. It's solid brass and you could run a truck over it—a really heavy-duty construction.

On a couple of items we had a couple of solo monks singing with the boys choir. We positioned them just by the altar and that worked out quite nicely in stereo. Again U+7s for that.

The organ was a problem at first, requiring the attention of a repairer in order to minimise squeaks and rattles from the pedals. Screens were set up to minimise noise transmission.

A shotgun microphone was initially aimed at the organ loft but proved to be unnecessary, after liaison with David Lawson, to establish the correct balance.

The signal chain was straightforward, with the microphones fed to the preamps of Fleetwood's Euphonix console and then directly to the convertors, without equalisation.

The convertors chosen were Prism AD-12+ A–D and DA-1 D–A. We originally thought of working 20-bit because I had heard that the Tascam system would do that. Summerhayes explains I phoned Tascam and there seemed to be all sorts of "yes, buts", so Ian said, Why don't we do it 24-bit?

Asking around people we

know, Prism seemed to come out on top, says Summerhayes, I had never used them before but they are considered the best by many people. They came down and set their systems up. They were very helpful indeed.

The specification of the convertors is exceptional and Prism managing director Graham Boswell states, 'No other company can better our A–D and D–A through-path for performance specifications, to the best of mv knowledge.

On the A–D side, we were the first to exploit the new monolithic oversampling chip sets in hybrid configurations to achieve extended performance. The AD-124 is quoted as having a 120dB dynamic range, while the DA-1 is quoted as having a 112dB dynamic range and a distortion figure of 0.0005%. And yes, upgrades to 96kHz operation will be available.

I get the impression that 8-track, 24-bit is an unusual way of working, says Summerhayes, although Dyckhoff notes, "Apogee has got their 24-bit, 8-track unit out now," Which would suggest a trend towards the format. Not everyone would have elected to use multitrack at all for a recording of this type.

Thever considered going straight to stereo because every straight to stereo recording Tve heard seems to leave a little bit to be desired," Summerhayes declares. Thean't see how you can do it really.

When the MD comes in and says. "That's great but can you just raise the trebles a little bit?" The answer is going to be, "Sorry, I can't". It's already gone to tape like that and you're screwed.

There are so many bad classical recordings out there. Summerhayes says, more than a little contentiously. Last year, when we did the first one, we did a fair bit of homework. I bought a load of Gregorian chant records and I just could not believe what I was listening to. One was a compilation and there was only one track out of 15 or 16 items that sounded even half decent.

Summerhayes reckons that some of the European labels that have made the most noise about their superior recording technology are responsible for some of the worst sounding releases.

I don't know how they get away with it, he says in exasperation. Technical excellence is fine, but if you don't put the mics in the right position, what's the point?

Mixing *Gregorian Moods* turned out to be a question of panning the channels to reflect the original position of the microphones. With the exception of removing the redundant shotgun microphone channel, levels were left almost as recorded

'It did pretty much look after itself, which I suppose is an argument for straight to stereo, but I wouldn't feel safe, Summerhayes notes. According to Dvckhoff, not all the participants had the same opinion of the mix. When you talk to the monks, they all want more of them in it, and when you talk to the boys, they want to be louder than the monks. Perhaps it was just as well they didn't have a drummer. Questions of balance were handled primarily by Summerhayes, David Lawson and Father Dunstan.

When it came to editing, the cooperative approach was again used. Dyckhoff used a SADiE system to edit the music, 'All the people were there and we could pick the good bits out of each take,' he explains. "Because everyone was locked into the project it was all very familiar. You didn't have the problem of getting back into it at a later date.

The SADiE is a very good system for editing that material, especially as it can handle 24-bit. Dyckhoff considers. We actually used v2.6, which is very good and very easy to use. I reckon v3 is even easier because of its rock 'n' roll cueing, which you can't do on the earlier version.'

Dyckhoff says he made extensive use of crossfading to obtain a seamless result. 'You had to use very long crossfades on some of the cuts because it had such a large reverb on it, so you can find yourself editing into reverbs.' he notes.

Editing *a cappela* choirs is very different from working with material that has a strongly defined beat, although even plainsong has a rhythm to it, despite the fact that it is scored without bar-lines. 'You have to use a vocal "ess", or some other sound, as a reference point,' Dyckhoff suggests. 'Gregorian chant isn't especially clear, and it's all Latin, of course.'

The edited material was transferred back onto two tracks of Hi-8, again to 24-bit resolution. Then the tapes went to Simon Hayward at Chop Em Out, who compiled the master over the course of two days. Recording, mixing and initial editing had taken a total of eight days—and left a lot of positive memories.

It was a great week, the first week in July. It is a beautiful place: and we were working with lovely people, says Summerhayes

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It's Christmas time and a new James Bond film has been released. But the real hi-tech action behind the resourceful British spy is off-screen, as **Simon Croft** discovers



OND IS BACK. And his new production has some of the best techno-toys a boy could wish for. We're not talking about half a kilo of stun potatoes, a watch with self-dispensing garroting wire or even a jet black stealth ship here; although there is one in the film. Rather, this Bond is brought to you via a collection of hi-tech that includes beta-version Dolby Drives and the first 24-bit Digidesign Pro Tools sold into the UK. The plot hinges on a media mogul who decides to trigger World War III in order to boost the ratings of his global empire. Cue lots of international satellite conferences using giant video walls.

Suffice it to say, the film is visually, as well as sonically stunning, with locations ranging from the South China Seas in the East, to the latest incarnation of the Bond set at Pinewood Studios, west of London—also scene of all the

Tomorrow Never Dies Music editor Dina Eaton

editing and dubbing. Chris Munro was location sound recordist. A long-time champion of the DAT format, these days he is using his Fostex PD-4 in favour of the earlier PD-2. He also recorded multitrack, as he explains: 'When I have a lot of mics up, I mix them on the fly as you would normally, but I also leave all the direct tracks onto Hi-8 unmixed so that if it has to be remixed, it can be.

We might have three or four radio mics and a boom, he notes. The multitrack he uses is a Sony PCM-800, because of the balanced connectors. It's the best value machine you can ever buy, Munro enthuses. The thing about Hi-8 machines is that they are only \$2,000. They're incredible.

Munro worked closely with computer and video supervisor Justin Owen. He was responsible for on-screen video ranging from tiny monitors showing loops of what were apparently other areas of the stealth ship to huge video walls which included 42-inch plasma screens driven by 40 or so sources at any time.

It could have been a disaster, but it's worked out very well, says Munro. One of the things that helped is that before Justin got

involved with films he worked in the corporate sector-not just video, but lighting and sound, so we can talk the same language."

The conference scenes used live actors on set, plaving against other characters who had been pre-shot with ENG-style cameras to get the right look from the video screens.

'For our purposes, for playback on set, we would only need relatively low-quality audio, purely for a guide.' Owen explains. But we would have our time code and the sound department's time code in sync with each other. Their DAT recording would then be used further down the line, for the postproduction part of the process. So they would use our audio as a guide track."

So far so good, but the setup was not without its complications. However consistent they may be, actors are not robots, so Owen needed to be able to re-cue the video and perhaps re-edit the sections, at the drop of a hat. To this end Owen, who reckons he will soon own around £200,000 of equipment. enlisted 12 hard-disk video players running on 200MHz Pentium PCs. These were a small fraction of the 80 or so computers used on set in one guise or another, but sufficiently specialised that they are almost impossible to hire. So when he needed more sources, Owen resorted to DVCAM VTRs.

A minor complication was the sound from the videos, used for cue on set, but unwanted for the actual dialogue recording.

Sometimes, depending on the scene, I would mix the sound live, to take the video sound out, knowing that we can overlay it later.' Munro reveals. 'On other occasions we used gates, so the live sound gated the video sound.

Another complication was a variable amount of delay-typically around four frames-on the live video feeds, caused by the fact it was going through a video mixer and having various effects applied. It's doing it as fast as it can,' says Owen, 'But there has to be a certain amount of processing time and so it is very difficult. And the bigger you get the more noticeable it is.

A trickier problem was caused by the fact that interlaced video monitors can show a bar across them unless their frame rate and that of the camera are in sync. Often this means cranking the camera up to 25fps, then back to 24fps for standard replay. This causes a similar situation to when the camera is run at nonstandard speed for special effect, as Munro explains,

Because the picture is replayed at 24 frames, we end up with out of sync sound." he notes. We then have to very accurately adjust the speed of the sound to the camera. That is a lot more complex than it first appears.

Before we were into digital sound and before we were into nonlinear editing systems. like the Avid, they just used to record sound onto 35mm mag, running the mag machine at 25 frames and play it back at 24 frames," says Munro. Okay, the pitch would be lower, but they would put it through a Harmonizer and get it as close as they could.

Now we can't do this because there is much greater accuracy in editing, with non-linear and time codes, he notes. So we have a DAR SoundStation Sabre as part of the sound department. We take the original sound we recorded and we stretch it exactly to the camera speed. We also pitch correct it at the same time, using the Time Warp facility. So we end up with a new master that is absolutely pitch and sync corrected.

As with many other areas of production, this correction process was done on a continuous basis by the sound department, rather than handed over to postproduction for them to sort out later. A major reason for this was the relatively short time between the end of shooting and Tomorrou Neter Dies' release; just 11 weeks.

This impacted on other areas, such as music recording and editing. The first part of the score-composed by David Arnold-was recorded at Air Lyndhurst back in May (see side bar 'Knowing The Score'), but shooting finished in September.

Stirred, but not shaken by the music is music editor Dina Eaton: 'David Arnold is the most terrific composer to work with, and, of course, that makes the whole Assistant

thing so much easier as well. music editor she says. Earlier in the year 'easier'

Dominic Gibbs

was an unlikely word to apply to the music editing. Air's Geoff Foster was particularly keen to keep the music at 20 bits. Dina Eaton and Dominic Gibbs-who has been working with her on music editing-were particularly keen to stick with the Digidesign Pro Tools systems they were familiar with. At the time Pro Tools 24 didn't exist. Problem.

Fortunately, Digidesign has come up with the goods and the pair were able to use the new 2+bit capable system at 20-bit throughout the editing process. But does it sound any better?

emphatically. We were at Air Lyndhurst saving, "what's the differ-ence between 16-bit and of Nick Dodd 20-bit", but quite by

chance there was a 16-bit version of one piece of music, so we were able to compare them, just as an exercise.

"I turned my back. Dominic played one section and I just went, "that's the 20-bit", and he said, "you're absolutely right". I was amazed. I thought I would be straining to hear the difference. It was a volume thing, a richness a "bigness" that was terrific.

Gibbs is a seasoned Pro Tools user and he is equally enthusiastic. 'It is superb,' he >>>>













Data flow and dialogue

John Cochrane (above) was dialogue editor on *Tomorrow Never Dies.* While the picture was edited on multiple Avid Media Composers with shared storage, effects and dialogue was edited on Avid AudioVision systems. As each editor tended to have a unique selection of material, it was decided that a networked approach would not be particularly beneficial.

Formerly an employee at AMS and SSL, Cochrane was involved in the decision making and testing process that proceeded the mammoth editing task.

'Going back to the beginning of the year, we declded to look into everything that could possibly go wrong. Bearing in mind the way we were going to work.

'We've done this with a common foundation of media which started at the rushes stage. Everything which was selected, was digitised into the picture Avid system and we cloned the disks. <----- confirms. 'It's very like analogue, but without any noise.'

Gibbs had only one reservation before editing began, no-one had used the new system 'in anger' before. As it was to be used as the playout format for music in the dubbing theatre, the thought of it crashing worried him a lot. It need not have.

The backup from Digidesign was great and it runs really well. It's excellent—which is pretty much the same as ever, 'he says. There are a few added functions. The SMPTE Slave Driver

So before I started work, I had a foundation of original sound.

Then, when a reel was completed they'd give me an OMF Composition, which is a bit like an EDL, but the advantage is that you don't have to go through the conforming process. It literally finds the right bits off the disk. The advantage is that it is quick and flexible.

'There were a few unknown factors compared to previous jobs. The picture Avids were going through a hardware and software change. In its own right, it wasn't an issue, but we had not worked with that combination before, so there were quite a lot of tests transferring to our systems. They are on PCI now, whereas the sound equipment is still NuBus.

'Theoretically, there was no problem, but we had to test the bridge between the two. We worked everything out down to the routine for assigning tape names, so that we never ended up with duplication or ambiguity.

Apart from that, the main concern was just the

has been replaced with the Universal Slave Driver, which means it will lock to the biphase they use in the dubbing theatre. It won't run backwards, but it probably will soon.'

Gibbs is also appreciative of the 24-track architecture of the Pro Tools system. The music tracks are mainly mixed as five orchestral tracks—corresponding to LCR and two surround—plus three tracks for synthesisers or the oriental instruments. Chequer-board that little lot and 16 tracks are already accounted for. And there has been a lot of

amount of storage that we needed. Previously, we had worked with a couple of 9Gb drives and that would be it. On this we knew we would have to clone all the media whether is was used or not. So there was massive amounts of storage which, in practice, we had never hung off the end of one of these systems before.

'Connected to my system, I've got 80Gb of storage—that's for picture and sound. As I finish a reel of dialogue here, I walk into Theatre Two with a removable drive, with just the edit data, I've preprepared myself with a duplicate set of disks in the theatre. The effects premixes go a slightly different route: onto Hi-8 and then onto Dolby drives.'

Cochrane mentions the need to digitise the picture into AudioVision at 25fps separately from Media Composer which runs at 24fps.

'The media is becoming very closely entwined but we are still going through a transitional stage where we need to massage the system a little. But its not a problem.'



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music editing to do because the film has been evolving as the picture is edited.

We did reel one in May, which is the opening sequence where Bond gets into the situation, says Eaton. Then in Reel 2 you get into the titles and setup. That was all very exciting because it was back to Big Bond!

That reel has obviously been cut a lot over the months, which is why the Pro Tools was so fantastic. Eaton explains. 'We were able to constantly keep up with the editing and get away with murderous cuts which you probably couldn't do on magnetic—which is where I come from.

'If you have a piece of music and they've made **42** cuts in the reel, you have to work out how you can start off where the music starts and still have the Bond theme come in where there is that particular bit of action. That's the tricky part.'

Editing of effects and dialogue were done at Pinewood (see side bar 'Data-flow and Dialogue') using Avid Audio Vision systems, while picture editing relied on Avid Media Composers. Audio editing alone required around 10 Avid systems.

'It is very expensive, but the reason we had to do this was because the schedule was very short, and because it is a very complicated soundtrack,' offers Martin Evans, supervising sound editor, or sound designer as he would be known in the US.

It is not so much that the time is compressed, it's just that there is so much more to do in that time. I believe that *Diamonds Are For Ever* [vintage and all-mag Bond] had the same

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EX £293 INC In USA \$449 schedule as this."

Evans worked on that film, too, but he was just an assistant in those days. The film was completed quickly, and no changes were made. In comparison, *Tomorrow Neuer Dies* has involved a 'huge amount of recorded sound' which has needed to be edited and re-edited to follow changes to the picture. In order to cope with the workload, effects premixes have been done in Theatre One, while Theatre Two handled dialogue premixes.

Because of the sheer number of edits in the film—reckoned to be the fastest-paced Bond film ever—Avid systems were not used in the dubbing theatre. John Hayward is the film's dubbing mixer, or rerecording mixer to give him the US title. He says that Avid systems 'can do the most wonderful, amazing things' in editing, but that the combination of slower lock-up times than mag machines, and the inability to play backwards meant that they were not the best choice in the theatre for *Tomorrow Never Dies*. Even Tascam DA-88s were not going to provide the palpable lock that Hayward got from sprocketed mag film; although they were used for effects premixes.

In the old days, there used to be brown stuff that these guys would lay and that ran backwards-forwards with instant pickup, says Hayward. It was on the line, controlled by the holes down the edge of the film.

In the event, the final mix used a combination of traditional mag machines, Pro Tools 24 (which Gibbs says benefits from very fast lock) and Dolby Drives.

Pinewood managed to secure two >>>>





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In Pinewood's Theatre One, the SSL desk is used in a fairly conventional, 3-operator setup. From Left to right Richard Pryke; John Hayward and Michael Parker. Michael Parker has been handling effects and the Dolby 8-track dubber. Richard Pryke has dealt with more effects and other inputs.

<<<< Dolby drives, one of which was used to download from DA-88, and the other in the final mix. Hayward says that the Dolby Drives have finally given him back the speed of lockup he used to enjoy a few years ago. The mag dubbers were used on this film 'purely because the Dolby Drives are only in beta at the moment. They haven't got enough machines for me to mix on.' But he adds, I must say, they've done the business for me.'

Hayward describes the full range of source material for the final mix. Tve got four 6-track effects premixes. I've got a 6-track Foley and a 6-track dialogue. Then I've got a 4-track Foley, and a 4-track dialogue, and an 8-track Dolby mix on some reels. and 8-tracks of music. So we're using a full complement of faders, about 60 inputs for the final mix.'

The fader automation of the SSL is used, but Hayward says equaliser automation would find limited use, due to the work that has been done at the premix stage.



Air Lyndhurst: Left to right, Geoff Foster, David Arnold and Nick Dodd

Watching old films, one is sometimes struck by the simplicity of the soundtrack. Because adding another layer of effects meant yet another mag reel—most of which would be full of spacer film—it was not a decision editors took lightly.

Both Hayward and Evans feel that the ease with which editors can now add more effects, sometimes leads them to over spice the whole recipe. In the case of *Tomorrow Never Dies*. Hayward sometimes found himself removing material if it clashed with the dialogue.

The reason for that is because the schedule is so short that the dialogue premix has been going on in another room and John has been mixing the effects,' Evans explains. 'Because of the schedule, we couldn't get the two in one theatre. Normally John would

Knowing The Score

David Arnold (below centre) knows his Bond, as his success with the *Shaken and Stirred* album proves. His score for *Tomorrow Never Dies* was a big production. Geoff Foster, chief engineer at Air Studios, explains the scale of the recording.

'We had a lot of Japanese drums, synthesisers, a band and a full orchestra, so for about a third of the film we were running a 96-channel mix. It is a 72-channel board, and, as It is an in-line, you have 140 returns, but, of course, you also have the reverbs and other effects to accommodate, so it got quite hairy. We have an Allen & Heath that we normally use for headphones, but we used that sometimes as a supplement to the Neve VR.

'For orchestral stuff, I tend to use a Decca tree with TLM50s on it. Then I tend to use AKG C12VRs as outriggers. That Is 90% of the orchestral front-mix sound. Then you have a pair of discrete mics. Different people use different things. I use TLM170s which are set up in the gods behind the conductor, and above the orchestra, which then gets sent uniquely to the surround and creates, hopefully, a phasecoherent thing that sounds as if the orchestra's around you.

'It's a relatively purist approach, but I'm not a purist by any stretch of the imagination. I have no problems with putting out spot mics, to the point of one per string player. Indeed it happened on some places with the score.

The orchestra and the bulk of the band were recorded onto analogue 48, with Dolby SR at 15ips. The synth and percussion we would have preferred to record onto analogue, but we talked it through and the sheer mathematics of it said that digital was the more realistic choice. Some of

the digital tracks were submixed back to the analogue to give it that warmth. The digital was a Sony 48-track recording at 48kHz.

'Because It was going for digital release, I mixed the whole thing in 5-track surround. I haven't bothered with the subbass---the .1 of the 5.1---because there are a number of ways of encoding it, depending which system gets used. They are different versions of a block-off at around about 120Hz.

'I tend to mix to the five speakers only, bearing in mind that with the Surround speakers there is always a danger that it will get pulled because it's upsetting the DS4 matrix. So anything that you definitely want to be in the film should be in the front three. premix the dialogue, and then premix the effects that sit on top of that.'

Despite the demands of a tight schedule and the challenges of new technology, the pair have no doubt that the introduction of digital recording is an advance.

"I would say at least 50% better,' Hayward estimates. 'It's got to be, because you never have to worry about transfer. If everyone did their job right, there was never any problem with brown stuff,' he qualifies. 'But people would sometimes ask you to use secondhand stock because it was cheaper and you'd get oxide pile-ups on the heads.'

Secondhand. for Bond? Never. Mind you, there is a line that begins, 'Mr Bond, I don't think you realise the predicament you are in...', which sounds faintly familiar.

'In my experience, sometimes there are elements that become an issue because they get in the way of dialogue, or because they get confused, in a fight sequence for instance, so you end up with a load of percussion stuff that sounds like punches, but is out of sync with the real punches.

'Because of that I gave Dina Eaton a separate set of five tracks which had just the orchestral elements on it, and then another set of five tracks which had all the band elements.

'There are two advantages to doing it that way. One is that come the dubbing stage John Hayward will have the percussion as a separate element should he need to duck it or raise it. The other advantage is that Dina can edit the various orchestral takes in 5-track mode, instead of having to edit 10 tracks, including the ones that are not wanted. From a housekeeping point of view, and from a memory space point of view, on the Pro Tools, that makes life a bit easier.

'I mixed to a Genex, the reason being that I could record at 20-bit to time code. The time code is relevant to the picture. With the earlier stuff, the picture wasn't locked when we recorded it. Dina has had to edit it and move it about to make it fit the picture. From Reel 9 onwards, the picture was locked when we recorded, so she basically puts it in, locks it up to the same time code as the video that we were all working to, which is in itself locked to the time code of the actual dubbing stage. They hit Play, and there it is.

'There was a certain amount of uncertainty when I said, "I want to do it 20-bit, can you get a 20-bit Pro Tools?" At the time the 20-bit Pro Tools hadn't come out but a few arms were twisted.

'Everyone was saying, "Oh, you can't hear the difference and this is quite a lot of money." But everybody that then heard it said, "Oh, you can hear the difference". So it's going to be a superb sounding score.'

Foster has two valuable tips to prevent reverb from collapsing into the centre or surround speaker when music is encoded for analogue Dolby Stereo.

One is to keep, for instance, violins and cellos hard left and right respectively, with two discrete reverb channels with identical settings. The other is to rerecord real ambience with as near to point source miking as possible, rather than use digital reverb. Foster has found the real thing to be more phase coherent than the artificial alternative.

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A brand new facility has raised the profile of the Greek Capital on the world's recording scene. **Tim Goodyer** explores a mix of local culture and international appeal

ITTING JUST OUTSIDE the immediate area commanded by Athens' Acropolis is one commanded by the audio business. The building programme here is younger and less visually impressive than that of the ancient Greeks', but it embodies aspects of science they would certainly have applauded. They would also have commended the lengths to which this studio community has gone to serve a market made almost exclusive by its language. Located on and around Messogion Street many of Athens' studios, postproduction and mastering facilities, and a number of the record and media companies they serve -this coalition provides for almost all of the country's music production needs.

Right on Messogion Street in the basement of a modern building a completely new music studio was born. Called Odeon, this is a 2-room facility offering Euphonix and Otari consoles. Quested monitoring, a modern acoustic design, separate machine room and comprehensive patching of audio, sync and MIDI. The facility is primarily the result of three peoples' efforts: Makis Matsas, head of the Minos-EMI record company now based in the Odeon building: John Smirneos, veteran of the Athenian studio scene: and Takis Heliotis, studio designer and technical expert. All have enjoyed prior relationships in the city's recording history, but today sees them concentrating on making Odeon the jewel in Athens' crown.

Finished earlier this year and funded by Gema SA. Odeon is really the confluence of a number of individual concepts and ambitions. John Smirneos and Takis Heliotis had been partners in previous projects spanning some 20 years. Studio general manager, technical adviser and sound engineer Smirneos owned and ran Athens' Polysound studios for around 25 years during which his relationship with Minos-EMI flourished. A graduate of New York Institute of Technology and now dividing his efforts between studio design work, equipment support and development of the area's fledgling digital telecommunications infrastructure, Heliotis takes credit for the design of Odeon.

"I had discussions with Mr Matsas about building a studio that was better than Polysound," recalls Smirneos. "There were good studios in Athens, but there was nothing so up-to-date as this. I suppose this is one of the best studios in Europe at this time: there is certainly nothing else of this quality in Athens."

'It took quite a long time,' continues Heliotis, 'because Mr Matsas has been in the record business for a long time, and could appreciate the value of a large recording area and a comfortable control room. Of course, more room means a bigger project and greater investment and we had to find the best balance. >>>>





Studio Sound December 1997









<<<< I think it was a good marriage of our ideas, and the impression so far is that it is one of the best studios around."</p>

The early concept was to run Studio 1 as a fully commercial room and for Studio 2 to be a Mackie-based MIDI programming suite. Somewhere down the line, however, Studio 2 also evolved into a commercial room. Minos, son of Makis Matsas comments. Tin a composer so I'm here as a customer, That's life."

Having seen Minos complete an album project and a number of outsiders come through the doors, the verdict on the facility seems unanimous: 'When you build a studio, it's first the sound and the technology, but it's also about the atmosphere it has,' Minos says. 'When you play music, you must feel at home and it feels very friendly in here. And the musicians who have used the live rooms are very pleased; they hear their instruments in an extraordinary way because of the acoustics. We've had two Greek folk music projects through, and the music for a film is the next project. I am happy with the results and more importantly, the others are happy.'

There are a couple of Hidley rooms, there is an SSL console and there is a Capricorn at The Music Hall so I can assure you that it is the best studio in Greece. says Heliotis. 'More sophisticated equipment is not the only way to achieve success'

'I am very, very pleased with how it's all worked out.' Smirneos adds. 'The desk, the monitors... Believe me, it's very good.'

With the location chosen, the assembled team set about designing the studio.

The desire to make Odeon a thoroughly modern facility meant scouring magazines and touring European and American trade shows. Choices were carefully made to reconcile the quest for excellence with the budget—leading to the selection of a 56-fader Euphonix CS2000 over an equivalent SSL 9000j.

I prefer the MCIs I had at Polysound to the other SSLs. Smirneos comments. The sound of the 4000 is very hard; it's just my opinion, but I don't like the sound quality. The Euphonix is very good sounding and very quiet. And the musicians seem to like it.

The improved Studio 2 was awarded an Otari Status console and Quested H208 midfield monitors. Quested HQ210 main monitors and VS2108 close-fields were chosen for Studio 1 and 7 Tascam DA-88 multitracks were ordered for the machine room.

T prefer the PCM3324, of course, says Smirneos of the tape machines, but the Tascam is a very efficient solution. For us it will fill the gap between analogue recorders and hard disk, until hard-disk recorders are better. For, perhaps, a year they will be the bridge.

There are two or three other studios with big digital multitrack machines,' he continues. 'Others still have analogue and many have ADATs, but I think they are not very good for the sound or mechanically; I prefer the Tascam. We are alone in using it for music just now —postproduction studios in Athens use it but one or two modules only, not so many tracks.'

In keeping with its programming brief, there is also an Alesis ADAT assigned to Studio 2 and this is destined to have a further four machines join it to bring in more work from local project studios.

The Status has automation and is a little

more professional than the Mackie.' Minos says of the console selection. 'But compared to the Euphonix it's night to day; first of all it's the sound. If you record on the Status and then record in here it's totally different—the Euphonix is more soft and human. Then, of course, you have a huge number of options like MIDI control and the dynamics. And the Euphonix is actually very easy to use; it's like a computer game. Perhaps for someone who's not familiar with computers would find it hard to begin with, but it's easy to learn.

The first problem facing the designer, meanwhile, was that the studio had to be accommodated in an area adjacent to the car parking space. The first obstacle, then, was to include sufficient acoustic isolation in a limited ceiling height.

We had some restrictions, confirms Heliotis. We had to build the whole studio in an existing skeleton, so this skeleton is completely isolated from the rest of the building.

Having the restrictions in dimensions we have had to take advantage of every inch of available space. We had to work very efficiently to make the sound isolation and the sound treatment work well. There was not much space, for example, to put in large bass traps. So we searched for very efficient elements to put in. We used elements related to the structure to help the acoustics—for example, the diffusing wall in the live area is also a load-bearing wall floating on another base around the floating floor. We did not want the slightest vibration coming in from the car park.'

The preparatory work took a year to complete and involved endless revisions of the plans to optimise use of the building space. And the careful decisions made in the interests of the isolation of the rooms provided a start when the acoustic treatment was begun. But first the shell was measured and the required treatment was calculated. The aim for Studio 1 was to produce a moderately live recording area and a diffuse control room with a 0.27s reverberation time.

We had already used materials that gave good acoustics themselves and we had to put with them treatments that would give us good and predictable reverberation times.' Heliotis explains. 'I had seen RPG at the Paris AES and knew that their panels are really efficient—the elements are very good looking so you don't have to use special fabrics on top and you can order them in the colours you want. They are also very easy to install.'

While the room measured well as compared with the calculations, it was with the bass that most problems lay—a particular complication in a confined space.

Again the RPGs were very efficient indeed, comments Heliotis, although they weren't the only bass traps we used. The Helmholtz resonators in the control room are custom built and there is another huge bass trap behind the mirror in the live room. We also used bass traps in the monitor wall —there is plaster at the front, then layers of absorption, and then RPG traps to prevent any bass entering the recording room. There is also a space in the centre to have a centresurround speaker or TV monitor if they decide to have one.

'On the rear wall there are wide- >>>>
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<<<< band diffusers, RPG skylines, and bass traps behind the wall and in the corners. With the monitors. Heliotis wanted a big listening area rather than a small sweet-spot, and seems to have achieved it.

The producers we have had in have said, "I do not understand, I'm working all around the control room and the sound is exactly the same"," he says, "With some designs, when the producer is working he will have to squeeze in next to the engineer to hear what he is hearing. This was where we have got it better than we thought possible when we were designing it.

Installing the main monitors involved seeking advice from their designer Roger Quested: 'He recommended we install the monitors as solidly as possible, so the nest of the speaker is isolated using a material used for x-ray isolation-it's thin, but very massive-and around this is a small patch of neoprene. It's all airtight."

The other thing Heliotis claims to have been particularly important is the right choice of cables.

The microphones have to be some way from the console so you must take care of the signal. I cannot tell you how many miles of cable we used, but they are Mogami cables throughout for the audio path. They are very well screened, have low handling noise and excellent phase stability. For the mains distribution we have used a special custom-made industrial screened cable that I have not seen used in the audio business.

The mystique that surrounds hi-fi cabling is not what is at issue here: We put a coil of the mains cable around a mic cable to test it and we found it very clean,' he assures-and that's not all.

When we were setting up the remotes for the machines we called Tascam and asked them how long the cables could be and they said, I think, about 15m, but that we had to





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use their own cables. Because we had a session on we made a cable up out of top-quality computer cable, but the engineers in London were right-it would not work. We were sad because it was a very good Greek composer we had in, so we tried making a remote control cable using Mogami microphone cable and it worked with no problem. We now have the Tascam cable because our cable snake was too thick, but it was interesting that the cable that was meant to be dedicated to audio worked when the broad-band computer cable did not. And I think the cabling is one of the secrets of this studio."

Although only recently finished, plans for the next stage of the studio have already been laid-if only loosely. These include the construction of a mastering room and for a third studio in an adjacent area of the building.

We're not going to build anything for a couple of years,' Smirncos says, 'but we already have a place for a mastering room and we will have another bigger studio to handle a bigger choir or an orchestra. We have work now for cinema scoring and the composer likes to work with about 20 musicians and this room is only for 15 or 16 instruments. Sometimes we need to have 24 or 25 and at the moment we have to do that by overdubbing.

More immediately, Smirneos is awaiting the arrival of Euphonix' TL007 which will give control over MIDI units and all the control, including for the DA-88s, from the desk itself.

The hunt is also on for an Otari MTR90 with Dolby SR to improve media compatibility with other facilities. 'There is more

Odeon Studios, Gema SA, Messogion Street 253, 15451 Athens, Greece. Tel: +30 1 6727 954-5.

than enough work out there to keep us busy," Minos asserts.

The studio will handle any sort of music productions," says Smirneos, "but its only for the Greek market at the moment-unfortunately. We have very good and very interesting music but the language is strange for everyone else in the world. We need someone to become internationally successful, or for people to come from Europe, or anywhere else, to see that we can mix local tradition with the demands of a top professional recording facil-

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Surround acoustics

The demands of equipping a studio control room for multichannel recording and mixing stretch far beyond console buses and joysticks. **Tim Goodyer** sounds out the acoustics of surround

OMMITTING TO MULTICHANNEL audio working is not a move to be taken lightly. The realisation that you have 'graduated' from using a single pair of loudspeakers located somewhere in front of vou, to needing up to eight of the damned things littered around the room is just the beginning. Even accepting the necessity of more buses on your mixing console and the limitations of your existing --- if impressive---rack of 2-channel outboard processors is to miss the full impact of surround sound production. The real deal begins with the fact that the room in which you're going to be working probably hasn't been built vet-even if there's a stereo studio there already, and you're accepting bookings for surround projects.

Although the application is fundamentally different, a surround acoustic does not involve a significant departure from the basic tenets of good stereo monitoring. Indeed, an appropriately-designed discrete surround room should give equivalent performance when a stereo mix is delivered from the surround monitors alone. However, the first thing about discrete surround working is to recognise is that you've left the cosy world of phase-encoded surround systems that simply require a mono surround channel to excite a diffuse soundfield-the kind of thing that's regularly dealt with by wheeling in an odd pair of additional speakers. In fact, White Mark's David Bell believes that there is much interesting work still to be done in the Dolby Stereo field in the area of rear-facing (surround) monitors addressing wide-band diffusion treatments.

Tomorrow's surround systems carry up to eight discrete full-bandwidth channels plus a subchannel. And they require an acoustic design capable of delivering the output of these speakers to the listening position. It's also time to let go of the idea that the centre channel requires anything less than fullyfledged, full-bandwidth reproduction—you're no longer creating 'phantom' images, you're using a discrete centre channel that may be called upon to deliver audio every bit as demanding as the LR pair. The same is obviously true of the speakers all around the room.

Until recently, almost all of the studio design development work had been based around 2-channel, stereo monitoring—with the notable exceptions of a few facilities such as Andy Munro's Air Lyndhurst and Neil

Grant's designs for Whitfield Studios in London, and the Hit Factory in New York where an early decision was made to install a centre monitor channel into the monitor wall. and allow for a single-channel rear surround source as an option. In the case of the Whitfield and Hit Factory facilities. Grant points out that the centre channel made little difference to the design of the room-the forward side walls and ceilings were splayed to minimise and control early reflections, rear side walls are hard, reflective, and splayed to multiply returns into the diffuser integrated with the room's back wall, increasing the reflection density of this wall, and the density of the overall decay of the room. The diffuser ensures that no later returns are specular -that is, not phase coherent with the initial monitor arrival, and are thus not recombined destructively with the monitor signal, but reintegrated by the ear.

These designs are intended to accommodate the LCRS format made familiar by Dolby Stereo. Discrete digital surround systems raise the stakes quite considerably, however. Consider five or seven full-bandwidth reproduction channels focused on the centre of the room: the traditional listening sweet spot. Where a pair of forward facing speakers can legitimately be regarded as originating sound in one soundfield and presenting it to another, surround necessarily finds multiple speakers addressing each others' originating soundfields. Previously, the acoustics of the originating and receiving soundfields could be used as comple->>>>



Grand Central, Control Room 4-London





<<<<< mentary aspect of a single system. Now they must be regarded as interactive aspects of separate acoustic systems.

The demand for multiple channel monitoring formats has forced designers to reappraise the conventional stereo approach completely and take on a multitude of new considerations—from our 15dB hearing discrepancy between front and rear sources to problems with mutual coupling between the speakers serving separate channels. And while the film industry has already become conversant with multichannel rooms, there are few lessons for the audio industry here since film mix rooms have the LCR monitors located so far in front of the console that the listening position is well into the reverberant field, while the surrounds are considerably closer.

A suitable 5.1-format audio room possesses significantly different proportions to film rooms and requires the designer to integrate a 2-channel 'surround' pair, and a single channel 'effects' source fed from a low-pass filter, in addition to the left, centre, and right front channels. It is important to specify centre and rear cabinets to the same level as the LR pair to maintain consistent dynamic range. frequency response, and headroom. Previously, we had only been concerned with maintaining a stereo image at an acceptable height in front of the engineer, and making a reasonable trade-off between console reflections, ceiling reflections, and image height; now panning around and across the room are key issues. and lateral image height is as important as forward image height. Given that early reflections break down the positional information cues





Strong Room Studio 2

The control room space is roughly square giving an even distance between the monitors and the listening position. The monitors are closely matched except that the front 'stereo' pair is capable of delivering higher power levels than the remainder of the installation, in keeping with the requirements of the room in conventional usage—in its surround configuration, the total level is obviously the product of the output of all the monitors.

Front and rear walls are similar in nature with the loudspeaker enclosures set into a dead wall with wide-band absorption. The rear areas of the side walls and the equipment racks provide the reflective areas required to liven the listening environment giving the slightly dry acoustic favoured by the studio. Because of this absorbent quality, the levels required from the monitors are higher than would otherwise be necessary.

When used for stereo mixing, the use of diffusion elements in the central portion of the rear wall tends to spread the image; it also increases the perceived depth of the room in both stereo and surround configurations.

Future development of the room involves the fitting of RPG diffractal units and a programme of measurement and consultation with the studio's clients.

coded onto the original arrival, it is now just as important to control reflections around the whole of the room, as it was around the front of the room when we were dealing with a stereo pair. Having learned how to sustain a smooth pan from left to right, designers now have to deal with panning across the rear pair, and between the rear and forward sections of the system in a similar fashion.

In domestic installations, dipole rear speakers are commonly used to create the illusion of space and depth in the soundfield. Dipoles are particularly well-suited to this application as they display a marked lack of energy normal to the main speaker axis. It is safe to assume, however, that in almost all control-room applications, all of the speakers used will be monopoles, and should match the stereo pair.

As an integral part of the monitoring environment, conventional techniques can be used to predict early reflections, and acoustic treatment used to absorb where necessary. and provide later diffusion and reflection control as necessary. This leads to a room with distributed acoustic treatment around the perimeter, alternatively absorbing, reflecting and diffusing. A number of such rooms have been designed by Harris Grant Associates and are being successfully used already. But there are limitations to the spread of an acceptable image that can be achieved. If a room is being used by a number of people; production staff and assistants over and above the engineering staff, for example, then it is necessary to attempt to 'displace' the rear 2-channel pair acoustically, in order to prevent the listeners localising on the direct field from the monitor. and distorting the image created. It is possible to do this by reversing the speaker into a broad-band 2-dimensional diffuser and creating a virtual source outside the bounds of the room. This not only very successfully removes the apparent source in time and space, but also provides exceptionally smooth panning between the sources, and greatly increases the monitoring stage, the first of these was Grand Central Studios' Control Room Four, which is used for film and video postproduction, voice-over, dubbing to picture, and ADR.

Alternatively, Philip Newell recently observed that the requirements of a multichannel listening room seem to be pushing us toward a more dead environment than we've become used to. Among other things, this will, of course, bring with it the physical discomfort of working in old-style environments from which we thought we'd escaped. But the conflicts implicit in a single-room design to meet all working requirements strongly suggest that it might prove necessary to use separate rooms for tracking and mixing. In this eventuality then, audio facilities would be following the example already set by the film sound community.

As music facilities develop multichannel rooms, an evolutionary step should be taken-these rooms will be being used by engineers who have little knowledge, or indeed little interest, in the film ancestry of the formats, and will only be interested in the effects available to them mixing in surround sound. As a result, recording and mixing conventions will be derived for pure audio effect. rather than because of the necessity of positioning an image against a large-format screen. This is likely to lead to a new generation of mixes that will have real impact on listeners who increasingly have more sophisticated reproduction available in their domestic environments than many of the major professional studios have in their control rooms today.

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US: You're surrounded

Standing on the threshold of a brave new surround world, proaudio is developing new marketing strategies writes **Dan Daley**

OR ONCE, the journalists get some mileage out of a story before the principals in the story themselves do. For the most part, anyway. That's the deal with surround audio, which at a time when the studio business is still a bit on the ropes, holds forth a tantalising promise of profit. Or does i?

It all depends on how you look at it. Surround is different things to different people over here. The engineers who have had the opportunity to do surround mixes are ecstatic. For them, it's like discovering a new world, a 3-dimensional one, where the rules are few and mostly made up by the players as they go along. If you're fortunate enough to be mixing for one of the major contenders in the surround game, all the better, since they're paying top dollar to have certain masters remixed for surround release as part of the marketing strategies for licensing their proprietary technologies. If that's the most lucrative assignments that a few lucky engineers get for surround at the moment, the largess will likely be spread around a bit more widely as record companies-who, like everyone else, really don't have any clue as to whether consumers will demand surround music releases in sufficient numbers to justify their costs—start stockpiling alternative surround mixes of almost everything just in case. At the very least, it should keep more than a few engineers in the studio for an extra week or two per project.

As for the patent holders of these surround technologies—most notably Dolby and DTS, but who are being joined by other contenders—their rewards are already in the pipeline as far as movies are concerned, and they're using some of those profits to proselytise those schemes into more consumer playback systems, where licenses will pay them back for years to come.

Equipment manufacturers love the notion of surround since it will sell more gear—particularly speakers. Think about it: a typical discrete 5.1 systems requires three times as many speakers as your conventional mix room monitoring configuration, not to mention at least twice as many pieces of certain outboard equipment like compressors and effects, and not to also mention the hybrid technologies that are already appearing as appendages to surround systems. This brings us to the studios; how will they fare with surround? Mastering, a black art by nature, is even more opaque when it comes to multichannel audio. Discrete multichannel mastering will give the upper strata of mastering facilities a new upper hand, at least for a while, to counterbalance the proliferation of mid-level mastering facilities that sprouted in the wake of the independent record rage of the early-1990s. It's also the kind of technology that will help justify the triple-digit hourly rates that upscale mastering houses had come to expect.

For the average recording studio, though, surround is a bit more of a slippery slope. It will be a chimera for some, a quick and apparently easy new service to offer that won't cost as much as a new console. Not nearly as much; if it looks to many like all it takes to enter the surround sweepstakes is three or four more speakers and a few new amps, they're right. In fact, in the current fluid environment of surround mixing rules, the less formal the arrangement, the better, allowing engineers to place the surround speakers where they want to create custom soundfields.

But at some point, style will become as important as substance. It always does, and it will be in the trenches of the conventional studio that this will first take place. For starters, it will be as impossible to keep enough different types of speakers on hand to satisfy all tastes, so adroit marketing by manufacturers—which is already under way—is going to inject a fashion element into the surround game, possible

Europe: Testing times

The difficulties of agreeing a standard for DVD-Audio are being compounded by disagreements over watermarking writes **Barry**

WO IMPORTANT DEVELOPMENTS for the recording industry appeared during the month of November. Despite protests from the record companies, the IFPI and RIAA confirmed that their International Steering Committee could not honour the promise to test competing DVE-Audio systems. At the same time the IFPI finished vetting proposals for audio watermarking systems, and passed them on to the independent TNO Laboratories in Holland for testing under the European Esprit project called Muse. If these tests aren't also abandoned, results are now expected during the first quarter of 1998. The DVD Forum has its own team of experts, the Data Hiding Subgroup, working in Los Angeles on similar tests.

The ISC underestimated the difficulty of finding a test method that satisfies the audio industry, which itself has never been able to agree on the best way to judge sound quality. The Committee did not foresee the heavy political and commercials agendas which create disagreement on what system is best even within record companies and between their own artists. North America's draconian anti-trust laws stop business competitors collaborating. 'Abandoning centralised listening tests does

Abandoning centransed insteming

not mean that the introduction of DVD-Audio will be slowed,' say the IFPI and RIAA. Far from it. Sony has already said it will launch DSD in 1999, to try and create a de facto standard. You can be sure that the PCM supporters, like Pioneer, will match the date. The ISC hopes vaguely that, 'the availability of a number of sound formats need not result in separate DVD players in the marketplace'. Having failed even to try and set a standard, the ISC now, cheekily, says the electronics industry should develop a DVD Audio player that automatically self-adjusts to whatever disc standard the owner tries to play on it. The IFPI has for ten years been promoting the use of ISRC, International Standard Recording Codes, which can be buried in the digital bit stream of the CD. But the ISRC code is lost as soon as the music is converted into analogue sound or another digital code format. The target now is to mark the sound itself, so that its source can be identified after digital-analogue-digital conversion, copying and transmission over the Internet. The IFP1 is believed to be referring eight rival proposals to TNO for testing. Because the tests are confidential, the only details available are those released by the companies that are taking part in the tests. US company Arbitron took over Abbey Road in

London recently to demonstrate one of the selected systems, Critical Band Encoding Technology. What makes CBET particularly interesting is that it was developed in cooperation with Martin Marietta, the US aerospace company which had been working for the US Government on methods of recognising submarines by their audible signature. Martin Marietta's team even included Russian scientists who had previously been snooping on US submarines. In a neat twist of irony, Arbitron was demonstrating in the same studio at Abbey Road where ten years ago CBS had tried to convince the audio press and industry that its Copycode system was inaudible and robust. Whereas Copycode sucked notches out of the sound to add an identity signature, CEBT adds a spread of acoustically masked noise code.

Arbitron heard by chance that Martin Marietta had written digital signal processing software to strip submarine sound from ocean noise. Says Ronald Kolessar, Arbitron's Director of Technology: 'The military treats the submarine as wanted sound, buried in noise. For our system the music is the noise, and the label is the wanted sound. But we cannot let the label spoil the noise.' The CBET encoder uses similar DSP software to monitor music in the frequency band 1kHz-3kHz, which contains most of the useful energy and is passed by even the most low-fi radio or phone system. The coder has an armoury of 1026 different combinations of frequencies, each of which conveys a digital symbol. In switched combination the markers convey the ISRC code. The symbols are added in 0.5s bursts, with their frequencies matched to the music causing the kind of equipment scramble familiar to anyone who was in a studio in the early days of MIDI and digital signal processors. This will likely be followed by the need to dress the physical surround environment up, the prospect of which is already making studio designers happy. It won't be long before what looked like a cheap entry fee to a new theme park begins to get costly.

That said, though, there is, perhaps, more opportunity for studios in surround-and on more of a level playing field-than has been seen in years in this business. Those who choose well and wisely could prosper in the most effective way possible in the studio business-by building a lasting perception of being foremost in a field. With the right kind of nurturing of this new service, a facility could quickly reach the critical mass number of successful productions in the surround market to give it a significant perceptual edge. That's the lesson of marketing in general, whether it's in the studio business or any other. But the interesting thing in this instance, I think, is that the advent of surround is the first time a new and widespread service has come into the studio business after the consciousness of the industry had been raised as it pertains to marketing. Now that studios know that perceptions can count as much as realities, it should be interesting to see how differently many of them go about projecting three new speakers to their customers. Gentlemen, start your engines.

frequencies but always at lower levels. So the music always masks the marker, because at any given frequency it is always louder. DSP chips in the decoder are programmed to look for the predetermined burst and frequency patterns, while ignoring the changing sound of the music.

If the music is quiet and subtle, the label must be at very low level, and the decoder builds positive recognition over several seconds. For brash pop, the marker can be stronger and more quickly recognised. So the system could, perhaps, also be used to prove the source of snatched samples. With Copycode it was easy—even the most clotheared listener could hear what the system did to pure tones near the notched frequency. Arbitron gave A-B demonstrations of the Peter Gabriel-Kate Bush duet 'Don't Give Up', from the CD album So. No differences were immediately audible, and the decoder reliably recognised the marker.

If the IFPI, TNO and DVD Forum now have to compare more than half a dozen systems which all work as well as CBET, then declaring any one the outright winner will be an impossible task. But without an agreed common standard, the music and broadcast industry will not adopt any system.

The most constructive outcome may be for the testers to recommend several rival systems, the record industry then encouraging the winning developers to work together on a single standard. More likely, though, there will be no agreed standard and the record companies will continue to complain that they cannot prove their music has been transmitted and copied without authorisation.

Insurance: the mark of a professional

Widely regarded as one of life's little necessary evils, your ability to secure life insurance can be an indictment of your professional lifestyle, writes **Kevin Hilton**

S A YOUNGER person I used to go caving and rock climbing—a hobby I discovered to be fail-safe means of getting rid of door-to-door insurance

salesmen whose brief precluded recruiting those with a death wish. With certain occupations, however, it can be difficult getting any kind of insurance. A few years ago I was arranging contents cover for a new flat; when the company found out that I was a journalist, there were questions about whether I visited 'dangerous' places. Sadly I had to admit that the most dangerous place I was expecting to go to was the registration hall on the first day of IBC.

So I find myself finding new ways of describing what I do for a living: 'writer' is a more acceptable title than 'journalist'. (I have found that it is best to use the alternative in social situations as well, as people now regard journalists as a category only just above grave robbers). For others it is not so simple to disguise what they do; there is, for example, no easy alternative for freelance TV news camera operators, people who often find themselves in the midst of life-threatening situations and who cannot get insured to

save anyone's life.

This fact Was brought sharply into focus at the recent Rorv Peck Awards, held in London and supported by such broadcasters as BBC World and CNN International. The Trust was set up by the family of the highly respected news cameraman who was killed on assignment, with the intention of

providing support for families in similar circumstances and to recognise the work of freelance news gatherers.

This year's Award, sponsored again by Sony Broadcast Europe, was won by Shams Odeh for his coverage for Reuters TV of clashes between Israeli troops and Palestinian police and civilians on the West Bank and Gaza Strip in September 1996. Part of the ceremony was given over as a tribute to one of the most flamboyant freelance news cameramen, Mo Amin, who died last year in somewhat ironic circumstances.

Having worked on some of the most dangerous assignments in the last 20 years, including one in which he lost an arm, he was killed in an on-going news story, as a passenger on a hijacked airplane that crashed into the ocean. Michael Buerk, the BBC reporter who worked with Amin on coverage

of the Ethiopian famine (which inspired Live Aid). said he appeared almost indestructible, a reasonable conclusion given how much the man had survived. It is almost irrelevant to say that those such as Mo Amin and Rory Peck lived right on the edge in their pursuit of a story, sure it was their choice, but in doing it they should have had the best possible kinds of support. One of these should be insurance, if only to lessen the blow to their families if tragedy falls. This is now a possibility. During the Awards, the BBC and CNN announced that pressure was being put on insurance companies to provide basic training and cover for freelance camera operators.

In all this, equipment and technology has a place. In the keynote speech, Peter Annett, a CNN anchor, said that his colleagues were 'increasingly bearing the brunt of coverage' because rates were being pushed down and the onus was on freelancers to buy their own gear. In an essay for the Rory Peck Trust web-site, Ted Taylor, director of technology at UK news organisation ITN, writes that camcorders based on half-inch tape

'have done little to address the weight and bulk issues which are of perennial concern to cameramen and women.' He continues that the use of 6mm cassettes has made units smaller, lighter and more affordable.

Such advances have meant that the term 'film crew' is becoming increasingly redundant; there is now less need for so many people to be involved in location productions,

while the equipment is infinitely more transportable. But this may have brought some of the dangers faced by today's freelancers. In 1990 I interviewed director of photography Barry Noakes, who voiced concern for the well-being of camera crews and criticised the use of one-piece camcorders. Bemoaning the loss of camera assistants, he said, 'If you're filming in somewhere like Beirut, the sound man becomes your eyes. You rely on him to stop you walking in the way of bullets'.

This is sometimes not enough: during the mortar attack in which Mo Amin lost an arm, his sound recordist was killed. As the conflict in the former Yugoslavia has shown, little badges proclaiming 'press' are not shields to gun fire. It is a tragedy in itself that those back at base behind the safe desks are not always aware of this.



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Digital audio recording

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NCE AUDIO has been digitised, it becomes data in the form of binary numbers. The quality of the sound is limited by the sampling rate, which determines the bandwidth, the wordlength, which limits the SNR and the skill of the designer, which determines how close a practical device can come to the limits. In other words after an A–D convertor, the damage has already been done. The advantage of this approach is that the audio signal is now discrete and in a well-engineered system no more damage need occur to it anywhere, ever.

Of course we can't hear this audio signal without going through a D–A convertor, but the D–A is less of a problem. As the A–D determined the quality, a well-engineered D–A won't make it significantly worse.

As the quality is determined in the A–D convertor and to a lesser extent the D–A convertor, it means that the rest of the system doesn't have a sound quality provided it doesn't corrupt any of the binary values.

In fact, there is no such thing as a digital audio recorder. The recording consists of data and the recording mechanism doesn't



Fig.1: Important bits of digital audio recording—see text

know what these data represent. Hence the possibility of playing CD-ROMs and audio CDs on the same drive, or recording digital audio and digital video on the same heads and tracks in a digital VTR.

Consequently, any recording technology which can exist in two states will do. It can be magnetic, optical, disk or tape. The recording could equally be made by a flock of trained woodpeckers or you could leave a pattern of house bricks all over a football field. With this much freedom, the choice of medium is made on other grounds, because they all sound the same. For rapid access, choose a disk; for lowcost archiving, choose tape. For sheer entertainment value, go for the woodpeckers. Around 16.000 of them could record digital audio in real time.

A digital audio recorder is any data recorder that has sufficient data integrity coupled to audiofriendly interfaces. Fig.1 shows what a recorder looks like. Analogue inputs find an A-D convertor (a). AES-EBU digital inputs bypass that. In both cases there is an unbroken real-time sample stream (b) at a constant sampling rate-typically 48kHz. This is a pain to record, so it is time compressed (c). Samples are written into RAM as they arrive, but are read out with a faster clock so they are squeezed together >>>>



Fig.2: Concealment shown here is only used when correction fails Studio Sound December 1997





<<<<<>><><< into blocks (d). The gaps can be used to insert addresses and error-correcting words (e) into the bitstream.

Separate blocks are much easier to record on discontinuous media such as hard disks (f) and rotary head formats such as DAT (g).

On replay, the time compression needs to be reversed. This is done with a second RAM (h) called a timebase corrector, which is read out with a stable clock from a crystal. If this is done properly, the time base stability of the storage medium is completely decoupled from the timebase of the audio sample output (i). Thus a digital audio recorder doesn't have wow and flutter, unless there is something wrong with it.

Note that the timebase accuracy of the timebase corrector is only sufficient to produce an AES-EBU output with high data integrity. It categorically does not mean the clock or the data stream are stable enough to drive a D–A convertor. The D–A must have its own clock reconstruction system (j) so that only the sample values affect the output waveform, not the sample timing or jitter. So it is possible to bypass the recorder completely (k) without any change in sound quality. Clearly if the sound quality doesn't change when the recorder is bypassed, the recorder doesn't have a quality.

I find it pretty hilarious that people still do listening tests on different digital audio recorders and formats. If any differences can be heard with the same data through the same D-A convertor, and they often can with real equipment, there's something wrong. The trouble with statements like this is that they can only be said once, and this is unacceptable when it is necessary to continue to fill the pages of hi-fi magazines. The practical solution reached is that the foregoing theory simply doesn't apply to hi-fi journalists who find it exciting to suspend the laws of physics for a good varn. I don't have a problem with that at all, provided the reader knows where reality gets off and science fiction comes in.

Back on earth, the digital recorder simply has to record and recover data without changing any of the numbers. As error-free media don't exist except under Stalin, this can only be done with the help of error correction systems. There isn't space to explain how error correction works here, but it's probably more important to the user to know what it does. As one record producer actually said: 'All I know about Reed-Solomon is that it allows me to throw up on the medium'.

Prior to recording, the data have check words, known as redundancy, added. On replay, the check words are compared with the data and if the discrepancy is within certain limits, the missing values can be calculated and the erroneous samples can be replaced. Now there's no difference between the value 26,123 which was never wrong and the value of 26,123 which was corrupted to 26,10⁻⁷ and then put back to 26,123, so people who claim to be able to hear error correction working have either a good imagination or poor equipment —which it is doesn't interest me.

An infinitely powerful error-correction system would be great. Imagine taking the

CD out half-played and the machine finishes playing it. Don't get excited, it's not going to happen. Error correction systems have finite power which is enough for most circumstances. On a bad day there will be too many errors. Suppose your DAT tape has been left in the boot of the car all night and it's freezing when it's put in the machine. The tape condenses and the head starts water skiing. Suppose your CD has just too many fingerprints on it.

In this case, the capacity of the error correction system will be exceeded and calculation of the missing values is impossible. In order to degrade gracefully, practical machines switch to concealing the errors. Concealment consists of estimating what the missing samples might have been from those that are left. Clearly it's not perfect, but if the value of 26.123 is lost, it's better to have the machine guess at 25.997 than to reproduce the erroneous value of 2.221 which would put a massive click in the waveform.

Fig.2 shows that concealment is made possible by re-ordering the samples so that odd samples and even samples are recorded in different places on the medium. If a dropout does a lot of damage, after putting the samples back in the right order the result will be that only, say, odd samples are lost at one point. The even samples can then be interpolated to estimate the value of the missing samples.

Concealment like this was used in the Ferrite Age (c.1982) to allow digital tapes to be edited by splicing. The splice did the recording no good at all, but the mess was concealed using a huge odd-even shift. Naturally manufacturers provided splice editing because users asked for it. The manufacturers said, 'do you want digital tapes to be splice editable?', and everyone said, 'Yes!'. It's a pity the manufacturers didn't say, 'Now that fast, alterable, sample-accurate electronic editing is feasible do you still want people playing with razor blades?'. It must have been the origin of Blood on the Tracks.

Concealment isn't perfect and so it can be audible, whereas correction is perfect and isn't audible. All practical digital audio formats are designed so that the error correction handles all of the errors in normal operation. Concealments should be negligible. In practice digital recorders degrade in service. Tape heads get dirty and wear, focus circuits drift and lasers wear out, causing the raw error rate to rise. With an error correction system, the correction just works harder and there's no audible result until the power of the correction is exceeded and it dies.

This is dangerous, as you have no idea when a digital recorder is going to die just by listening to it. The only safe option is to display the rate at which corrections are being made, compared to the maximum correction rate possible. This allows a tangible assessment of how well the machine is operating. In fact this is such an important requirement that I would say a machine without some kind of error correction rate display simply isn't professional. In all error correction systems the necessary information exists within the processing, so it's only a matter of adding the display. ■

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Trouble in paradise

With the rise in interest in audio-only surround production, issues such as centre-channel, low frequency and mutual coupling arise. **Philip Newell** and **Keith Holland** offer some critical observations

MAINSTREAM music-only surround looming ever nearer, the audiophiles' microscopes will soon be brought to bear. Fivechannel or 5.1-channel surround frees us from many of the limitations of stereo and offers us a new range of possibilities, but it does bring along its own problems, one of which is the optimisation of how to reproduce the low frequencies most accurately.

In discrete 5-channel surround, we have 5 possible lowfrequency sound sources: left, centre, and right front, and two rear loudspeakers. In relatively anechoic listening conditions, as would seem to be optimum for frequency surround. the response at the listening position would largely be that of the axial response of each loudspeaker for any sound source routed to one loudspeaker only. One of the beauties of surround is that it

Too many people have seen stereo as a simple solution to everything, whereas in reality, it is actually a complex solution to very little. The complexity of stereo pales when compared to the complexity of the interactions between surround loudspeakers

offers a discrete centre-front channel, which not only anchors that sound positionally, but gives us a more predictable reproduction in a variety of listening conditions.

A mono source drives a room in a much simpler manner than a stereo source as there is much less interaction with the complex modal patterns, but even in an anechoic chamber, or any other free-field listening conditions, two sources cannot give a uniform response except on the centre-line which runs between them. A perfect single source of

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low frequencies, in a perfectly anechoic chamber, would produce a frequency response as shown in Fig.1, and would have a polar pattern as shown in Fig.2. If we were to replace our single source with a pair of sources, in order to derive a phantom centre image, then we would still achieve the frequency response of Fig.1 on the centre line between the sources. However, if we were to move off axis, then the responses typical of those shown in Fig.3 would result. This is because the path-length differences to the two loudspeakers would cause comb filtering, as different frequencies arrived with different phase relationships. This fact is well known, but it is highlighted by the quite startling polar patterns shown in Fig.4 which shows the combined polar patterns for two loudspeakers reproducing the same signal, at the same level, as would be the case when reproducing a central phantom image in stereo.

If we transfer our loudspeakers into a more typical set of listening conditions, such as would be the case for most of the music buying public, then room reflections, even on-axis, interfere in such a way that a response somewhat like that shown in Fig.5 results. Only at low frequencies. where the wavelengths are long. does the interference remain wholly constructive. The result is a 3dB rise in low frequencies, up to a frequency determined by the distance between the loudspeakers. This is due to acoustic mutual coupling between the loudspeaker sources, where the pressure produced by each loudspeaker superimposes itself on the diaphragm of the other loudspeaker. As moving-coil cone loudspeakers are constant-velocity sources, more work will be done when the added pressure from the other loudspeaker is resisted (in order to maintain constant velocity), and more work done will result in more power being radiated as the cones become more resistively loaded.

Except when listening on-axis in anechoic monitoring conditions, we will perceive the mutual coupling, bass boost,







Fig.2: Polar response of a 'perfect' loudspeaker in an anechoic chamber





phenomenon as we pass through the centre when panningalthough on signals without much low frequency content, this may be of little relevance. In all cases, the timbral quality will be affected by the inter-aural distance differences, which cause cancellations around 2kHz, as each ear receives sound via one shorter and one longer path length to each loudspeaker when listening to a centre image. At the left ear the distance to the left loudspeaker will be shorter than that to the right loudspeaker by typically about half a wavelength (out of phase) at 2kHz, and vice versa for the right ear-see Fig.6.

We have learned to live with these problems in stereo recording and reproduction, though they are the cause of many sound compatibility problems from one environment to another. Too many people have seen stereo as a simple solution to everything (if they could only get it right), whereas in reality, it is actually a complex solution to very little. Nevertheless, the complexity of stereo pales when compared to the complexity of the interactions between surround loud->>>>


Fig.4: Polar response of a pair of 'perfect' loudspeakers when producing a central phantom image



Fig.5: Frequency response of a pair of 'perfect' loudspeakers when producing a central phantom image, on-axis, in a typically reflective room. The OdB level is the response for one of the loudspeakers in anechoic conditions



Fig.6: Path length anomalies for phantom central image

<<<< speakers. Stereo supports one phantom sound stage from two loudspeakers: surround can support 10 phantom sound stages between any 2 of the 5 loudspeakers, and many more when one signal is shared by 3, 4, or 5 loudspeakers.

As the same rules of acoustics apply in stereo and in surround, we should consider a situation where a signal is panned into an in-room location-sav by being fed to the corner loudspeakers. The same rules as for the plot shown in Fig.5 apply, but with 4 loudspeakers interacting, we get the response in Fig.7. The problem has clearly become serious, as a 6dB low-frequency boost is unlikely to be subjectively acceptable. What is more, it is not possible to 'equalise as you pan', because the upper limit of the boost is dependent upon the distance between the loudspeakers.

The response at different listening positions is also a greatly variable factor for a signal panned to the corner loudspeakers. Fig.8 shows the low-fi response plots for a listener at two positions in a room other than the centre between the loudspeakers. Again, the problems are significantly worse than in the corresponding situation for a stereo pair of loudspeakers. If such a 'creative' use of surround sound is envisioned, then the use of a single subwoofer is desirable.

There is still much discussion about the subjective quality differences between the use of fullrange stereo loudspeakers, and satellites and sub-woofers. It has been my experience that separate sources sound more natural, but the difference reduces with the frequency of crossover. For stereo installations in discotheques, I prefer to restrict the subwoofers to frequencies below 80Hz, though to let them go down to 15Hz or less: 100Hz appears to be about the upper limit before dislocation of the respective sources becomes apparent. On classical music, though, when surround is used to maximise the sensation of being in a large hall, the LF wash appears to be a rather important aspect of the sensation of realism. It has also been noted that with organ music recorded in churches odd phase artefacts are evident in the low frequencies, which close in if a single subwoofer is used. I believe that a more natural reproduction is achieved by the use of separate LF sources for each position of

the surround, but this only holds true in listening environments with well controlled low frequencies. As LF reflections increase, the situation rapidly changes in favour of the single sub-woofer approach. From looking at Fig.[–] and Fig.8, it is little wonder why this should be the case.

We thus appear to have a situation whereby discrete low frequency sources can scale greater heights of fidelity-if recordings are made and mixed with due regard to the electro-acoustic constraints of a 5-channel system. If ignorance prevails, then a 5.1channel system is a safer option. However, it should be noted that due to mutual coupling and multiple-source effects, the optimum electrical balance of a mix, at low frequencies, will differ in the 5 or 5.1 formats. Let us consider what is good recording mixing practice within the 5-channel limits.

Discrete low frequency sources can scale greater heights of fidelity if recordings are mixed with due regard to the constraints of a 5-channel system. If ignorance prevails, then a 5.1-channel system is a safer option

The strength of surround comes from it having 5 (or more) discrete sources of sound. This can be further enhanced by ceiling loudspeakers, but let us restrict ourselves to the current format, for now. The weakness of surround comes in the number of possible phantom stages that can be supported-26 or so. The question is; given 5 discrete sources, do we need to use the phantom sources for anything of great importance, and, if so, can we maximise the potential and limit the damage? So much of the compromise to stereo has been as a result of trying to over exploit what it had to offer. Similar greed in the use of surround sound can carry even greater grief potential. Surely the best use of surround is to provide a 3channel stereo front sound stage with ambience in the rear loudspeakers. Any centrally placed instruments or voices, whether in classical or more modern >>>>

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Fig.7: Frequency response of four 'perfect' loudspeakers reproducing the same programme in a typical room



Fig.8: Combined frequency response of four omnidirectional loudspeakers at two different positions in an anechoic chamber

<<<< music, should be fed only to the centre-front channel. We then have two possibilities for the centre-to-left, and centreto right images. We can pan them between left and right, as in conventional stereo, or we can pan them between either side and centre. Psychoacoustic considerations suggest that the former is better for on-the-centre-line listening positions, and the latter for off-the-centre-line positions. perhaps trading stability of the image for some perceived clarity. The subtended angle of the LR pair could also have an influence on the choice

When single sub-woofers are used it is often good practice to mount them off-centre, so that low-frequency room modes are not symmetrically driven. This practice tends to cause more peaks and dips in the overall response in the room, but of less severity than the fewer superimposed response irregularities caused by symmetrical driving. When this is done, I recommend offsetting the sub-woofer to the right, because in orchestral music, this is where the basses will be coming from. Much has been published, and much talk is bandied about, on the subject of the lack of directional information below 300Hz, but this early work was done with sine waves. In the case of music, arrival times and the shapes of transient wavefronts carry much directional and timbral information, and I can guarantee that in almost all cases, except in acoustically poor rooms, an orchestral bass section on the right with a sub-woofer on the left will not sound as natural as with the sources co-located. Obviously, with a centre-panned bass guitar, the offset to the left or right is immaterial. This is just one of many aspects of low frequency reproduction to be taken into account that are so often ignored. And what of the case of discrete rear-positioned instruments with a single front subwoofer? Different listening positions cause unacceptable arrival time differences, which muddy the response. There are many points which need to be considered further before launching recklessly into audiophile surround mixing.

Certainly in the cinema world, it is relatively unusual for any signal to be fed to the front and surround loudspeakers simultaneously, except, perhaps, for an explosion, or something similar, where the full power of the whole system is needed for effect. Cinema people also tend to refrain from putting discrete images in the surround loudspeakers, because they do not want people to be distracted from the screen. It will almost certainly be the exploitation of such practices by the music-only surround mixers which will cause the greatest problems in surround compatibility, and the greatest disappointments in their domestic reproduction.

What the music-only mixers must realise is that acoustic sum-

mation of the outputs of surround loudspeakers in non-ideal, domestic circumstances, and the electrical summations which they will need for folddown into stereo and mono can be wildly different. If they do not fully respect this fact, then they will only have their own ignorance to blame when they find that their surround mixes do not travel well to other systems. A careless mix will also be unlikely to folddown to stereo with the instrumental and tonal balances intact, and what will come out of a mono radio is anybody's guess.

There is also another point which people will have to get used to for music-only surround-that the most important loudspeaker is in the the centrefront position. The nonsense of using a smaller loudspeaker here. and considering it to be a dialogue-only channel is folly in the extreme. This loudspeaker needs to handle, alone, things such as bass guitars, bass drums, and vocals, which are currently often split between an LR pair. If anything, the centre loudspeaker should be twice the size of the left or right loudspeakers, not half the size.

If special mixing rooms need to be specially constructed for audio-only surround, then so be it. Many sound recording studios have been adapted for surround use, and there has been a tendency to compromise centrefront loudspeakers for the sake of better vision into the studio, but how long the situation can remain is uncertain. Once audioonly surround becomes more common-place, dedicated surround sound-only mixing rooms will be needed, just as the film industry uses dedicated mix-only rooms. This may lead to a situation where the video people are left out on a limb with their concept of the compromised centrefront channel. Further 'clarifications' may well be needed when it comes not only to the appropriate use of centre loudspeakers and sub-woofers, but also to the choice of rear loudspeakers.

The rear loudspeakers make their own different demands for the different concepts of reproduction. If we are going to want to place discrete instruments in the rear channels, then good fullrange loudspeakers will be needed here. This is another area which may conflict with the film and video world, though there are good reasons why the audioonly mixes should not exploit this option, except, possibly, as in the video film worlds, when special effects or occasional effects are required. Ambient surround needs good dispersion of the sound, and may well be best generated from relatively diffuse sources. Panel loudspeakers seem to show some potential in this area. Hi-fi sound at the rear, however, tends to demand discrete, localised sources, but under most common circumstances, this is fraught with problems.

As can be seen from this brief look at music-only surround, there are a number of areas which need multidisciplinary discussion and assessment if we are not soon to find ourselves in a situation where the film, video, music, specialised home theatre, and possibly other interests are not to all go their separate ways, leaving a confusing overall situation for the consumers, and expensive multiple mixes for producers. The very diversity of use which DVD has to offer will soon force many decisions upon us, and it would be a pity if the whole thing degenerated into the sort of incompatibility nonsenses which led to such a fiasco with the quadrophonic recordings of 20 years ago. We have been warned, so if we fail, there will be no excuse.

One final point worth mentioning, here, is the required storage capacity for DVD. To restrict the format to 6-channel (or 5.1) to-bit ++.1kHz is ridiculous. however, 24-bit 96kHz on all channels would be overkill, and time limiting, George Massenburg seems to be currently arguing the case for 21-bit 96kHz as being as far as we need to go for audiophile audio, in general. In my opinion, this would be ideal for the three front channels, and 16bit 48kHz would be sufficient for the rear channels, as we do not have super-critical sonic resolution ability when listening with the backs of our heads. The '.1' channel, of course, needs only a narrow bandwidth. As an industry, we should begin to demand what we need, as professionals, and not roll-over in submission to the whims of the politics of marketing departments who dare to tell us what they think that they will grant us. Just say no to data compression in audio-only surround.

Reference

1) Newell, PR, 'From Mono and Stereo. Through Quadrophony. to Surround'. *Proceedings of the Institute of Acoustics Reproduced Sound 13*, Vol. 19, Part 6, pp135—pp154. October 1997

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Black market

What sort of image does our industry present, and what part do magazines play in creating this image? **Nick Owen**, commercial director of Soundscape Digital Technology, asks the questions

ALK INTO ANY MUSIC shop, any-where in the country, get a price, then call us and we'll beat it' ran the claim in a magazine advertisement earlier. this year. The advertiser, one of the UK's leading dealer-distributors, was also advertising certain products as 'POA' (Price On Application) along with an adjacent 'now shipping' star. These products were not, in fact, available and had only ever been seen as empty boxes and a spec sheet at a trade fair.

The result is that as these manufacturers have no product to sell some have resorted to unprofessional and dubious sales techniques during public demonstrations at our trade shows. I witnessed examples of this at the AES Convention in New York, where several demonstrators resorted to slandering other manufacturers' procuets in order to make their own sound good. What baffled me was that some of the products mentioned were not even targeted at the same market area or price range.

Recently I received a transcript of a demonstration, which was recorded by a journalist using a lapel microphone, given by one particular offending Jemonstrator who on numerous occasions throughout his demorepeatedly spouted untruths about other gear What was totally astcunding was not only the lack of knowledge of his own products but the technical inaccuracy of the other products denigrated in his demo.

Surely products should be sold on their merits, specifications, price and target market and not by these types of scare tactics. It is totally unprofessional, unethical and often slanderous and this type of activity. should not be condoned by any manufacturer to sell its products.

With so many new 18V phantom power companies now climbing aboard the digit.d audio recording bandwagon, and consequently presenting the potential customer with a bew ldering (UK) array of information (some of it accurate, some of it not so accu rate), isn't it time that our magazines took a more responsible and consumerfriendly attitude towards helping people separate the wheat from the chaff? This

involves not only identifying good and poor products, but actually helping to define the market they serve-that is, identify which products genuinely represent good value in terms of cost, build, performance, upgradability and application. The other key point here, and one that is often overlooked as I mentioned earlier, is availability.

Does it really benefit anyone to have sections of the market perpetually delaying purchasing decisions on the basis of a new spec sheet

To cite a case in point. I recently read a t-page review-cum-article, written by a staff writer in one particular pro-audio publication, on a product range that not only lacks a firm. shipping date, but doesn't actually exist---two crucial points for any genuine buyer that were not addressed anywhere in the piece. There have been other articles which refer to prod-

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ucts launched for a year or so, and yet are still unavailable. For my money, the number of column inches devoted to these products is a complete waste of time-both from the point of view of the writers and the readers who read them.

F MANUFACTURERS continue to pursue this kind of marketing strategy unchecked. the result will surely be a confused and bewildered market. Surely, this is a situation that is bad for both customers and manufacturers, and one that will create a poor image for the industry as a whole. This latter point is particularly unfortunate, as we could well do with casting ourselves more clearly, not less so. What would be far more helpful would be a sensible level of vetting of new products, indicating shipping dates if required. Does it really benefit anyone to have sections of the market perpetually delaying purchasing decisions on the basis of a new spec sheet and a photo, or a computer-generated image?

In addition, it is clear that many computer audio customers actually buy a high-spec computer, software and an audio card(s), and expect to be disappointed by its performance. When their expectations are met, they accept them as being representative of the audio industry as a whole. In contrast, few car buyers would spend their money on a product that whimsically turned on and off as more speed was required from it, or failed to deliver the driver to the desired destination dry and in comfort. I'm sure that a company selling mixing consoles that regularly failed during a live gig, would not get away with such a low performance, so what makes computers and software such an anomaly?

In the light of all this, would it not be reasonable to expect that our industry publications pay attention to what companies have done in the past as well as what is in the future when evaluating new products? The factors that determine quality of products in our market are no different to any other, it's just that no one seems to have noticed yet.

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broadband musical

Constant loudness pan

Stereo in-place Solo.

Mute button routes

signal to "bonus"

Alt 3-4 stereo bus

2.5kHz peaking Midrange & 80Hz Low

shelving EQ,

shelving EQ.

controls.

CHS.

BOTH MIXERS ARE QUITE AFFORD-ABLE AND **ARE NOW** IN STOCK AT YOUR LOCAL MACKIE DEALER CHECK THEM

TODAY!

recording and live applications. MS1402-VLZ DNLY: 60mm logarithmictaper faders based

outputs & Control Room

matrix. Handy for both

on our exclusive 8.Bus design. Long-wearing wiper material and tight polymer lip seals to protect against dust & other crud.

Built-in power supplies — no outlet-eating wall warts or hum-inducing line lumps.

Phantom power so you can use high quality condenser microphones. XLR outputs with mic-line level switch (along with 1/4" TRS outputs on top

panel).

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RIGHTS RESERVED

MS1202-VLZ = 12x2 = 4 MIC PREAMPS



MS1402-VLZ = 14x2 = 6 MIC PREAMPS



All inputs & autputs are balanced¹ to cut A A hum & allow extralong cable runs, but can also be used with unbalanced electronics. ¹ except RCA tape jacks, heaphone jack & inserts.

VLZ (Very Low Impedance) circuitry first developed for our 8.Bus console series dramatically reduces thermal noise & crosstalk in critical areas.

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Control Room outputs feed monitor speakers without tying up the headphone jack.

Radio Frequency <u>R</u> Interference protection via metal jacks &

BOTH MODELS HAVE:

washers plus internal shunting capacitors. High-output headphone amp can drive virtually any set of phones to levels even a drummer can appreciate.

Aux I Master level control & pre/post switch.

Effects Return to Monitor Return 1 effects into a stage monitor mix via Aux Return 2 level control.

RCA-type tape inputs & outputs. Peak-reading LED meters with Level Set LED combined with

In-Place Solo allows fast, accurate setting of channel operating levels for maximum headroom and lowest noise floor.

Control Room/ E. Phones Matrix adds monitoring, mixdown & metering flexibility. Select any combination of Main Mix, Tape In and Alt 3-4 sianals for routing to phones, Control Room

outputs and meters. Can be used as extra monitor or headphone mix, tape monitor, or separate submix. Way cool.

Tape Assign To Main Mix assigns unbalanced RCA tape inputs to main mix. Besides its obvious use as a tape monitor, it can also add an extra stereo tape or CD feed into a mix or play music during a break.

- MS1402-VLZ only: Global Sola Mode selects PFL or AFL solo modes.
- Solid steel chassis & thick fiberglass internal circuit boards resist abuse. Channel inserts on mono channels.

at every production project

requires dozens of input channels and boatloads of buses.

But doing ANY audio iob well requires a mixer with superb specs...and the right combination of useful features.

Our MicroSeries 1202-VLZ and 1402-VLZ might have small footorints, but when it comes to performance, they walk very tall.

Since both are basically chips off our blockbuster 8-Bus Series consoles, they have big-board specs: greater than 3 90dBu signal to noise ratio, less than

0.005% distortion, more dynamic range than compact discs and frequency response that's only down 1dB at 60.000 Hz.

Why own an imitation when you can own the brand of compact mixer that serious pros prefer. Call for info today.



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4 MOND CHS. **4 STERED** CHS.

MICRO

SERIES

1202

VLZ

MICRO SERIES 1402 **VLZ** 6 MOND CHS. **4 STERED**



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OUT

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Extensive surround sound capabilities make the D950 the most exciting Digital Mixing Console for today and tomorrow.

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